Real Time Digital Signal Processing Speech Enhancement Project

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Abstract

This report describes the implementation and analysis of a dynamic real-time noise filtering system for a speech signal, done primarily through spectral subtraction of estimated noise. It will describe and analyse a series of techniques to improve the filter performance, and conclude with the the specific implementation chosen.

Declaration

Declaration: We confirm that this submission is our own work. In it, we give references and citations whenever we refer to or use the published, or unpublished, work of others. We are aware that this course is bound by penalties as set out in the College examination offences policy.

Signed: Yong Wen Chua & Ryan Savitski

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1 Introduction

In this digital age, more communication is now happening over networks rather than in person. Driven primarily by telephony, this increase in prevalence of digital communication has made the transmission of clear and noise-free sound more important than ever before. Implementations of such a system is complicated by the fact that the implementers can never fully know what sort of noise environment users are in when they are speaking through the digital communication system. In addition, such a noise reduction system have to run in real-time, or users would not be able to communicate in real-time.

This project aims to implement such a noise-reduction system on a set of noise corrupted sound files in real-time. Various heuristics are used to estimate the noise profile. Once estimated, spectral subtraction is used to "subtract" the noise spectrum estimate to obtain a clearer version of the corrupted speech signal.

In section §2, a basic implementation of the noise filter is explored. This will also establish the basic framework for the enhancements explored in section §3, along with details of the parameters used in those enhancements. In section §4, the combination of the enhancements chosen to be used for the final filter version are described. Finally, further ideas and heuristics that could enhance the filter are discussed in section §**??**. The code for the project can be found in section §A.

2 Basic Implementation

Filtering of the noisy input signal is done primarily in the frequency domain. This process is illustrated in figure 2.1. The Fast Fourier Transform (DFT) algorithm is employed to perform a Discrete Fourier Transform (DFT) on the input signal. Then assuming that the noise is additive, the noise spectrum is estimated and subtracted from the noisy signal. Finally, an inverse DFT (IDFT) is performed on the input before sending it to the output.

Due to the real-time requirements of the system, it is not plausible to take a DFT of the entire noisy input. Thus, frames are passed through DFT and then processed accordingly. In order to prevent discontinuities at frame boundaries (which manifest as clicking sounds heard in the output), frames are processed in an overlapping manner, with appropriate windowing done on the frames so that the overall gain at the frame boundaries add up to unity.

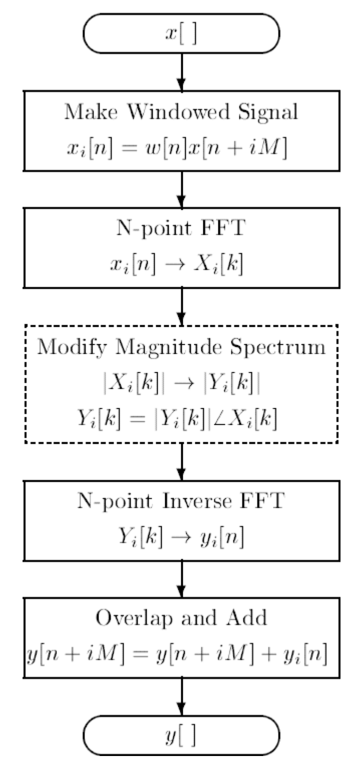


Figure 2.1: Block diagrams of how the samples are processed. (Compernolle 1992)

2.1 Frame Processing Implementation

A frame size of 256 samples was chosen for frame processing, and an over-sampling ratio of four was chosen along the the square root of the Hamming Window. This ensures that there are no sharp discontinuities at frame edges and that the overall gain of the overlapped windows sum to unity. This means that a full frame is taken for frequency domain processing every $1/4$ of a frame, known as a frame segment. Each segment is thus 64 samples long. The processing is illustrated in figure 2.2.

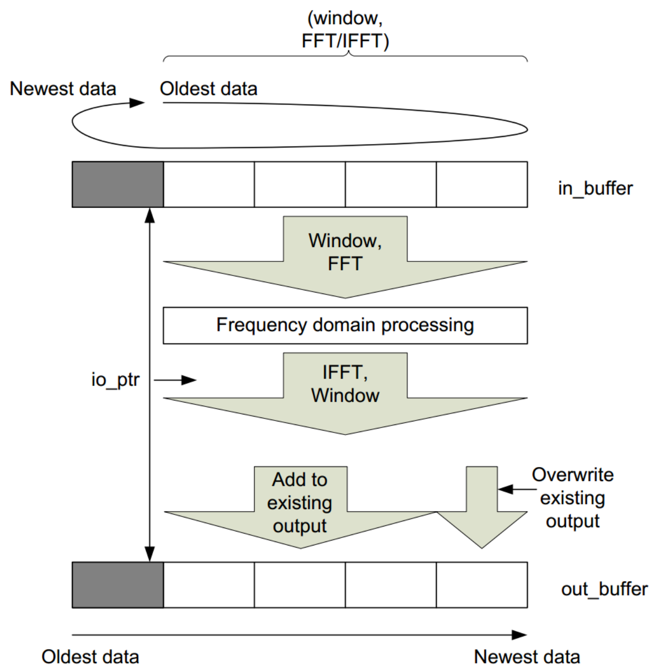


Figure 2.2: Implementation of the overlapped frame processing. (Mitcheson 2013)

The input and output buffer are implemented as circular buffers with five segments each. At any time, four of those input segments will be fully filled (and can be used for frame processing) with one being used to store the inputs coming into the system. At the same time, one of the output segments will have been fully added with its overlapped frame (and therefore ready for output), while the other four frames are being added with the current processed frame. The inputs are read from the input port via an interrupt, and the relevant output is then sent to the output port. This results in a 1.25 frame latency from the input port to the output port.

2.2 Noise Spectrum Estimation

The heuristics used in the estimation of noise assumes that within a 10 second window, the speaker would have at least paused for a fraction of the time. During this time, the spectrum of the input would simply correspond to that of the noise spectrum. Thus, the minimum of the spectrum content over a ten-second sliding window can be used to estimate the noise spectrum. Due to the fact that the phase of the noise spectrum cannot be known, and the fact that slight phase distortion is not audible to the human ears, only the absolute of the spectrum is analysed and processed.

In practice, however, it is not feasible to store and minimise over ten seconds worth of spectral content. Instead, the noise buffer is split into four, each holding a minimum over a 2.5 second period. These four separate noise minimum buffers are implemented as a single circular buffer, with the appropriate program logic to separate the segments. For the current window $M_i(\omega)$ (where $i \in [1,4]$) and an input spectrum of $X(\omega)$, $M_i(\omega) = \min\left[|X(\omega)|, M_i(\omega)\right]$.

When the noise spectrum is to be estimated for use in spectral subtraction, the minimum over the four 2.5 seconds segments are used. Noting that the nature of the naive noise estimation always underestimates the noise content, therefore an over-subtraction factor α is used for compensation. Thus, the minimum estimated noise spectrum $N(\omega)$ is given by

$$N(\omega) = \alpha \min_{i \in [1,4]} \left[M_i(\omega)\right] \tag{2.1}$$

where α was set to a value of 20 for the naive implementation, producing a very aggressive filter. Although this is higher than the suggested values by Berouti et al. (1979), it works decently in practice. However, this results in high amplitude musical noise and some voice distortion. Lowering α would let more noise through the filter, but at the same time reduce the amount of musical noise (see section 2.3.1) present.

2.3 Noise Spectrum Subtraction

Let the output be $Y(\omega)$, and the input and noise be $X(\omega)$ and $N(\omega)$ respectively. To preserve the phase of $X(\omega)$ (and achieve a zero-phase filtering), only the magnitude should be altered, as mentioned before. Thus, the output is given by

$$|Y(\omega)| = |X(\omega)| - |N(\omega)| = |X(\omega)|\left(1 - \frac{|N(\omega)}{|X(\omega)|}\right) \tag{2.2}$$

$$Y(\omega) = X(\omega)G(\omega) \tag{2.3}$$

$$G(\omega) = \max\left(\lambda, 1 - \frac{|N(\omega)|}{|X(\omega)|}\right) \tag{2.4}$$

where $G(\omega)$ is a gain factor that is calculated from the estimated noise spectrum derived from equation (2.2). Because of the fact that the expression $1 - \frac{|N(\omega)|}{|X(\omega)|}$ can potentially go negative (i.e. the noise is over-estimated), $G(\omega)$ should be lower bounded. The input is not simply attenuated to zero (but to the noise floor λ) to reduce the amount of perceivable "musical noise".

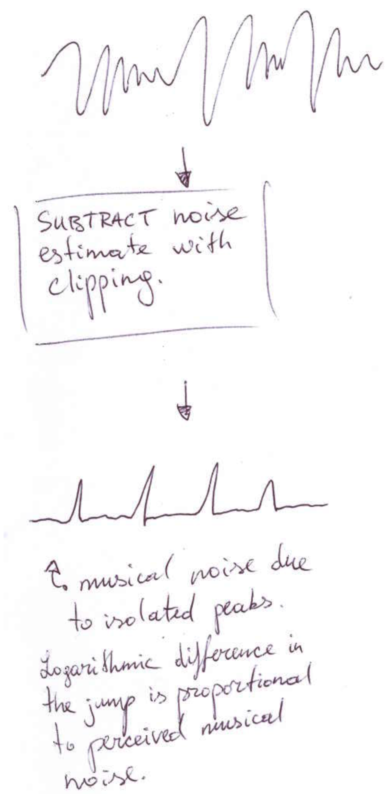
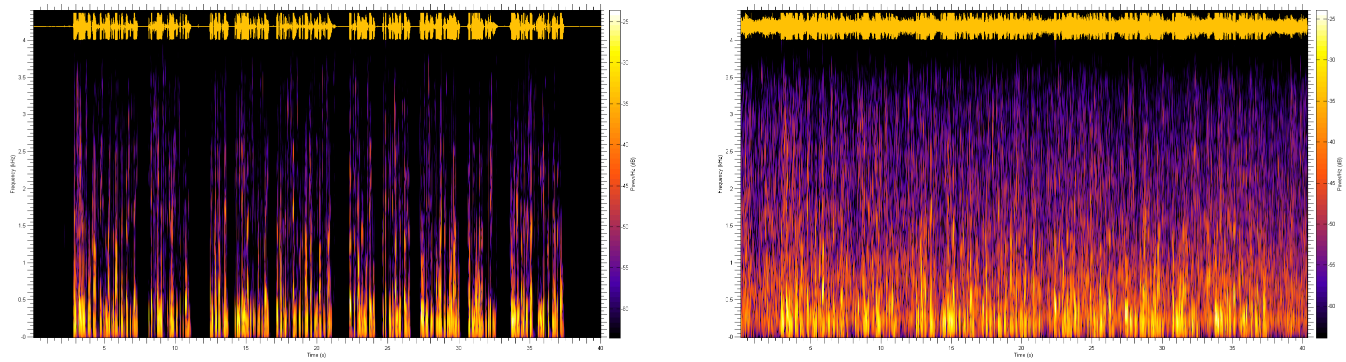


Figure 2.3: Illustration of the generation of musical noise by noise clipping.

2.3.1 Musical Noise

When subtracting an estimate of the noise with clipping, imperfect noise estimation produces isolated peaks in the spectrum of the filtered signal, which is perceived as low-tone musical notes. This is illustrated in figure 2.3. As human sound perception is logarithmic (Compernolle 1992), the difference between the zeroed floor and a sharp peak is very pronounced. Instead, by clipping to a small positive value λ, the differential step between the post-filter noise floor (λ) and the spectral peaks that are a result of imperfect noise estimation can be reduced. In this basic implementation, the spectral noise floor constant λ was chosen to be 0.05 by empirical methods, choosing from the ranges suggested by Berouti et al. (1979). Generally, a higher λ would result in a higher white noise level, but lower perceived musical noise. Lower λ results in a lower white noise output but a more pronounced musical noise effect.

2.4 Evaluation



(a) Spectrogram of the clean file (b) Spectrogram of "factory2" file

Figure 2.4: Spectrograms of provided files, generated in Matlab

As human perception of volume, noise and in general audio signal clarity is very subjective and lacks a complete mathematical model, spectrograms are used to introduce some form of objectivity when evaluating the performance of the filterThe "factory2" corrupted sample is used for evaluation of the filter. Figure 2.4a shows the spectrogram of the provided clean sound file, and figure 2.4b shows the spectrogram of the "factory2" corrupted signal.

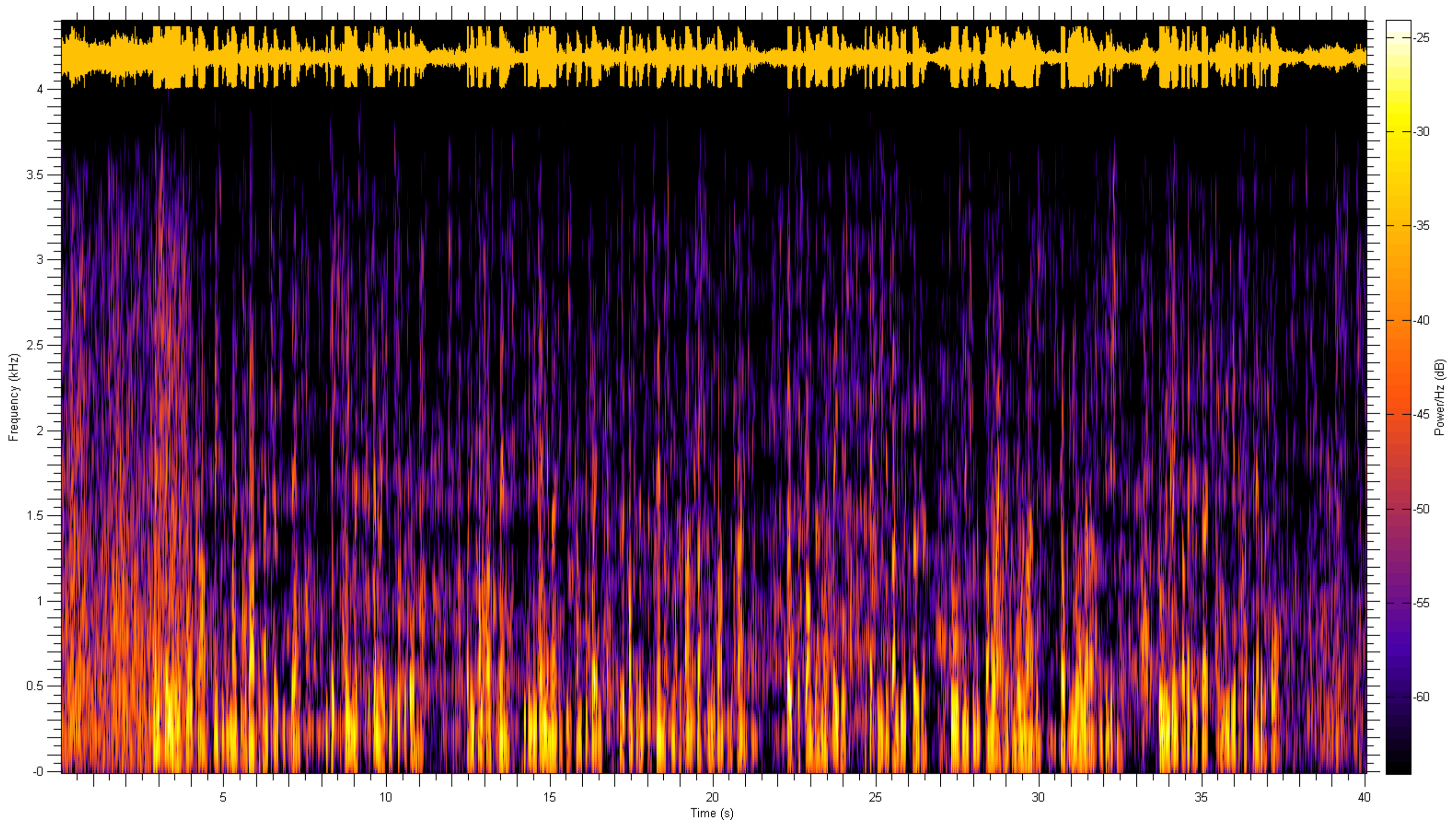


Figure 2.5: Spectrogram of the basic filtering of "factory2" at 8000 Hz sampling frequency.

After applying the basic filter, the results can be seen in figure 2.5. From listening tests, the basic implementation has managed to attenuate the background noise slightly, with darker areas seen in figure 2.5 in between speech. The filter is slow in responding to background noise changes, especially when the noise level transits from a low level to a high level.

This is due to the ten-second window used for noise estimation. As the filter is always pessimistic about the noise level, the lower estimate before the transition is always dominating until the relevant minimum noise buffer is discarded (around 7.5 seconds). Due to the high over-subtraction factor α introduced in equation (2.1) to correct the underestimation of noise, some of the speech is also removed and results in the voice being slurry. There is also a very pronounced level of musical noise.

The unnatural sounding musical noise is a result of the magnitude of the frequency exhibiting strong fluctuations in the noisy area (Cappe 1994). Referring to figure 2.4a, it can be seen that the speech is paused between 11-12.5 s and the spectrogram is completely empty during that period. In the filtered version in figure 2.5, it can be seen that there is noise during this period of time, and the magnitude of the frequency bins fluctuates very rapidly, exhibiting the musical noise effect mentioned above (see section 2.3.1).

3 Enhancements

Various enhancements are implemented, and evaluated for their effectiveness in removing the noise from the signal. Some of these enhancements improve the noise removal, while some exacerbate the noise. Each are thus evaluated in turn for their effectiveness, and a final combination of these enhancements is described in section §4.

3.1 Structural Optimisation

The frequency spectrum processing can be sped up by slightly less than a factor of two simply by only processing the first half, plus one, of the frequency bins in the DFT. This is because the inputs to the DFT are real, and thus the DFT of the inputs will be complex conjugates around the middle. For our frame size of 256, where X_k denotes the k^{th} frequency bin, $X_{256-i} = X_{-i} = X_i^*$ for $i \in [1, 127]$, and X_0 and X_{128} are simply real-valued. This also allows some of the buffers used in the enhancements described below to be almost halved in size.

This helps to prevent "clicking" in the output, resulting from frame computation taking longer than one frame length.

3.2 Low-Pass Filter Input for Noise Estimation

A problem with the basic approach is that each frame's DFT is considered alone when updating the estimate of the minimum noise. Therefore, even a single frame with an instantaneous magnitude drop for an DFT bin will potentially stay as the overly pessimistic noise estimate for the duration of the noise buffer. Since the frame length for 256 samples at 8000 Hz sampling frequency is 32 ms, one particularly underestimated noise value will stay within the minimum noise buffer for a period that can last seconds. This is the reason for a high over-subtraction coefficient for the basic filter above, to compensate for the filter's optimism of the noise level. Furthermore, for the same reasons of the filter being overly pessimistic, any noise with small spectral pauses (that need not be aligned to one frame) will defeat the basic implementation.

3.2.1 Magnitude domain

One way of improving on the naive spectral subtractor is to low-pass filter (LPF) the DFT magnitude bins across time according to:

$$P_t(\omega) = (1-k) \times |X(\omega)| + k \times P_{t-1}(\omega) \tag{3.1}$$

$$k = e^{(-T/\tau)} \tag{3.2}$$

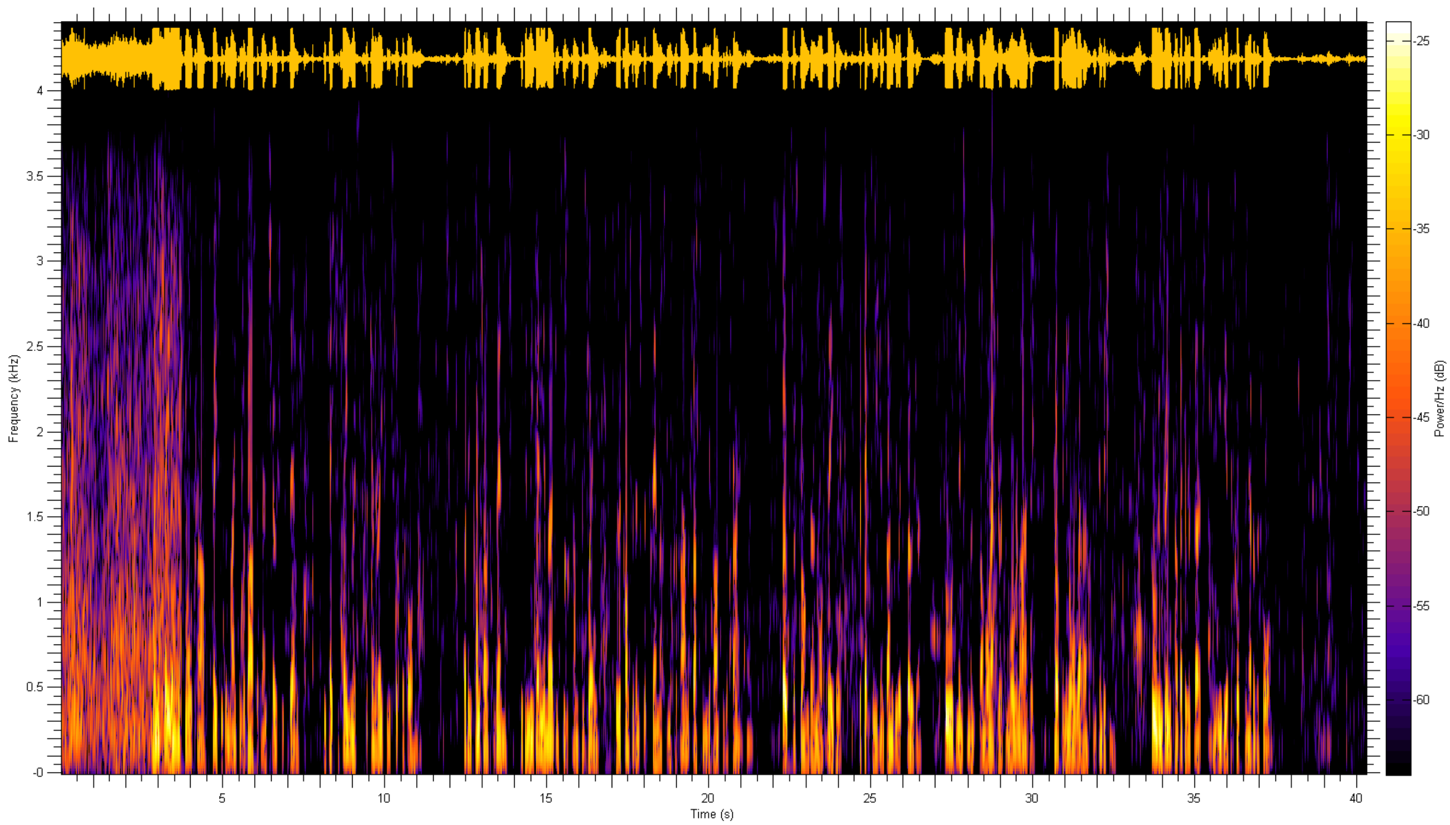


Figure 3.1: Spectrogram of the filtering with the low-pass filter input for spectral content across time

where P_t is the input estimate to be presented to the minimum noise buffer as per equation (2.1), T is the frame rate (the time between frame calculation), and τ is the time constant parameter for this filter.

$P_t(\omega)$ is the low pass filtered DFT bin that corresponds to the persistent average level of the magnitude at that frequency bin, with the smoothness of the filtering controlled by the time constant τ. By taking the minimum of the low pass filtered signal as the noise minimum (equation (2.1)), the filter will not suffer from the degenerating upon encountering sudden magnitude jumps in the noise signal's spectral content. Result is such that the noise buffer's noise estimate now more closely matches the persistent noise level. This is illustrated in figure 3.2.

The over-subtraction coefficient, α, can thus be reduced from 20 to 3.2. This also reduces the musical noise perceived by lowering the level of fluctuation in the magnitude at each frequency bin between clipped and unclipped samples. This is due to the subtracted spectrum being more closely matched to the actual noise level.

In the basic filter, when the input $|X(\omega)|$ falls sharply, this would cause a sharp fall in $|N(\omega)|$ and cause a discontunity in the output. Low pass filtering removes this discontinuity by introducing a smoother transition. This also has the effect of lowering the responsiveness of the filter, but is worth the trade-off due to much better noise estimation.

The low pass filter can also be seen as a system where the inputs are integrated over a period of time and then averaged. Therefore, by having a longer period to average over, the pauses between speech can be used to get a much more accurate average noise estimate. Figure 3.1 shows the output of the filter being more successful in filtering out the noise when compared to figure 2.5. This is because the filter is now less susceptible to problems caused by a momentary drop in the input. The amount of musical noise is also significantly reduced as well (for example in the pause in speech from 11-12.5s).

The time constant factor τ can be adjusted in this filter, and 50 ms was chosen for the implementation. This value was obtained empirically through blind listening tests with the set of samples given in the range of 20 to 80 ms. This range maximises the integration period of noise by being matched with the inter-word pause timings. Having a higher time constant results in a longer integration period, which when bigger than the pauses between words will start integrating voice spectrum as part of the noise estimate. A higher τ also means a slightly slower response to noise level change, although the 7.5 seconds period of the full noise buffer rotation latency has a bigger effect on the response speed (see section 2.2). A smaller τ on the other hand will be similar to the basic implementation through averaging over a shorter period.

3.2.2 Power domain

Alternatively, the LPF of the frequency bins can be done in power domain:

$$P_t(\omega) = \sqrt{(1-k) \times |X(\omega)|^2 + k \times [P_{t-1}(\omega)]^2} \tag{3.3}$$

Power domain filtering provides slightly more accurate filtering, because human hearing tracks the power of the signal closer than its magnitude. Hence, the perceivable noise could potentially be reduced.

An approach where the entire noise estimation and noise subtraction is performed in the power domain was explored, converting to magnitude domain only before output. Thus, equation (3.3) becomes

$$P_t(\omega) = (1-k) \times |X(\omega)|^2 + k \times P_{t-1}(\omega) \tag{3.4}$$

and noise subtraction originally in the form of equation (3.8) can be performed by

$$|Y(\omega)| = \sqrt{|X(\omega)|^2 - |N_p(\omega)|} \tag{3.5}$$

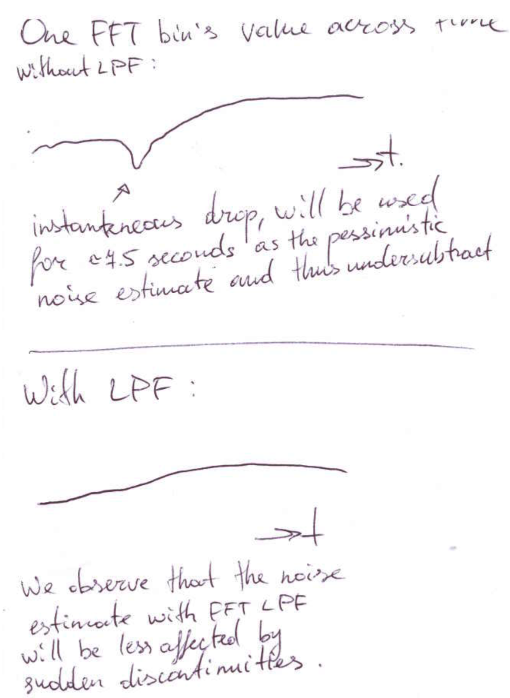


Figure 3.2: Illustration of the low pass filtering of the DFT bins.

where $|N_p(\omega)|$ is the minimum noise estimate in the power domain estimated in the same way as in equation (2.1) .

Testing showed no perceivable difference regardless of whether the filtering was done in the magnitude or power domain (including both variants), agreeing with Compernolle (1992).

3.3 Low-Pass Filter Noise Estimate

Another potential for discontinuities in the output is when the minimum noise buffers are rotated. These steps can be perceivable if the noise level changes significantly. They can be smoothed out by low pass filtering over the noise estimates. This ensures a smoother transition between noise buffer rotations. The filtering is done by

$$|N_{lpf}(w)| = (1-k_n) \times |N(w)| + k_n \times |N_{lpf}(w)| \tag{3.6}$$

$$k_n = e^{(-T/\tau_n)} \tag{3.7}$$

where τ_n is the time constant for this low pass filter. The time constant determines how smooth,and less responsive the transitions will be, with higher time constant giving a smoother transition.

Experimental testing showed that $\tau_n = 100\text{ms}$ gave the best performance for the samples given.

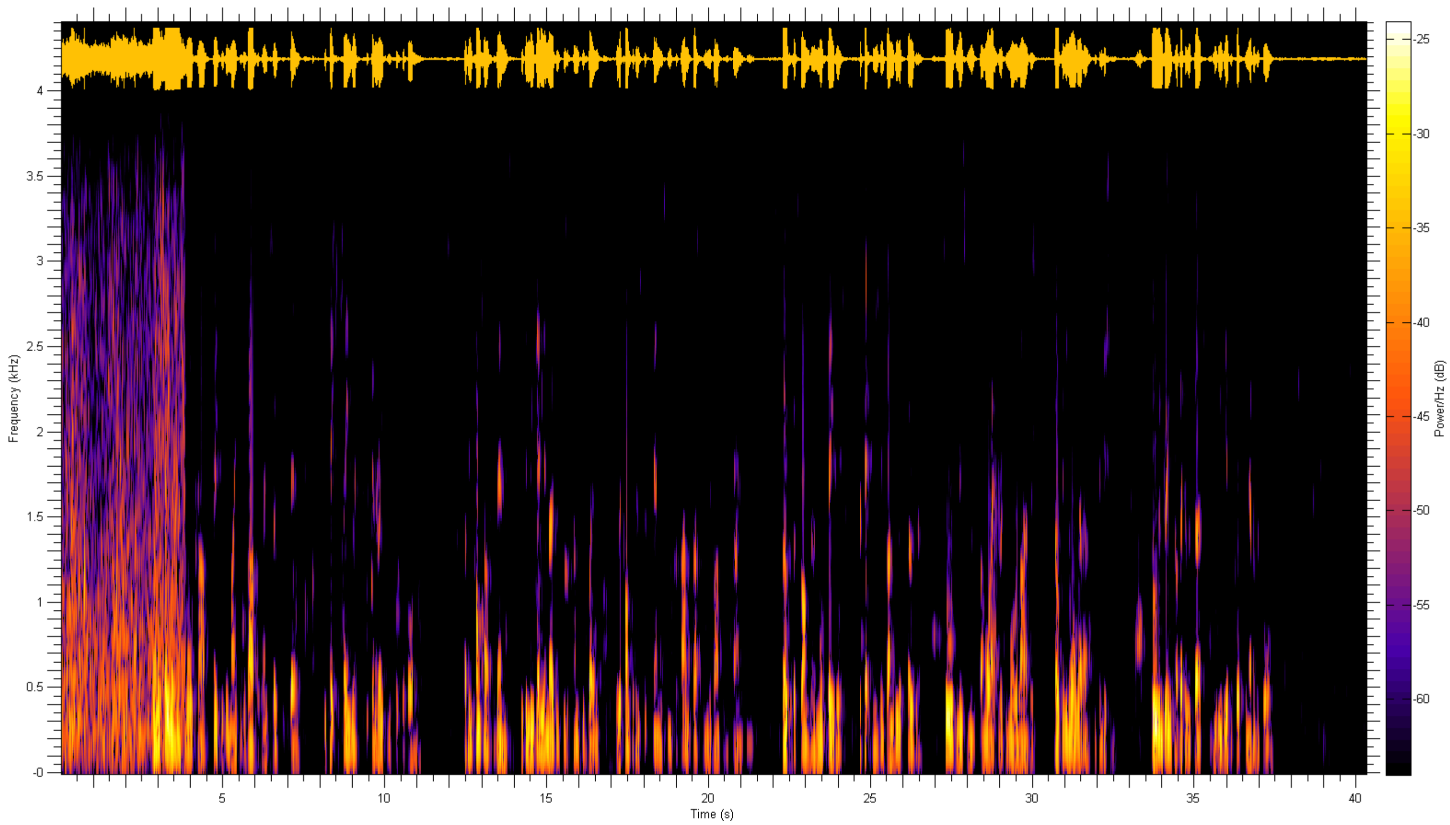


Figure 3.3: Spectrogram using an alternate calculation for $G(\omega)$, combined with input LPF for noise estimation

3.4 Alternate Calculation for $G(\omega)$

The nature of the noise estimation algorithm is such that a lower noise estimate (even momentarily) will cause an under-estimate of the noise spectrum. While this effect is moderated partly by the LPF in section 3.2, it still does not handle situations where the noise level increases. Consider that the factor $G(\omega)$ derived in equation (2.4) can be calculated using another form to perform noise subtraction form the input $X(\omega)$:

$$|Y(\omega)| = |X(\omega)| - |X(\omega)|\frac{|N(\omega)|}{|P(\omega)|} = |X(\omega)|\left(1 - \frac{|N(\omega)|}{|P(\omega)|}\right) \tag{3.8}$$

$$\therefore G(\omega) = \max\left[\lambda, 1 - \frac{|N(\omega)|}{|P(\omega)|}\right] \tag{3.9}$$

where $|N(\omega)|$ is the current minimum noise estimate over ten seconds from equation (2.1), and $|P(\omega)|$ is the low-pass filtered noise estimate for the current frame being processed from equation (3.1). equation (3.8) gives rise to different effects depending on the values of $|N(\omega)|$ and $|P(\omega)|$.

3.4.1 Noise Level Matches Minimum Estimate

Consider a situation when the value of $|N(\omega)|$ is equal, or slightly more than$|P(\omega)|$. This will signify that the current minim noise estimate over ten seconds roughly matches the low-pass filtered noise estimate for the current frame being processed. Thus, there is a high confidence that the input $X(\omega)$ consists entirely of only noise. Hence, the fraction $\frac{|N(\omega)|}{|P(\omega)|} \approx 1$ and the entirety of $X(\omega)$ will be subtracted.

3.4.2 Noise Level Increase

Consider that $|N(\omega)|$ remains constant at a previously encountered minima and that the noise level has increased via an increase in $|X(\omega)|$. Then, because of the LPF in section 3.2, $|P(\omega)|$ does not rise sharply, but slowly. Thus, $|P(\omega)|$ is slightly more than $|N(\omega)|$ and $\frac{|N(\omega)|}{|P(\omega)|}$ will have a value that is close to one. In the original form of noise subtraction in equation (2.2), $|X(\omega)|$ will continued to be subtracted by a very low $|N(\omega)|$ while the noise level has risen. This causes the increased noise to not be filtered, and can cause musical noise. In equation (3.8), this will result in a faster reduction in noise as $|X(\omega)|$ will be subtracted by a larger value. A numerical example of this happening is illustrated in table 3.1.

However, as time goes by and $P(\omega)$ increases, $\frac{|N(\omega)|}{|P(\omega)|} \to |N(\omega)|$ as $|P(\omega)| \to 1$. This means that, given sufficient time, equation (3.8) will perform as badly, or even worse as equation (2.2) in the event of a noise increase. Thus, the time constant τ in equation (3.1) can be increased to slow down this process so that eventually when the noise buffer described in section 2.2 rotates, the old value will be discarded. Alternatively, the time interval in which the minimum noise is estimated across can be reduced.

$\lvert N(\omega)\rvert = 0.1, \lvert X(\omega)\rvert = 0.9$	$\lvert X(\omega)\rvert - \lvert N(\omega)\rvert = 0.8$
$\lvert P(\omega)\rvert$	$\lvert X(\omega)\rvert - \lvert X(\omega)\rvert \frac{\lvert N(\omega)\rvert}{\lvert P(\omega)\rvert}$
0.2	0.45
0.3	0.6
0.4	0.675
0.5	0.72
0.6	0.75
0.7	0.771429
0.8	0.7875
0.9	0.8

Table 3.1: Numerical example depicting the noise subtraction during an increase in the noise level.

3.4.3 Evaluation

The caveat with this enhancement is that $|P(\omega)|$ is unable to distinguish speech from noise, and thus will cause some of the speech to be removed as well.

The output with this enhancement implemented can be seen in figure 3.3. When compared with the results from figure 3.1, it can be seen that more noise (and musical noise) has been removed. It can be noted that the noise in the period 11-12.5 s when the speaker has paused his speech is almost fully removed. However, it is also observed that some of the speech has also been removed, resulting in a slightly slurry voice.

3.5 Frame lengths

In general, frame lengths should be of powers of two so that the Fast Fourier Transform (FFT) used for DFT can be the most efficient. The frame length can be increased to give a better frequency resolution when doing the DFT of the samples. This would give a more granular approach in processing the frequency bins. However, this comes at the expense of temporal resolution, although it is not as important in this system. Due to the use of the frame processing structure described in section 2.1, there is a 1.25 frame delay between the input and output, and increasing the frame length will increase the time delay more. The increase in frame length will also cause more processing to be done per frame, and if the increased processing takes up more time than the frame time interval, it would cause "clicking" to be heard in the final output. The increase in frame length would also increase the heap memory used, and the limited memory would be a limiting factor.

Through experimentation, it was found that increasing the frame length did not yield much noticeable improvements. Reducing the frame length caused a degradation in the filter quality, as the frequency resolution would be reduced. It is noted that a frame length of 256 provides sufficient granularity for a sampling rate of 8 kHz.

3.6 Dynamic Over-subtraction

The over-subtraction factor α described in equation (2.1) can be dynamically adjusted based on the signal to noise ratio (SNR) given by $\frac{|X(\omega)|}{|N(\omega)|}$. When the SNR falls below a certain threshold, the over-subtraction factor α can be increased to cause more of the spectrum to be subtracted. It is known that in the frequency range of 0-50 Hz, no human speech is present. Thus, this frequency range is considered for increased over-subtraction. Alternatively, instead of over-subtracting and allowing these frequency bins to fall to the noise floor level, λ (see equation (2.4)), these frequency bins can simply be zeroed.

Through experimentation, with a low SNR threshold (approximately 2), no discernible difference could be heard and the spectrogram output did not reflect much improvement. When the SNR threshold was raised (to about 5), a degradation in the speech could be heard. This is because the noise level $|N(\omega)|$ is simply a minimum estimate and does not discern between noise and speech. When the 0-50 Hz frequency bins were simply zeroed, some of the low frequency musical noise was eliminated (e.g. in input "car"), and no degradation in speech was perceived.

3.7 Residual Musical Noise Reduction

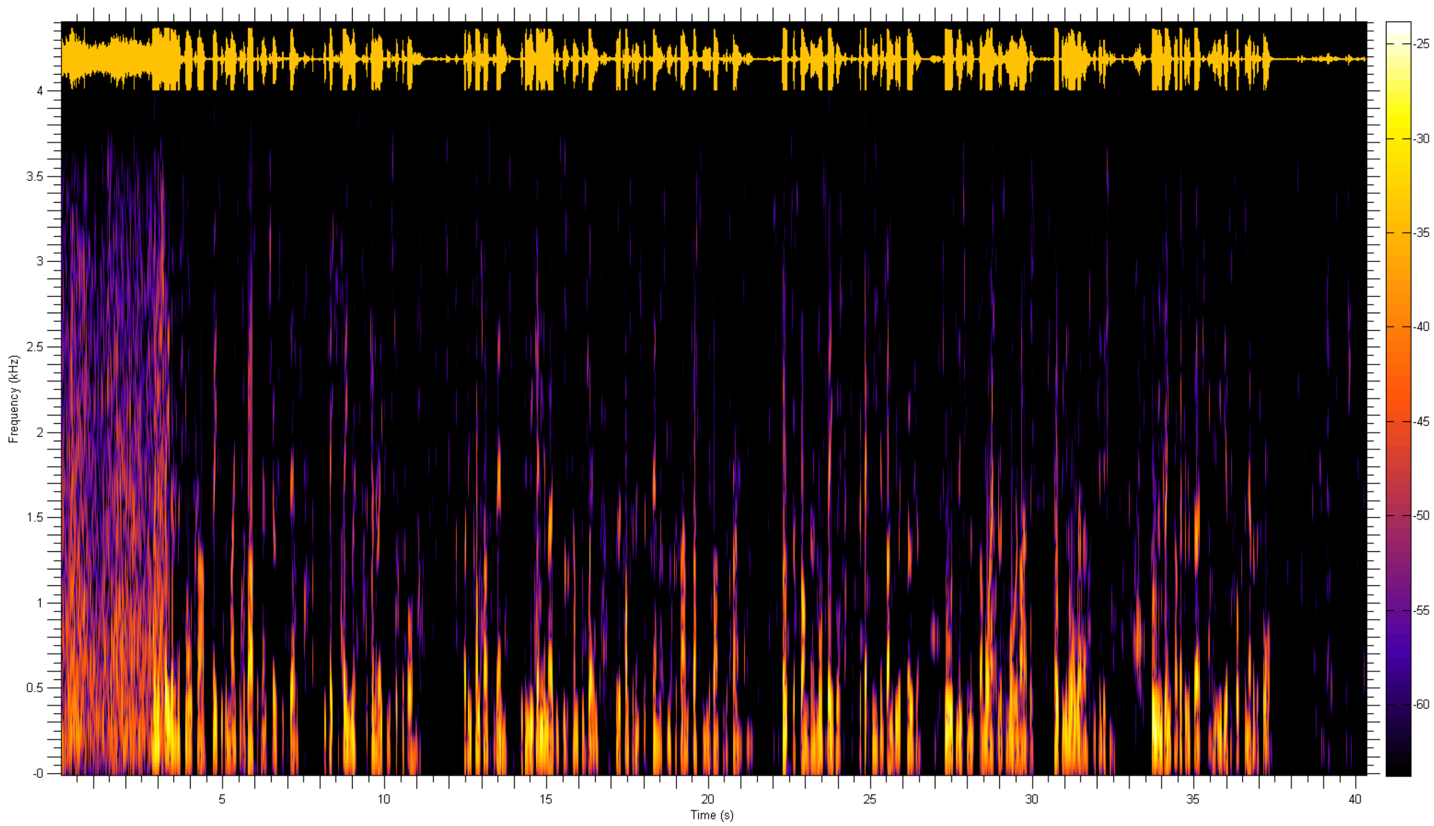


Figure 3.4: Spectrogram of the output with residual musical noise reduction.

The residual musical noise (see section 2.3.1) can be suppressed further. For a specific frequency bin, the musical noise is due to random fluctuation in its amplitude across different frames, therefore, it can be replaced with its minimum value from adjacent frames (Boll 1979).

For a current frame k, if the ratio $|N(\omega)|/|X(\omega)|$ is high, more noise is estimated than there is input. Then, take the output $|Y(\omega)| = \min_{i\in\{k-1,k,k+1\}} [|Y_i(\omega)|]$. The motivation for doing so is that if $|N(\omega)|/|X(\omega)|$ is high and $|X(\omega)|$ is rapidly fluctuating, then there is a high probability that it is simply noise, and can be minimised. If $|X(\omega)|$ is constant, it is likely to be low energy speech. Therefore, taking the minimum will retain the speech content.

It should be noted that this would require the low-pass filtering described in section 3.2 to be implemented. Otherwise, every time $|X(\omega)|$ goes lower than $|N(\omega)|$, $|N(\omega)|$ will simply be updated with this new value. The LPF smooths out the sharp drops, thus allowing $|N(\omega)|/|X(\omega)|$ to go above one.

Generally, the threshold for $|N(\omega)|/|X(\omega)|$ should be high so that rapidly fluctuating noise can be filtered out. However, if the threshold is too high, the minimisation might not kick in often enough because $|N(\omega)|$ would simply "catch up" with the lower value of $|X(\omega)|$. Increasing the time constant τ in equation (3.1) would allow the threshold to go higher. A lower threshold would cause less fluctuating noise to be removed.

The spectrogram of the output for a threshold of 3 can be seen in figure 3.4. Compared with the output from figure 3.1, it can be seen that more of the musical noise (c.f. 11-12.5s) has been removed.

3.8 Changing the Noise Estimation Period

The noise estimation window described in section 2.2 can be reduced to allow the noise filter to react faster to changes (especially increase) in noise level. This will also allow the alternate $G(\omega)$ calculation described in section 3.4 to perform better due to the faster rotation of the noise buffers. However, if the speaker does not pause at all during the period, his voice at its lowest level will be taken as noise and then erroneously subtracted from the spectrum as a result of estimating noise over a shorter time period.

By analysing the spectrogram of the clean version of the sound file (see figure 2.4a), it can be seen that the speaker does pause at least once every four seconds (however short). Therefore the total length of all the noise buffers combined contributes to 4 seconds in our implementation.

4 Final Implementation

In the final implementation of the filter, the enhancements described in sections 3.1, 3.2, 3.3, 3.4, 3.6, and 3.8 were implemented. The structural optimisation described in section 3.1 was implemented to improve the computational performance of the filter.

Quantity	Value
Over-subtraction factor, α	3.2
Noise floor, λ	0.05
Input LPF time constant, τ	0.05s
Noise Estimate LPF time constant, τ_n	0.1s

Table 4.1: Parameters for the final implementation.

The input low pass filtering enhancement described in section 3.2 gave the most significant improvement in terms of musical noise reduction. The alternate $G(\omega)$ calculation described in section 3.4 improves upon the LPF by addressing its shortcomings. This requires that τ be set to a slightly higher value. The noise estimation period (section3.8) was also reduced to four seconds to improve the performance of the alternate $|G(\omega)|$ calculation and increase the response rate of the filter. The frequency bins for 0-50 Hz were also zeroed, as described in section 3.6 because no human speech spectral content in that band is insignificant.

The enhancement described in section 3.5 was not implemented due to the adverse effects on the output. The enhancement described in section 3.7, while achieving good results, requires a longer τ to further improve the results already gotten from the other enhancements, and this will cause the noise estimate to be slower in reacting to changes.

The spectrogram of the output with enhancements combined can be found in figure 4.1, and the parameters used can be found in table 4.1. The filter has removed the musical noise present during perioids when speech is paused significantly (for example 11-12.5s) due to the filter being able to better react to changing noise levels. It has also reduced the amount of high frequency noise.

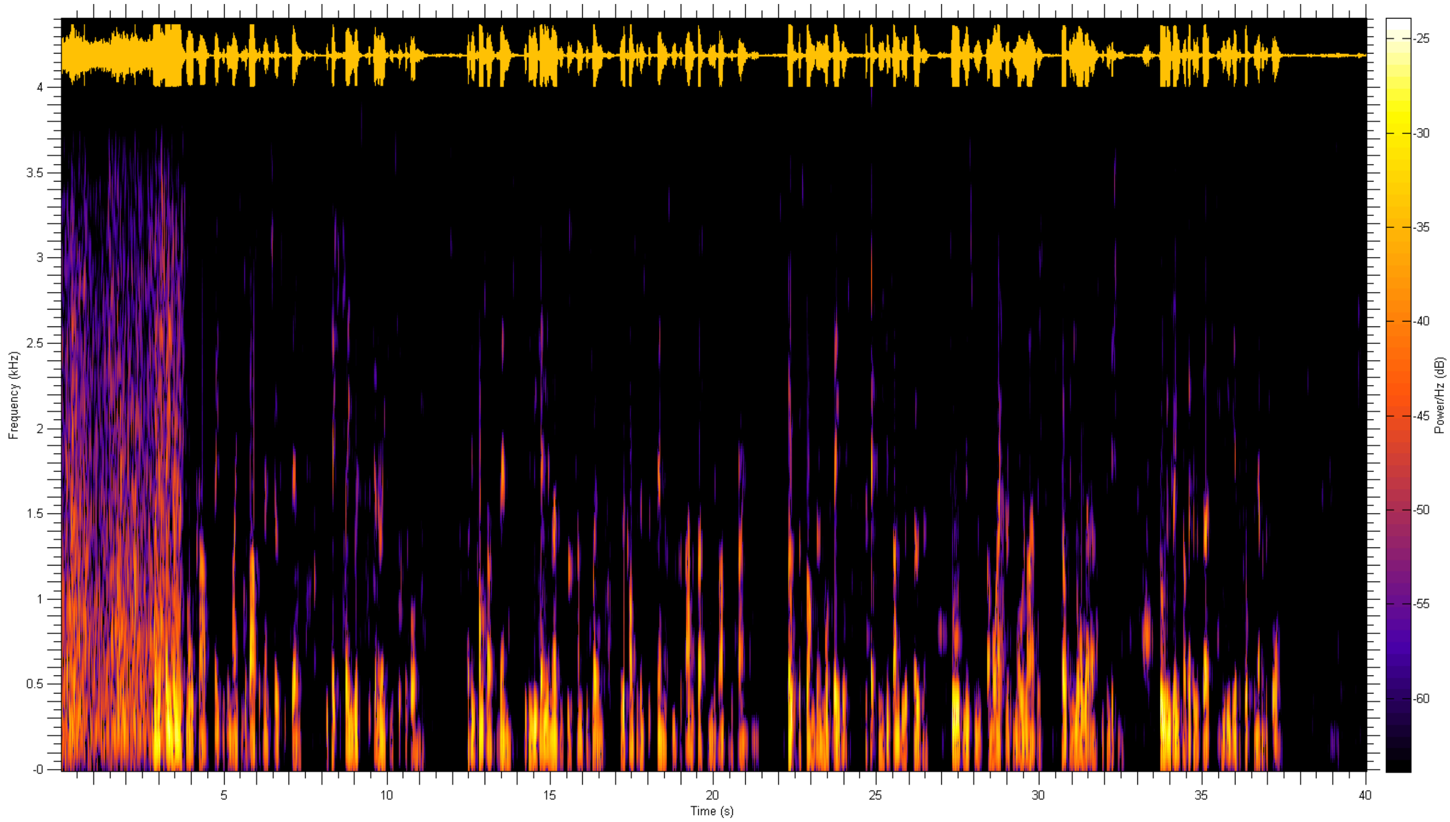


Figure 4.1: Spectrogram of the filtered signal for the final implementation.

5 Further ideas

A very appropriate application of spectral subtraction is mobile telephony due to its simplicity and low computational requirements. In such an application, when the connection for a call is initially established, the caller is not expected to immediately start speaking. As such, the first brief period after establishing a connection can be treated as being pure background noise. Noise can therefore be estimated before speech begins. This could be implemented by simply filling up the noise buffers with the initial samples and then continuing the estimation as before. This will remove the initial "warm up time" that the filter experience due to the way the noise buffers are set up in section2.2. Thus, the person on the other end of the line will receive a filtered output right from the beginning. Additionally, the filter could start tracking the noise even during the connection setup delay period.

Another trait of mobile telephony is that usually a device will usually only have one or very few associated users. Therefore adaptive filters that are trained to perform better based on the principal traits of the owner's speech can be implemented. This could be achieved in a variety of ways including neural networks, markovian models and Wiener filters (Compernolle 1992).

A Project Code

```c
/*************************** Pre-processor statements ******************************/
//  library required when using calloc
#include <stdlib.h>
//  Included so program can make use of DSP/BIOS configuration tool.
#include "dsp_bios_cfg.h"

/* The file dsk6713.h must be included in every program that uses the BSL.  This
   example also includes dsk6713_aic23.h because it uses the
   AIC23 codec module (audio interface). */
#include "dsk6713.h"
#include "dsk6713_aic23.h"

// math library (trig functions)
#include <math.h>

/* Some functions to help with Complex algebra and FFT. */
#include "cmplx.h"
#include "fft_functions.h"

// Some functions to help with writing/reading the audio ports when using interrupts.
#include <helper_functions_ISR.h>


// min/max macros
 #define min(a,b) \
   ({ __typeof__ (a) _a = (a); \
       __typeof__ (b) _b = (b); \
     _a < _b ? _a : _b; })

 #define max(a,b) \
   ({ __typeof__ (a) _a = (a); \
       __typeof__ (b) _b = (b); \
     _a > _b ? _a : _b; })


#define WINCONST 0.85185      /* 0.46/0.54 for Hamming window */
#define FSAMP 8000.0         /* sample frequency, ensure this matches Config for AIC */
#define FFTLEN 256           /* fft length = frame length 256/8000 = 32 ms*/
#define NFREQ (1+FFTLEN/2)      /* number of frequency bins from a real FFT */
#define OVERSAMP 4           /* oversampling ratio (2 or 4) */
#define FRAMEINC (FFTLEN/OVERSAMP)   /* Frame increment */
#define CIRCBUF (FFTLEN+FRAMEINC) /* length of I/O buffers */

#define OUTGAIN 16000.0       /* Output gain for DAC */
#define INGAIN  (1.0/OUTGAIN)   /* Input gain for ADC  */

#define PI 3.141592653589793    // PI defined here for use in your code
#define TFRAME (FRAMEINC/FSAMP)       /* time between calculation of each frame */

// Noise Minimum Buffer Related
#define NOISE_BUFFER_NUM 4.0    // this is the number of noise buffers we are keeping
#define NOISE_TIME 4.0         // the time, in seconds for the period of time that we are keeping
    the buffers for
#define FRAMES_PER_NOISE_BUF ((int)(NOISE_TIME/NOISE_BUFFER_NUM/TFRAME))  // this is the number of
     frames processed before a noise buffer rotation happens
```

```

/******************************** Global declarations ********************************/

/* Audio port configuration settings: these values set registers in the AIC23 audio
   interface to configure it. See TI doc SLWS106D 3-3 to 3-10 for more info. */
DSK6713_AIC23_Config Config = { \
       /**********************************************************************/
       /*   REGISTER              FUNCTION             SETTINGS          */
       /**********************************************************************/\
    0x0017,  /* 0 LEFTINVOL  Left line input channel volume  0dB                    */\
    0x0017,  /* 1 RIGHTINVOL Right line input channel volume 0dB                    */\
    0x01f9,  /* 2 LEFTHPVOL  Left channel headphone volume   0dB                    */\
    0x01f9,  /* 3 RIGHTHPVOL Right channel headphone volume  0dB                    */\
    0x0011,  /* 4 ANAPATH    Analog audio path control       DAC on, Mic boost 20dB*/\
    0x0000,  /* 5 DIGPATH    Digital audio path control      All Filters off        */\
    0x0000,  /* 6 DPOWERDOWN Power down control              All Hardware on        */\
    0x0043,  /* 7 DIGIF      Digital audio interface format  16 bit                 */\
    0x008d,  /* 8 SAMPLERATE Sample rate control        8 KHZ-ensure matches FSAMP */\
    0x0001   /* 9 DIGACT     Digital interface activation    On                     */\
       /**********************************************************************/
};

// Codec handle:- a variable used to identify audio interface
DSK6713_AIC23_CodecHandle H_Codec;

/*************** buffers **********************/
float *inbuffer, *outbuffer;      // Input/output circular buffers
complex *frameN;              // buffer for frame N
complex *frameN1;          // buffer frame N-1 (enhancement 8)
complex *frameN2;          // buffer frame N-2 (enhancement 8)
complex *outFrame;          // output frame content (enhancement 8)
float *inwin, *outwin;               // Input and output windows coefficients
float ingain, outgain;         // ADC and DAC gains
float cpufrac;             // Fraction of CPU time used
volatile int io_ptr=0;               // Input/ouput pointer for circular buffers
volatile int frame_ptr=0;            // Frame pointer

float *noiseBuffer;        // the noise circular buffer of all the M subbufs
int curM_offset = 0;       // noise sub-buffer pointer (which M buffer we're chosing)
float *previousFFTvalue;     // for LPFing the FFT bins

/* Enhancement 8 buffers */
float *previousFrameNXRatio;     // for storing previous |N(w)|/|X(w)|
float *frameN1ModY;          // for storing frame N-1 |Y(w)|
float *frameN2ModY;          // for storing frame N-2 |Y(w)|

float *noiseLpfBuffer;           // buffer to LPF the noise estimate to subtract (enhancement 3)

/************** parameters *****************/
float noiseLambda = 0.05f;        // lower bound coefficient for spectral substractions
float noiseOversubtract = 3.2f;      // noise oversubtraction parameter (alpha)

/* enhancements 1 & 2 & 3 */
float freqLpfTimeConstant = 0.06f;     // enhancement 1/2 time constant parameter for LPF
float noiseLpfTimeConstant = 0.1f;  // enhancement 3 time constant parameter for LPF
float freqLPF_K, noiseLPF_K;          // calculated factor for enhancement 1/2 & 3 respectively
float prev_freqLpfTimeConstant = 0;    // previous values for the above time constant values
```

```
float prev_noiseLpfTimeConstant=0;  // used for tracking and seeing if the factors needs
    recalculations

/* enhancement 6 parameters */
float enhance6HighFreqLowerBound = 0.00625f;  // fraction of frequency bin to consider as the
    lower bound for "high freq"
float enhance6HighFreqUpperBound = 0.99375f;  // the former two should add to one
float enhance6LowFreqGain = 10.f;       // factor to multiply with alpha for low freq
float enhance6HighFreqGain = 1.f;       // factor to multiply with alpha for high freq
float enhance6LowFreqThreshold = 0.5f;      // low freq SNR threshold (NOT in dB)
float enhance6HighFreqThreshold = 1.f;      // high freq SNR threshold (NOT in dB)

// calculated enhancement 6 parameters
int enhance6HighFreqBinLowerBound, enhance6HighFreqBinUpperBound; // calculated actual frequency
    bin numbers
float prev_enhance6HighFreqLowerBound = 0, prev_enhance6HighFreqUpperBound=0; // previous values
    for the thresholds above to see if recalculation is needed

// enhancement 8 threshold
float enhancement8Threshold = 3.f;        // N/X ratio threshold for enhancement 8

/************** enhancement switches ****************/
short enhancement1 = 1;
short enhancement2 = 1;    // this overrides enhancement 1
short enhancement3 = 1;
short enhancement4Choice = 4; // choose zero to turn enhancement 4/5 off.
short enhancement6 = 0;
short enhancement8 = 0;

short enhancementZero = 1;

 /****************************** Function prototypes ******************************/
void init_hardware(void);     /* Initialize codec */
void init_HWI(void);            /* Initialize hardware interrupts */
void ISR_AIC(void);             /* Interrupt service routine for codec */
void process_frame(void);       /* Frame processing routine */

/********************************** Main routine ***********************************/
void main()
{

    int k; // used in various for loops

  /*  Initialize and zero fill arrays */

  /*** buffers ***/
  inbuffer  = (float *) calloc(CIRCBUF, sizeof(float)); /* Input array */
    outbuffer = (float *) calloc(CIRCBUF, sizeof(float)); /* Output array */

  frameN    = (complex *) calloc(FFTLEN, sizeof(complex));  /* FrameN for processing*/
  frameN1   = (complex *) calloc(FFTLEN, sizeof(complex));  /* FrameN-1 for processing*/
  frameN2   = (complex *) calloc(FFTLEN, sizeof(complex));  /* FrameN-2 for processing*/
  outFrame  = (complex *) calloc(FFTLEN, sizeof(complex));  /* final Output Frame */

    inwin   = (float *) calloc(FFTLEN, sizeof(float));  /* Input window */
    outwin    = (float *) calloc(FFTLEN, sizeof(float));  /* Output window */

```

```
    noiseBuffer      = (float *) calloc(NOISE_BUFFER_NUM*(FFTLEN/2 + 1), sizeof(float)); // noise
        estmiation buffer

    previousFFTvalue  = (float *) calloc(FFTLEN/2+1, sizeof(float));    // enhancement 2 buffer

    noiseLpfBuffer    = (float *) calloc(FFTLEN/2+1, sizeof(float));    // enhancement 3 buffer

    /* enhancement 8 buffers */
    previousFrameNXRatio  = (float *) calloc(FFTLEN/2+1, sizeof(float));
    frameN1ModY       = (float *) calloc(FFTLEN/2+1, sizeof(float));
    frameN2ModY       = (float *) calloc(FFTLEN/2+1, sizeof(float));

  /* initialize board and the audio port */
    init_hardware();

    /* initialize hardware interrupts */
    init_HWI();

  /* initialize algorithm constants */

    for (k=0;k<FFTLEN;k++)
  {
    inwin[k] = sqrt((1.0-WINCONST*cos(PI*(2*k+1)/FFTLEN))/OVERSAMP);
    outwin[k] = inwin[k];
  }
    ingain=INGAIN;
    outgain=OUTGAIN;


    /* main loop, wait for interrupt */
    while(1)
    {
      // recalculate values if necessary (for manual runtime tweaking)
      if (prev_freqLpfTimeConstant != freqLpfTimeConstant)
      {
        prev_freqLpfTimeConstant = freqLpfTimeConstant;
        freqLPF_K = exp(-1.f*TFRAME/freqLpfTimeConstant);
      }

      if (prev_noiseLpfTimeConstant != noiseLpfTimeConstant)
      {
        prev_noiseLpfTimeConstant = noiseLpfTimeConstant;
        noiseLPF_K = exp(-1.f*TFRAME/noiseLpfTimeConstant);
      }

      if (prev_enhance6HighFreqLowerBound != enhance6HighFreqLowerBound)
      {
        prev_enhance6HighFreqLowerBound = enhance6HighFreqLowerBound;
        enhance6HighFreqBinLowerBound = (int) (FFTLEN*enhance6HighFreqLowerBound);
      }

      if (prev_enhance6HighFreqUpperBound != enhance6HighFreqUpperBound)
      {
        prev_enhance6HighFreqUpperBound = enhance6HighFreqUpperBound;
        enhance6HighFreqBinUpperBound = (int) (FFTLEN*enhance6HighFreqUpperBound);
      }

```

```c
        process_frame();

    }
}

/********************************** init_hardware() *********************************/
void init_hardware()
{
    // Initialize the board support library, must be called first
    DSK6713_init();

    // Start the AIC23 codec using the settings defined above in config
    H_Codec = DSK6713_AIC23_openCodec(0, &Config);

  /* Function below sets the number of bits in word used by MSBSP (serial port) for
  receives from AIC23 (audio port). We are using a 32 bit packet containing two
  16 bit numbers hence 32BIT is set for  receive */
  MCBSP_FSETS(RCR1, RWDLEN1, 32BIT);

  /* Configures interrupt to activate on each consecutive available 32 bits
  from Audio port hence an interrupt is generated for each L & R sample pair */
  MCBSP_FSETS(SPCR1, RINTM, FRM);

  /* These commands do the same thing as above but applied to data transfers to the
  audio port */
  MCBSP_FSETS(XCR1, XWDLEN1, 32BIT);
  MCBSP_FSETS(SPCR1, XINTM, FRM);


}
/********************************** init_HWI() **************************************/
void init_HWI(void)
{
  IRQ_globalDisable();      // Globally disables interrupts
  IRQ_nmiEnable();          // Enables the NMI interrupt (used by the debugger)
  IRQ_map(IRQ_EVT_RINT1,4);    // Maps an event to a physical interrupt
  IRQ_enable(IRQ_EVT_RINT1);     // Enables the event
  IRQ_globalEnable();       // Globally enables interrupts

}

/******************************** process_frame() ***********************************/
void process_frame(void)
{
  int i, j, k, m;    // various loop counters
  float noiseFactor, noiseMin;   // noise subtraction
  float noiseFactorA, noiseFactorB; // enhancement 4
  float x;       // holds the abs for the current sample

  static int frameCounter = 0;     // frame counter for noise buffer rotation

  short rotatedM;    // whether noise buffer rotation has been done
  float noiseVote;  // low pass filtered noise estimate P(w) (enhancement 1/2)

  int io_ptr0;    // IO pointer

  /* work out fraction of available CPU time used by algorithm */
```

```
cpufrac = ((float) (io_ptr & (FRAMEINC - 1)))/FRAMEINC;

/* wait until io_ptr is at the start of the current frame */
while((io_ptr/FRAMEINC) != frame_ptr);

/* then increment the framecount (wrapping if required) */
if (++frame_ptr >= (CIRCBUF/FRAMEINC)) frame_ptr=0;

/* save a pointer to the position in the I/O buffers (inbuffer/outbuffer) where the
data should be read (inbuffer) and saved (outbuffer) for the purpose of processing */
io_ptr0=frame_ptr * FRAMEINC;

/* copy input data from inbuffer into inframe (starting from the pointer position) */

m=io_ptr0;
  for (k=0;k<FFTLEN;k++)
{
  frameN[k].r = inbuffer[m] * inwin[k];
  frameN[k].i = 0.f;
  if (++m >= CIRCBUF) m=0; /* wrap if required */
}

/************************ DO PROCESSING OF FRAME  HERE ************************/

fft(FFTLEN, frameN);  // FFT of this frame

// Noise minimum buffer handling
if (++frameCounter >= FRAMES_PER_NOISE_BUF) // rotate noise buffer if time period passed
{
  frameCounter = 0;

  if (++curM_offset >= NOISE_BUFFER_NUM) // circular buffer wrap for noise bufs
    curM_offset = 0;

  // M buffers have been rotated, set corresponding flag
  rotatedM = 1;
}
else
{ // no rotations, usual operation
  rotatedM = 0;
}

// iterate over fft bins
/*
  The bins are complex-conjugate symmetrical about bin 128
  So we just process bins 0 - 128.
 **/
for (i = 0; i <= FFTLEN/2; i++)
{
  x = cabs(frameN[i]); // absolute of the signal's fft bin

  noiseVote = x; //default

  ////////////////////////////////////////////////////////////////////////
  // Optional enhancements
  if (enhancement1 && !enhancement2) // LPF the FFT bins
  {
```

```
      noiseVote = (1-freqLPF_K)*x + freqLPF_K*(previousFFTvalue[i]);
      previousFFTvalue[i] = noiseVote;
    }
    if (enhancement2) // power domain LPF of FFT bins
    {
      noiseVote = sqrt((1-freqLPF_K)*x*x + freqLPF_K*(previousFFTvalue[i]*previousFFTvalue[i]));
      previousFFTvalue[i] = noiseVote;
    }

    ////////////////////////////////////////////////////////////////////////

    // if M buffers rotated -> overwrite bin with new vote
    // store the minimum of the current noise value in M0 and the new bin value
    *(noiseBuffer + curM_offset*(FFTLEN/2 + 1) + i) = (rotatedM) ? noiseVote : min(noiseVote, *(
        noiseBuffer + curM_offset*(FFTLEN/2 + 1) + i)); //TODO: []?

    ////////////////////////////////////////////////////////////////////////
    // iterate over noise min buffers and select the smallest bin value

    noiseMin = *(noiseBuffer + i);
    for (j = 1; j < NOISE_BUFFER_NUM; ++j)
    {
      noiseMin = min(noiseMin, *(noiseBuffer + j*(FFTLEN/2 + 1) + i));
    }
    // oversubstract by alpha coefficient
    noiseMin *= noiseOversubtract;

    ////////////////////////////////////////////////////////////////////////


    /* enhancement 3 - LPF noise estimate */
    if (enhancement3)
    {
      noiseLpfBuffer[i] = (1-noiseLPF_K)*noiseMin + noiseLPF_K*noiseLpfBuffer[i];
      noiseMin = noiseLpfBuffer[i];
    }

    /* Enhancement 6 - further noise overestimation with a sharp 0/1 cutoff */
    if (enhancement6)
    {
      float SNR = x/noiseMin;
      if (i > enhance6HighFreqBinLowerBound && i < enhance6HighFreqBinUpperBound)
      { // high frequency handling
        if (SNR < enhance6HighFreqThreshold )
          noiseMin *= enhance6HighFreqGain;
      }
      else
      { // low frequency handling
        if (SNR < enhance6LowFreqThreshold  )
          noiseMin *= enhance6LowFreqGain ;
      }
    }

    /* enhancement 4 & 5 */
    switch (enhancement4Choice)   // calculate G(w), equation used chosen based on switch value
    {
      float temp;
```

```
      case 1:
        temp = noiseMin/x;
        noiseFactorA = noiseLambda * temp;
        noiseFactorB = 1.f - temp;
        break;
      case 2:
        noiseFactorA = noiseLambda * noiseVote/x;
        noiseFactorB = 1.f - noiseMin/x;
        break;
      case 3:
        temp = noiseMin/noiseVote;
        noiseFactorA = noiseLambda * temp;
        noiseFactorB = 1.f - temp;
        break;
      case 4:
        noiseFactorA =  noiseLambda;
        noiseFactorB = 1.f -  noiseMin/noiseVote;
        break;

      /* enhancement 5 power handling */
      case 5: // power version of no enhancement 4
        noiseFactorA =  noiseLambda;
        noiseFactorB =  sqrt(1.0 - noiseMin*noiseMin/(x*x));
        break;
      case 6: // power version of enhancement 4 - 1
        temp = noiseMin*noiseMin/(x*x);
        noiseFactorA = noiseLambda * sqrt(temp);
        noiseFactorB = sqrt(1.f - temp);
        break;
      case 7: // power version of enhancement 4 - 2
        noiseFactorA = noiseLambda * sqrt(noiseVote*noiseVote/(x*x));
        noiseFactorB = sqrt(1.f - noiseMin*noiseMin/(x*x));
        break;
      case 8: // power version of enhancement 4 - 3
        temp = noiseMin*noiseMin/(noiseVote*noiseVote);
        noiseFactorA = noiseLambda * sqrt(temp);
        noiseFactorB = sqrt(1.f - temp);
        break;
      case 9: // power version of enhancement 4 - 4
        noiseFactorA =  noiseLambda;
        noiseFactorB = sqrt(1.f -  noiseMin*noiseMin/(noiseVote*noiseVote));
        break;
      /* enhancement 4 & 5 off */
      case 0:
      default:
        noiseFactorA = noiseLambda;
        noiseFactorB = 1.0 - noiseMin/x;
        break;
    }

    ////////////////////////////////////////////////////////////////////////////////
    // Perform spectral substraction

    noiseFactor = max(noiseFactorA, noiseFactorB);
    frameN[i] = rmul(noiseFactor, frameN[i]);

    ////////////////////////////////////////////////////////////////////////////////
```

```
    /* Enhancement 8 further processing */
    if (enhancement8)
    {
      x *= noiseFactor;     // absolute of Yn

      // check if previous frame N/X ratio is above a certain threshold
      if (previousFrameNXRatio[i] > enhancement8Threshold)
      {
        // find the frame with the min abs(Y) and use that frame as the output
        if (x < frameN1ModY[i] && x < frameN2ModY[i])
          outFrame[i] = frameN[i];
        else if (frameN1ModY[i] < x && frameN1ModY[i] < frameN2ModY[i])
          outFrame[i] = frameN1[i];
        else
          outFrame[i] = frameN2[i];
      }
      else  // below threshold? just output the previous frame
      {
        outFrame[i] = frameN1[i];
      }

      previousFrameNXRatio[i] = noiseMin/x;    // update the N/X ratio
      frameN2ModY[i] = x;    // overwrite the value for N-2 frame
    }
    else // plain old no enhancement8
    {
      outFrame[i] = frameN[i]; // direct assignment
    }

    if ( enhancementZero && (i == 0 || i == 1) )
    {
      outFrame[i] = cmplx(0,0);
    }
  }

  if (enhancement8) // if enhancement 8, swap relevant buffers
  {
    complex *tempFrame;

    tempFrame = frameN2;
    frameN2 = frameN1; // new N-2 frame is old N-1 frame
    frameN1 = frameN;  // new N-1 frame is old N frame
    frameN = tempFrame; // new N frame reuses the old N-2 frame buffer for new input incoming

    // swap |Y(w)| buffers
    float *temp;

    temp = frameN2ModY;    // note that the mod Y for frame N is already stored here
    frameN2ModY = frameN1ModY;
    frameN1ModY = temp;

  }

  /*
    Bins 129 - 255 are complex conjugates of bins 127-1 respectively
   */
  for (i = FFTLEN/2+1 ; i < FFTLEN; i++)
```

```
  {
    outFrame[i] = conjg(outFrame[FFTLEN - i]);
  }

  ifft(FFTLEN, outFrame); // perform inverse FFT to return us back to time domain

  /*****************************************************************************/

    /* multiply outframe by output window and overlap-add into output buffer */

  m=io_ptr0;

    for (k=0;k<(FFTLEN-FRAMEINC);k++)
  {                             /* this loop adds into outbuffer */
      outbuffer[m] = outbuffer[m]+outFrame[k].r*outwin[k];
    if (++m >= CIRCBUF) m=0; /* wrap if required */
  }
    for (;k<FFTLEN;k++)
  {
    outbuffer[m] = outFrame[k].r*outwin[k];    /* this loop over-writes outbuffer */
      m++;
  }
}
/************************** INTERRUPT SERVICE ROUTINE  ****************************/

// Map this to the appropriate interrupt in the CDB file

void ISR_AIC(void)
{
  short sample;
  /* Read and write the ADC and DAC using inbuffer and outbuffer */

  sample = mono_read_16Bit();
  inbuffer[io_ptr] = ((float)sample)*ingain;
    /* write new output data */
  mono_write_16Bit((int)(outbuffer[io_ptr]*outgain));

  /* update io_ptr and check for buffer wraparound */

  if (++io_ptr >= CIRCBUF) io_ptr=0;
}

/******************************************************************************/
```

References

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