



Audio Engineering Society Convention Paper 8475

Presented at the 131st Convention
2011 October 20–23 New York, USA

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Proximity effect detection for directional microphones

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ABSTRACT

The proximity effect in directional microphones is characterised by an undesired boost in low frequency energy as the source to microphone distance decreases. Traditional methods for reducing the proximity effect use a high pass filter to cut low frequencies which alter the tonal characteristics of the sound and are not dependent on the input source. This paper proposes an intelligent approach to detect the proximity effect in a single capsule directional microphone in real time. The low frequency boost is detected by analysing the spectral flux of the signal over a number of bands over time. A comparison is then made between the bands to indicate the existence of the proximity effect. The proposed method is shown to accurately detect the proximity effect in test recordings of white noise and of other musical inputs. This work has applications in the reduction of the proximity effect.

1. INTRODUCTION

A microphone is a transducer that converts sound pressure waves to electrical signals. A linear non-directional, or omnidirectional, microphone has a flat frequency response and responds equally to sound pressure from all angles at all distances. A directional microphone responds to sound pressure primarily from one direction. This can be used to improve the signal to noise ratio of a single sound source in a noisy environment. A consequence of directionality is that a flat response has to be sacrificed due to the proximity effect, characterised by

an undesired boost in low frequency energy as a source moves closer to the microphone, beyond what is expected.

The proximity effect can cause distortion of the input signal as the low frequency boost will also boost the overall amplitude of the signal, for example if a person speaking unexpectedly moves closer to the microphone. This is particularly evident in teleconference situations. In a live musical performance for example, musicians naturally move while performing. This movement changes the source to microphone distance and can therefore cause

undesired tonal changes that cannot simply be corrected using equalisation.

In commercial products, the proximity effect is tackled in a number of ways. Some condenser microphones have two diaphragms to provide selectable polar patterns. This can also be used to reduce the proximity effect [1] by effectively enabling a cardioid polar pattern for high frequencies and a non-directional pattern for low frequencies. Although this will reduce the amount of low frequency boost the presence of a non-directional microphone even at low frequencies will increase the amount of noise in the microphone signal as it is reproducing sound waves from all directions. The additional components required will also increase the cost of the microphone.

Other microphones simply include a bass roll off in an attempt to reduce the effect but this can alter the sound and remove low frequencies that may not be boosted by the proximity effect. Equally, a multi band compressor can be used with the lowest band set to cover the critical proximity effect band which varies with each microphone. As with simply using a filter, sound that may contain a lot of low frequency information will also be affected.

To the author's knowledge the research into the proximity effect is limited. The causes of the proximity effect are not fully understood [2, 3, 4]. Prior work compares theoretical low frequency boost to real microphone data [5, 6, 7, 8] where theoretical models are shown to be lacking and do not correlate with recorded data. The proximity effect is generalised as a boost in low frequencies but varies for each microphone due to the differences in construction. An example microphone response in [5] exhibits a distinctive peak in low frequencies as distance decreases between source and microphone at around 160Hz with a general increase below around 500Hz whereas other microphones have a gradual increase in amplitude at all frequencies below around 500Hz. For this reason in this paper the upper limit of the proximity effect will be defined as 500Hz.

Attempts to reduce the proximity effect are limited as they are unable to take into account the absolute distance of the source and microphone. If absolute distance data could be found then this could be coupled with microphone data and the proximity effect accurately corrected. The majority of algorithms for calculating source to microphone distance and angle use microphone arrays

which require knowledge of the array and at least two microphones [9]. Research in [10] outlines a method to estimate the absolute distance between a single source and a single microphone by using statistical parameters of speech which inform a pattern estimator algorithm. The method is shown to perform for close distances but requires training of the algorithm and is only for speech signals.

2. PROXIMITY EFFECT

All directional microphones exhibit the proximity effect. The low frequency boost occurs due to the method used to enable directionality in microphones. Typically a microphone contains a diaphragm that is excited by incoming sound pressure waves and converts this vibration to electrical energy. Microphones are made directional by controlling where the sound pressure arrives at the diaphragm. In a non-directional microphone the rear of the diaphragm is sealed in a vacuum and the front open to respond to sound pressure. The output is therefore the absolute sound pressure at the diaphragm. A directional microphone is open at both the front and rear of the diaphragm and the output of the microphone is the difference in sound pressure at each side of the diaphragm, or the pressure gradient.

The difference in sound pressure is caused by a difference in amplitude and phase of a pressure wave as it arrives at either side of the diaphragm. An on-axis pressure wave travels further to reach the rear of the diaphragm thus the inverse square law dictates there will be a drop in amplitude and therefore a difference in pressure that is frequency independent. The distance also dictates there will be a frequency dependant difference in phase of the pressure wave from the front to the rear. These two components combine to provide an overall pressure gradient.

The amplitude gradient caused by a pressure wave arriving from a source far from the microphone will be small compared to the phase gradient. As a source moves closer to the microphone the phase gradient component decreases, especially at low frequencies, and the amplitude gradient component simultaneously increases resulting in a boost of low frequencies [11]. In addition to this, the proximity effect is also dependant on the angle of the source to the microphone. For example a bi-directional microphone reproduces sound primarily from sources arriving at angles of 0° and 180° . At source angles of 90° and 270° the microphone reproduces very little and therefore exhibits the least proximity effect.

The low frequency boost in decibels, β_{dB} , due to the proximity effect is described in [12] and adapted from [11] as follows

$$\beta_{dB} = \sqrt{\frac{1 + \left(\frac{2\pi r}{\lambda}\right)^2}{\frac{2\pi r}{\lambda}}} \quad (1)$$

where r is the distance in centimetres and λ is wavelength in centimetres. This is a generalisation as the cut off and gain of the equivalent low pass filter is dependant on the microphone architecture [4].

To the author's knowledge there is no previous literature on using signal processing and analysis to detect the proximity effect. Work in [13] attempts a similar goal with pop sounds which involves a 2 stage process of pop noise detection and suppression.

3. ANALYSIS

Detection of the proximity effect first requires understanding and analysis of how it affects microphones under real conditions. Figure 1 shows the low frequency gain below 500Hz of a white noise input signal recorded at various distances to a sound source using both an omnidirectional and cardioid microphone.

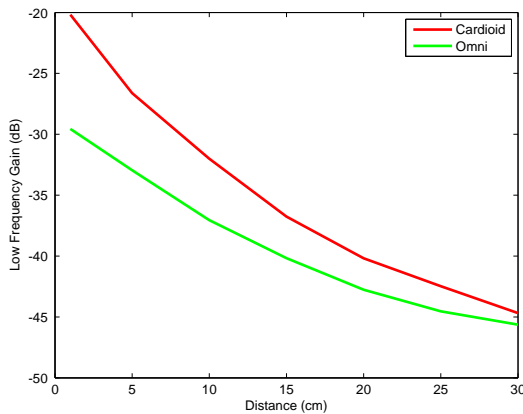


Fig. 1: Gain of white noise low pass filtered at 500Hz recording with cardioid and omnidirectional microphones at different distances

The proximity effect can clearly be seen. At the smallest distance of 1cm there is almost a 10dB difference in level. At 30cm and above, there is a less than 1dB difference in between the two microphones.

4. PROXIMITY EFFECT DETECTION

Proximity effect detection is more than a simple analysis of the low frequency content of an input signal. There are many occasions where a change in low frequency content is not due to the proximity effect and is due to other scenarios such as an instrument simply playing a lower note. The low frequency content of a signal will also be boosted, regardless of the microphone, but a directional microphone will boost the low frequency content further than is expected for the distance between source and microphone. The low frequency content will also be boosted if the sound source simply becomes louder.

Different microphones also exhibit the proximity effect in different ways. Previous work [5] analysing microphones shows that some microphones have a more uniform boost in low frequency content below a certain frequency and others may have a more prominent boost around an area of the low frequencies. It is therefore difficult to apply "one size fits all" approach. A generalisation can be made that the proximity effect is most apparent below 500Hz from analysis of previous publications and microphone data [4].

In this approach no assumptions are made or prior knowledge of the microphone and only the microphone data is available. The aim of this approach is to detect when the proximity effect is occurring.

4.1. Spectral Flux

As the proximity effect is a spectral effect certain spectral features can be extracted, such as spectral flux [14]. Spectral flux is a measure of the change of spectral content over time. It is calculated by taking the Euclidean distance of the magnitude of subsequent frames of data. This is described by

$$SF(n) = \sqrt{\sum_{k=0}^{N-1} [X(n, k) - X(n-1, k)]^2} \quad (2)$$

where X is the microphone signal x in the frequency domain, k is the bin number from $0, \dots, N-1$, N is the window size and n is the current time step.

It is expected that the spectral flux of low frequencies of a signal experiencing the proximity effect would increase as distance decreases at a higher level. This can therefore be used as an indicator to the proximity effect.

4.2. Algorithm

The incoming signal is first low pass filtered at 2kHz as most musical signals have the majority of information

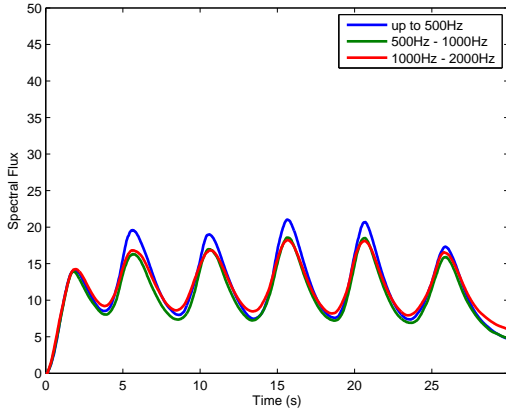


Fig. 2: Spectral flux of 3 bands of white noise recorded with an omnidirectional microphone.

below 2kHz [15]. The incoming signal is then taken into the frequency domain. The frequency bins are then split into a number of bands, i . The spectral flux for each band is calculated.

$$SF_i(n) = \sqrt{\sum_{k=p_i}^{Q_i-1} [X(n, k) - X(n-1, k)]^2} \quad (3)$$

where Q_i is the upper frequency bin limit of the i th band and p_i the lower limit of the i th band. This implementation uses 10 bands. The bands are then split into 2 sets at 700Hz to encompass all bands which may be effected by the proximity effect. In the ideal case of white noise recorded with an omnidirectional microphone the spectral flux will be equal for all frequency bands as all frequencies will exhibit an equal increase in amplitude as distance decreases. Figure 2 shows the spectral flux for 3 bands over time as the distance between a source and an omnidirectional microphone is changed. Figure 3 shows the same for a cardioid microphone.

In the cardioid microphone case with a white noise input the bands above 700Hz behave similarly to the omnidirectional microphone. Below 700Hz, bands will have greater spectral flux over time as the distance decreases due to the proximity effect. This can therefore be used as a measure for detection.

An average of the spectral flux of each set is then taken. The difference between the averages of the 2 sets of

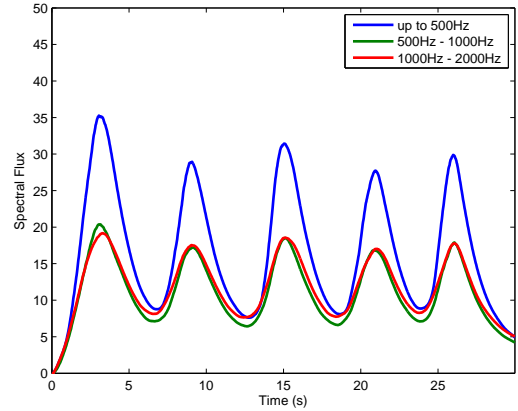


Fig. 3: Spectral flux of 3 bands of white noise recorded with a cardioid microphone

bands will indicate the presence of the proximity effect. The proximity effect is detected as follows

$$m = \begin{cases} 1 & \text{if } m \geq T, \\ 0 & \text{if } m < T. \end{cases}$$

where 1 indicates the detection of the proximity effect, T is the spectral flux difference threshold and m is the spectral flux difference $m = SR_L - SR_H$ where SR_L is the averaged low frequency spectral flux and SR_H is the averaged high frequency spectral flux.

5. EXPERIMENTATION

The detection algorithm was tested by recording different input signals with an omnidirectional and cardioid microphone. The distance between the source and microphone was changed over time.

The following results show the proximity effect detection algorithm on a number of input signals. Figure 4 shows the output of the proximity effect detector on the omnidirectional microphone recording with a white noise input and Figure 5 shows the same for the cardioid microphone. The detector outputs 1 if the proximity effect is detected and 0 if it is not detected.

The root mean squared (RMS) level of the input signal is shown in each case to indicate when the source moves towards the microphone. The amplitude from the source is kept constant and the output amplitude changes only

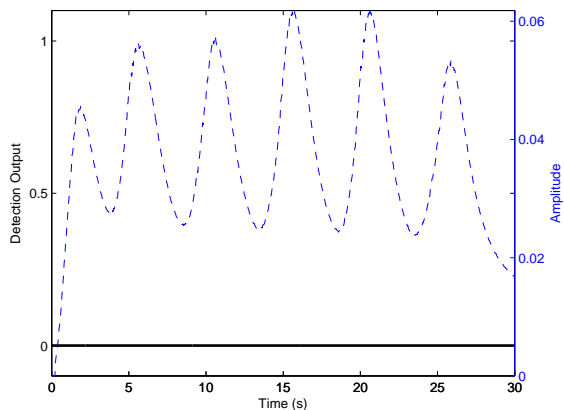


Fig. 4: Proximity effect detection of a white noise signal recorded with an omnidirectional microphone.

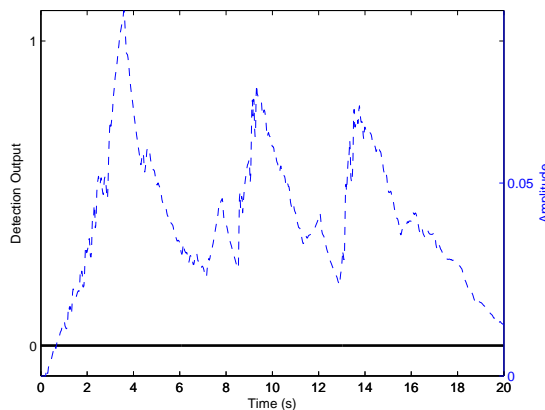


Fig. 6: Proximity effect detection of a vocal signal recorded with an omnidirectional microphone.

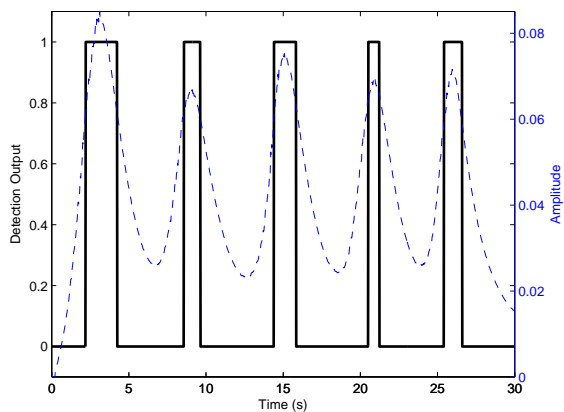


Fig. 5: Proximity effect detection of a white noise signal recorded with a cardioid microphone.

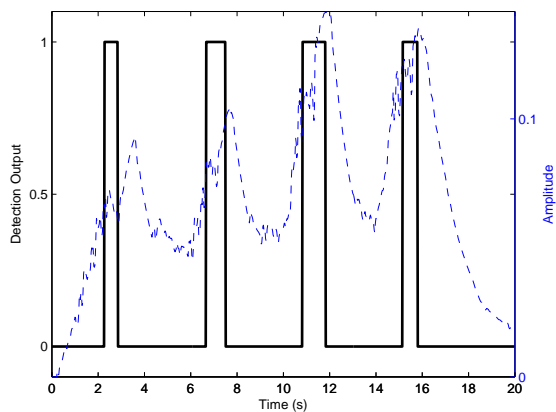


Fig. 7: Proximity effect detection of a vocal signal recorded with a cardioid microphone.

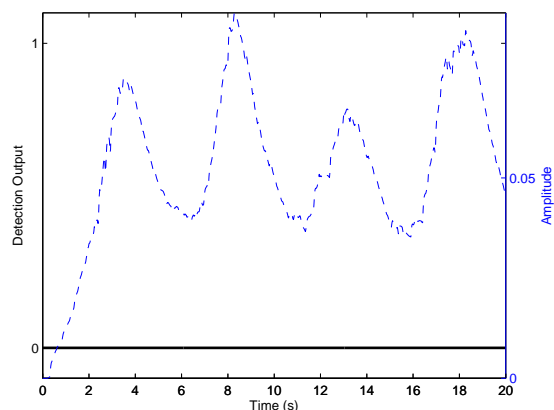


Fig. 8: Proximity effect detection of an electric guitar signal recorded with an omnidirectional microphone.

due to the distance between the source and microphone. The omnidirectional microphone picks up all frequencies equally with distance. Therefore no proximity effect is detected. The amplitude simply increases linearly over all frequencies due to less attenuation in air. The proximity effect on the cardioid microphone recording is accurately detected.

Figures 6 and 7 show the proximity detection output for a male vocal input signal with an omnidirectional and cardioid microphone respectively. The algorithm successfully detects when the source to microphone distance decreases and causes the proximity effect in the cardioid microphone case. The proximity effect is not detected in the omnidirectional microphone case.

Figures 8 and 9 show the output of omnidirectional and cardioid microphones with an electric guitar input. The proximity effect is successfully detected on the cardioid output and not detected on the omnidirectional output.

6. CONCLUSION

A method has been proposed for the detection of the proximity effect on microphone recordings. The proposed method uses spectral flux to measure the change in spectrum over time. The method is shown to accurately detect the proximity effect on recordings made with a cardioid microphone and equally to not detect the proximity effect in recordings made with an omnidirectional microphone.

In future work the detection method will be used to im-

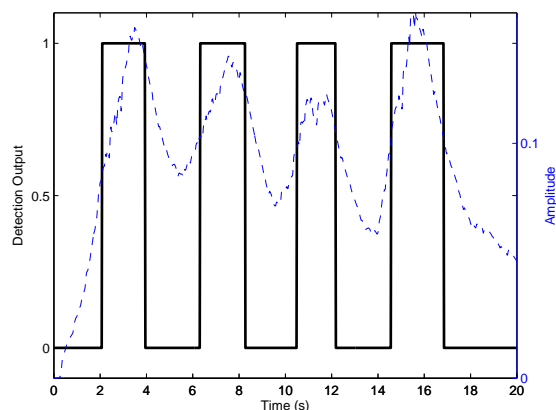


Fig. 9: Proximity effect detection of an electric guitar signal recorded with a cardioid microphone.

plement a correction algorithm which will reduce the proximity effect. The detection algorithm will be developed to improve the accuracy on input signals which have greater fluctuating frequency content such as an instrument playing a large range of notes.

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