LIVE MULTIBAND SPEAKER DELAY COMPENSATION

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ABSTRACT

This sales pitch will be delving into the technicality's and practicality's of a multiband delay compensation software package for outdoor live sound reinforcement which incorporates an automatic delay time calculator per frequency band using Generalised Cross-Correlation and Phase Transform.

Although this code has been implemented in MATLAB, the coding will not be analysed for the purpose of this sales pitch.

This software would be run in MATLAB using an external sound card with at least 4 inputs and two outputs, with hopes that with the correct funding it shall be implemented as a stand-alone rack mounted DSP unit.

1. INTRODUCTION

When multiple sources of sound are being produced and being played from multiple distance there will be an obvious time difference between the two sources, this is due to the speed of sound. This phenomenon is most apparent in larger live music situations where there is one set of speakers at front of house, then a second set of speaker's half way through the venue in order to get an equal dispersion of energy throughout the entire venue.

If there is not correct delay compensation applied to the second set of speakers this would create a perceivable two signals and in some cases cause comb filter, in order to correct this a certain delay is applied that corresponds to the time from the first set of speakers to the second which then would create an equal listening experience.

This can be calculated manually by using the inverse square law as an approximation (as its highly doubtful any venue would be a pure acoustical free field).

For instance, if the sound pressure level is 83dB from 1m of the sound source and the second sound source is 111m away, the time delay can be calculated by $\frac{Distance}{c}$ thus, the time delay being 323ms (with c being 343m/s) then using $dB\ Loss = 20log\frac{L1}{L2}$ it can be calculated that the amount of sound pressure lost is 40.90dB, therefore the total sound pressure level at 111m is now 42.1dB.

Unfortunately, this is purely theoretical and many external factors can effect these levels and times.

Although speed of sound is independent to frequency, certain factors may cause attenuation on different frequency bands, such as easier absorption of high frequencies from crowds and other objects, and also the effect of diffraction.

The multiband delay processor has been designed to have one microphone by the first speaker source to record a sine sweep then split the recorded information into predetermined frequency bands gathering information such as sound pressure level per frequency band, then the 2nd microphone set up by the second speaker source, this microphone then will determine difference in time between the first source and the second per frequency band then also the difference in sound pressure level per frequency band using generalised cross-correlation. These differences then will be automatically inputted into the system and it will correct the delay time and amplitude per frequency band in real time, this can be seen in Appendix A

2. GENERALISED CROSS-CORRELATION

Generalised Cross-Correlation and Phase Transform (GCC- PHAT) is a type of mathematics that is used to approximate the delay time between two different signals. Correlation takes two signals to produce a third signal (the third signal being the time delay in this case). In order to do so the signals are needed to be transported out of the time domain and into the frequency domain using the FFT, then they are recombined using GCC, then lastly the Phase Transfer is applied and the data displayed as a histogram (using the inverse FFT).

A single sound source being reproduced by two microphones can be shown as such

$$x_1[n] = s(t)[n\text{-}t_1]$$
$$x_2[n] = s(t)[n\text{-}t_2]$$

where the delay can be described by

$$t_s = t_2 - t_1$$

Where x_1 and x_2 are the microphone positions, t_1 and t_2 are the time traveled, s is the source and d_1 and d_2 are the distances. This can be seen in Fig.1.

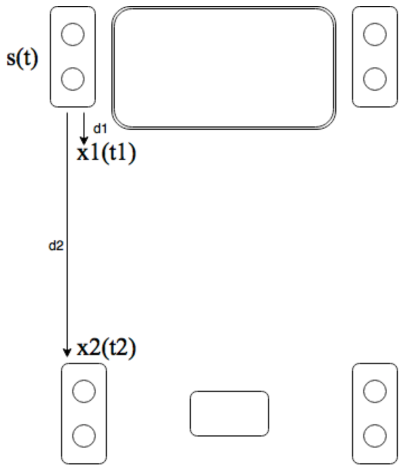

Fig.1

The GCC can be shown as

$$\Psi G[k] = X_1{}^*[k] \cdot X_2[k]$$

As this is in the frequency domain x_1 is X_1, Psi is the wave function, K is the frequency bin. The inverse FFT, where F^{-1} is the inverse FFT function can then be applied shown as such

$$\psi G[n] = F^{-1}\{\Psi G[k]\}$$

The Phase Transform can be applied prior to the inverse FFT to increase performance. This weights the GGC so all of the frequencies will be normalized, keeping the phase intact.

$$\Psi p[k] = \frac{\Psi G[k]}{\Psi G[k]}$$

Thus, to get the total time difference it can be shown as such where *t* is the time delay difference

t =arg max ψP[n]

This time delay worked out by GCC-PHAT and can now be implemented into the multiband delay configuration for live sound speaker delay compensation.

3. MULTIBAND DELAY

For a multiband filter arrangement for this application there are a few requirements for overall speed and reliability that need to be addressed in order to succeed with the devolvement.

As mentioned previously the concept is for the calibrated microphone at the first speaker position to split the broadband speaker spectrum into three bands then cross correlate these signals with the second microphone position which has also been multiband split to determine the individual frequency band attenuation as well as the time lag between the two locations, then using this information to compensate the lag on the certain frequency's for the second speaker arrangement.
For the live delay compensation, a three fixed frequency multiband delay will be needed, the frequency band will be split with cut offs being placed at 400Hz for the low pass and 4000hz for the band pass.

3.1 MULTIBAND FILTERS

An ideal multiband filter would consist of a high order roll off in the pass band between cross overs to minimise the attenuation in the stop band, and a flat frequency response to accurately reproduce the signal transparently without additional artifacts. Although these aspects are important for a healthy filter, the most vital is the speed in which these filters can process as this application will be used in real time.

For these reasons an Infinite Impulse Response (IIR) filter has been chosen over the FIR. The IIR filters work by weighting the past inputs with a combination of the past outputs, thus creating a recursion(Fig.2).

$$y[n] = \frac{1}{a_0}\left(\sum_{i=0}^{N} b_i x[n-i] - \sum_{i=0}^{N} a_j y[n-j]\right)$$

Fig.2. Transfer function of the IIR filter

Unlike FIR filters which use convolution, the IIR filters are carried out by recursion, thus increasing the processing speed.
Under these certain criteria the IIR filter 'Chebyshev Type II' with a high order roll off Is chosen, the Chebyshev Type II incorporates a equiripple stop band to "minimizes the absolute difference between the ideal and actual frequency response over the entire stopband" (Mathworks,2005) whilst retaining a monotonic passband.

3.2 DELAY

The purpose of the delay is take the stereo output from the front of house mixer and delay it accordingly by the previously calculated cross correlation times and by the level difference amount. A delay is simply said to take the input data, store this data into a medium (in this case a computer running MATLAB) then play this data back at a predetermined length of time (the cross correlation calculation). A source with a delay of two sampling periods can be viewed as such.

$$y(n) = x(n\text{-}2)$$

As IIR filters are less computationally intensive for real time operations a IIR comb filter will be used for the basis of the delay system.

4. IMPLEMENTATION AND USAGE

To first use this system a self-delay time calibration must be used in order to determine the total delay of the system, this is done by using a simple subtraction between the input and the output of the sound interface. A 10 second white noise burst will be recorded from the output into the input, then the time difference will be subtracted. This will output a time lag value which will be the total internal delay time that is to be subtracted from the overall delay.

Further, the two microphones need to be calibrated in order to get accurate readings for both sound pressure level and frequency. This is done by recording a 1khz tone at 94 dB into both of the microphones into the software via an acoustic microphone calibrator, as the sound pressure and reference frequency are now known the calibration offset will be applied into this software.

The physical configuration and placement of this system is as such;
The MATLAB run software, computer and a 4 in soundcard or the 'Brain' will be situated where the mixing desk is located, one microphone will be placed at the first front of house speakers then a XLR cable run into the 1st input of the soundcard, a microphone is then placed at the 2nd distant speaker distance with a XLR cable run into the 2nd soundcard channel. A stereo FOH send from the mixing desk is then attached into channel 3 and 4 of the sound card with a stereo output from the soundcard being sent to the power amps that power the speakers at distance two.

An exponential sweep of 30 seconds from 10hz to 20khz will be played from the first front of house speakers and then recorded from both the calibrated microphones at both the distances, the multiband delay will then be applied to both the recorded signals and split into three different frequency bands.
From this recorded multiband split the average sound pressure level for each of the frequency bands at both the microphone positions are displayed, the sound pressure level difference can then be calculated, this will be the total attenuation between speaker position 1 and distance speaker position 2 for each of the three frequency bands.

Using the same sweep recording GCC-PHAT can then be implemented to calculate the time lag between the two speaker distance per frequency band. The internal system delay calculated previously can now be subtracted from the total delay per frequency band.

The software now has the time delay and a sound pressure level difference between two distances, the system is now calibrated and ready to apply these calculations to a real time live signal.

The mixing desks stereo outputs can now be inputted into the MATLAB system in real time, these inputs are now split into the same frequency bands as previously used, with the delay previously calculated via GGC-PHAT being applied and then getting run through a gain stage to boost each individual frequency band by the difference calculated previously. This signal is then summated per left and right channel and sent to the distance speakers 2 for playback, thus creating a multiband live speaker delay compensation using GGC-PHAT.

5. DISCUSSION & CONCLUSION

This system currently runs through any computer system in which MATLAB and a soundcard can be used on, in order for this product to gain finical success it is believed that it should be implemented in a stand-alone rack mountable DSP unit, this could be done on a low cost Raspberry Pi unit running Linux and MATLAB.

6. REFERENCES

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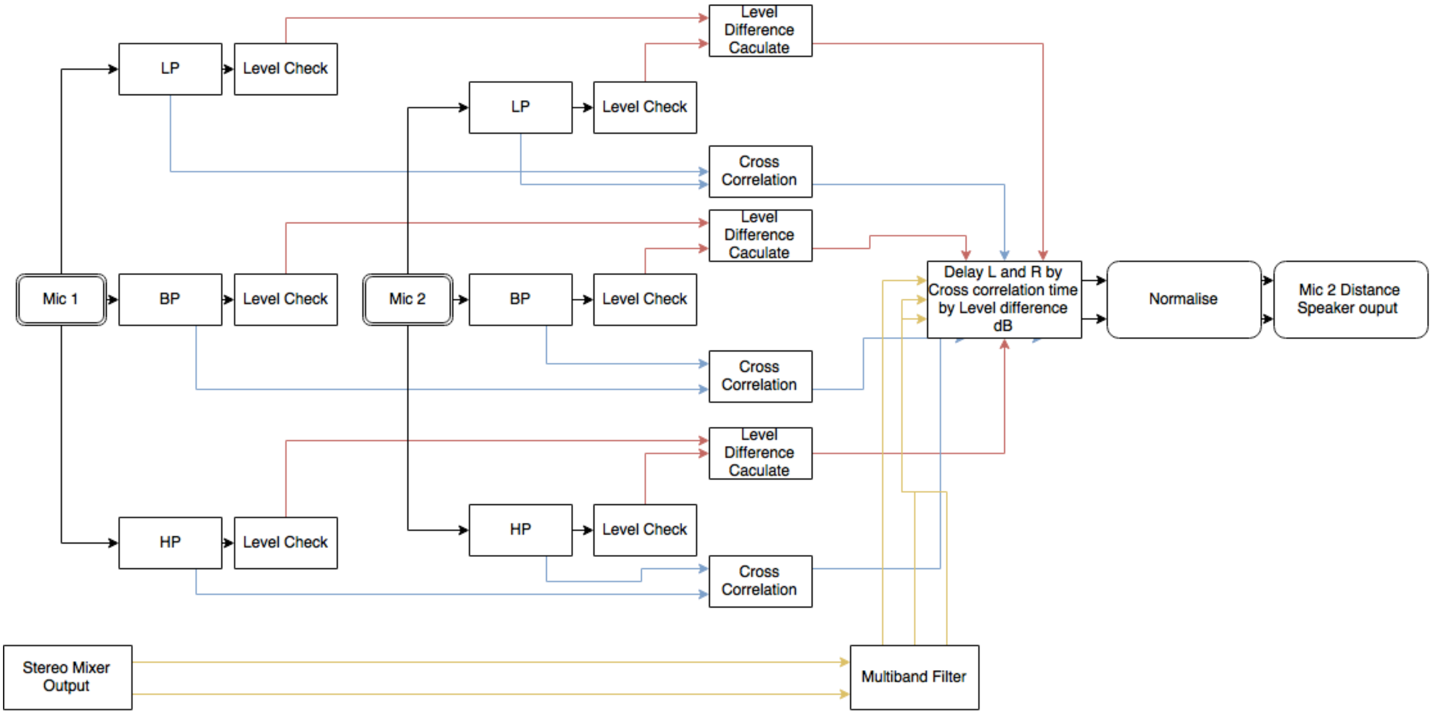
7. APPENDIX

Appendix A.


Level Difference Caculate
LP
Level Check
LP
Level Check
Cross Correlation
Level Difference Caculate
Mic 1
BP
Level Check
Mic 2
BP
Level Check
Delay L and R by Cross correlation time by Level difference dB
Normalise
Mic 2 Distance Speaker ouput
Cross Correlation
Level Difference Caculate
HP
Level Check
HP
Level Check
Cross Correlation
Stereo Mixer Output
Multiband Filter