DESC9115 2013 LAB REPORT 2

DIGITAL AUDIO SYSTEMS

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Abstract:

Presented in this report is a digital multiband compressor product development. With a working prototype built in the `matlab` environment called `mcompdav.m`. It calls the sub-function `holtcomp.m` to perform RMS level detection and compression. Two different filtering options during the band splitting process are offered and further wave shaping through the implementation of output gain controls per-band is permitted. Plot figures and sound file examples are also presented.

Introduction:

The ability to have precise and musical sounding digital dynamics processing for mastering is highly desirable for many producers and mastering engineers. Provision for different types of filtering during band splitting offering different phase characteristics is also desirable. Another useful and desirable feature is the ability to compensate for gain reduction and 'shape' the spectral output by controlling the output gain of each frequency band. The product development in this lab assignment addresses these desirable features and the accompanying documentation explains the processes that have been implemented to achieve them.

The Product Design: ***A digital 3-band compressor for mastering***.

The band splitting section of the system shall feature both linear-phase and minimum phase filtering options. The compression section shall have RMS level detection style compression. The output section shall have variable control over output gain per frequency band, allowing for considerable tonal correction or shaping. This product will be able to serve as a mastering processor for music mastering. Whilst currently in a single channel format, it could be quickly and simply extended for use as a stereo or multi-channel surround processor within the DSP code. The product has been developed as a fully functional and clearly commented `Matlab` function `mcompdav.m` accompanied by a sub-function `holtcomp.m` (based on m.file 4.2 by M.Holters, 2011) and a script, `callmulticomp.m`, which allows for easy inputting of variables.

The block diagram of the major system component is shown below in **Figure 1:**

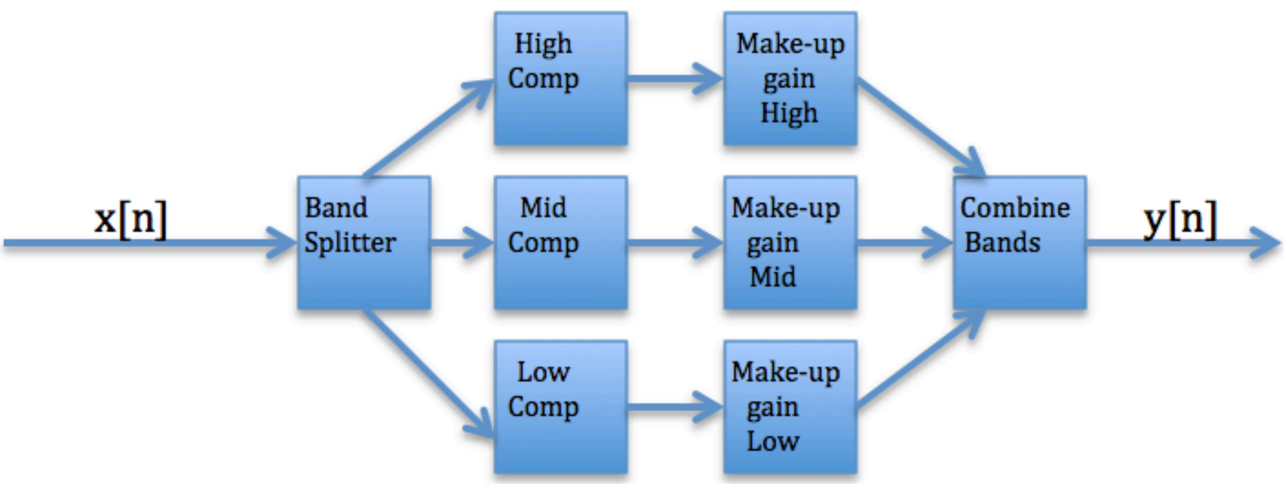

Fig 1. Block diagram showing the major components of the 3-band compressor.

Key Design Elements:
The Band Splitting Process:

The band splitting stage of the system will be carried out using two different, user definable methods. Firstly the linear phase method, which implements FIR digital filters. FIR filters can be designed to have extremely good performance in terms of stop band attenuation and a flat pass-band. They also offer a linear phase response (Dutilleux & Zolner, 2002, p. 48), which can be desirable if the mastering engineer wishes to preserve the phase characteristics of the signal. The FIR filter is designed using delay-and-add 'feed-forward' type processing, where the output of the system is a weighted sum of the delayed inputs.

The difference equation for an FIR filter shows how the output signal is derived from the input signal.

$$y[n] = b_0x[n] + b_1x[n-1] + b_2x[n-2] + \cdots + b_Nx[n-N]$$

Where **y[n]** is the output, **x[n]** is the input, **b[N]** are the feed-forward coefficients and **N** is the filter order.

There are some trade-offs to FIR filter implementation. FIR filters are less time and computationally efficient (Smith, 1997, p.348), however it can be interpreted that the speed becomes less of an issue when the material is pre-recorded to memory (a computer disk) as the system does not necessarily need to be real-time to be effective. The other potential detractor from implementing FIR filters is the potential for 'pre-ringing'. Pre-ringing can be audible to the listener as a subjective reduction of dynamics of a transient, or as an audible 'pre-echo' in extreme cases. Figures **2a** and **2b** (see appendix 1) show the pre ringing introduced by running a snare drum sample through a 2000 point FIR filter in our product versus the IIR implementation.

The second filtering option in the band-splitting path implements a 3rd order butterworth IIR design. 3rd order Butterworth filters offer a maximally flat pass-band (Smith, 1997, p.333) and a roll off which is useful for our application. They have also been tested to be stable down to 40Hz for this application, which is an important factor for flexibility of the system.

The difference equation for an IIR filter shows how the output signal is derived from the input signal.

$$y[n] = b_0x[n] + b_1x[n-1] + b_2x[n-2] + \cdots + b_Nx[n-N]$$
$$-a_1y[n-1] - a_2y[n-2] - a_3y[n-3] - \cdots + a_Ny[n-N]$$

Where **y[n]** is the output, **x[n]** is the input, **b[N]** are the feed-forward coefficients, **a[N]** are the feed-back coefficients and **N** the filter order.

The transfer function of the filter is given by:

$$H(z) = \frac{\sum_{i=0}^{M} b_i z^{-i}}{\sum_{i=1}^{N} a_i z^{-i}}$$

Where **b[i]** are the feed-forward coefficients, **a[i]** are the feedback coefficients and **z[i]** are the units of delay.

From this equation the roots of the numerator and denominator, which are often called poles and zeroes can be derived. These form the basis of our filter design.

$$H(z) = \frac{N(z)}{D(z)}$$

By selecting to the `'min'` variable as the filter choice, the band splitting will be carried out by the IIR method. No pre-ringing will be present. However, there is some inherent phase distortion introduced by the IIR filtering process. The phase response of system, for both minimum phase and zero/linear phase choices is shown in **Figure 3** and **4** (see appendix 2).

RMS Compression:

The compressor section in this system is implemented in the `'holtcomp.m'` function. In this process, the input value is delayed and fed forward. Meanwhile, a side-chain path is created and the RMS level of the input is calculated and then converted into the logarithmic domain, where a gain factor is then calculated. The block diagram below shows the signal flow through the RMS compressor.

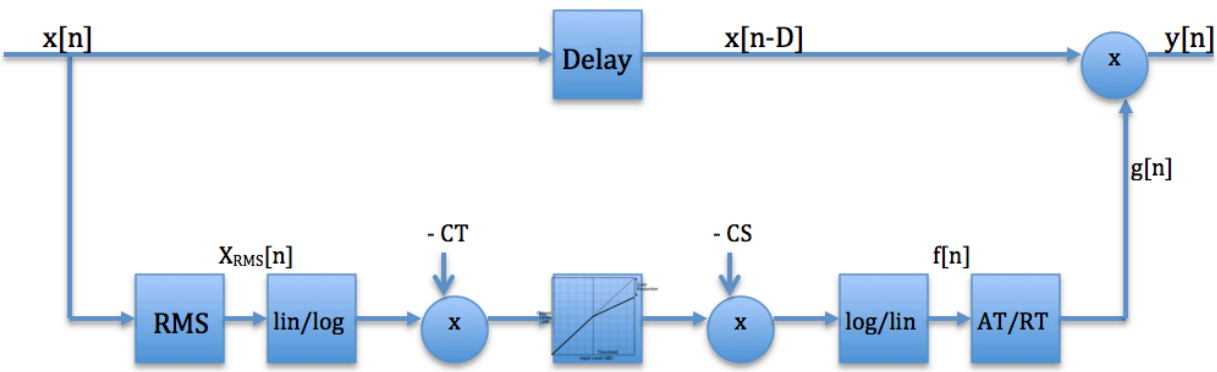

Fig 5
Block diagram of the RMS compressor

The mathematical equation for the gain factor calculation in the logarithmic domain is given by:

$$CS \times (CT - X)$$

Where CS is the compression 'slope', CT is the compressor threshold in dB and X is the RMS input value in dB.

This gain factor is then returned to a magnitude value f[n] where it is further smoothed/scaled by the attack time and release time variables. It is then applied g[n] to the delayed input value x[n-D], to create a weighted output value. In effect, when the RMS value X(dB) of the input is greater than the CT, the gain factor will be < 1 and the output will be reduced accordingly.

The fed-forward input value needs to be delayed to compensate for the calculation time of the side chain, as emphasized in (Zoelner, 2002, p.95).

Spectral Shaping:

Included in the system is the ability to 'shape' the output of the 3 frequency bands through the addition of output gain controls. This is a simple yet powerful tool for the spectral manipulation of the processed signal. For example if the system is used to control excessive dynamics in bass frequencies of a mix, they can be compressed quite strongly and then 'make up gain' applied to compensate for the gain reduction in the compressor. In this specific example 'tightened' bass can be observed.

The output gain for each band is calculated by transforming the dB input variables into magnitude values. The magnitude gain factor is then applied to the output of each compressor band accordingly. The bands are then added back together and the overall signal is normalized to ensure no clipping is introduced by the gain in the system,. Whilst the spectral changes introduced by the band output gains is preserved. The signal is then passed through a final gain stage defined by the `'outlevel'` variable, which can act as an output boost or attenuation as required by the user.

Operational Limits:

As with any DSP system, there are limits of system capabilities and an optimal range of use. Through initial testing, the system proves stable and capable of allowing a highly flexible range of inputs and is therefore a useful tool for the stated purpose of a mastering processor.

Within the system code a number of range limits are provided for safe and intuitive operation.

For example, a band output gain of more than +30dB is not permissible by the code, effectively limiting the band-output gain range from -∞ to +30dB. This helps define reasonable operating limits in which good results are obtained.

Similarly the crossover frequencies are limited to be between 40Hz and 19kHz. This is due to the limits of IIR filter stability, which dictate that the butterworth filter order must not exceed 3rd order over large range of cutoff frequencies that is required. Again the code will not permit the user to input variables outside of safe range of the system. These ranges are still very flexible for the stated function of this product.

Further to this, within the `Matlab` script `'callmulticomp.m'` there are some 'templates' or examples provided. It is hoped that these give a good demonstration of the capabilities of the product and provide a good starting point to achieve results in mastering applications. We hope that you enjoy using our product!

Appendix 1:

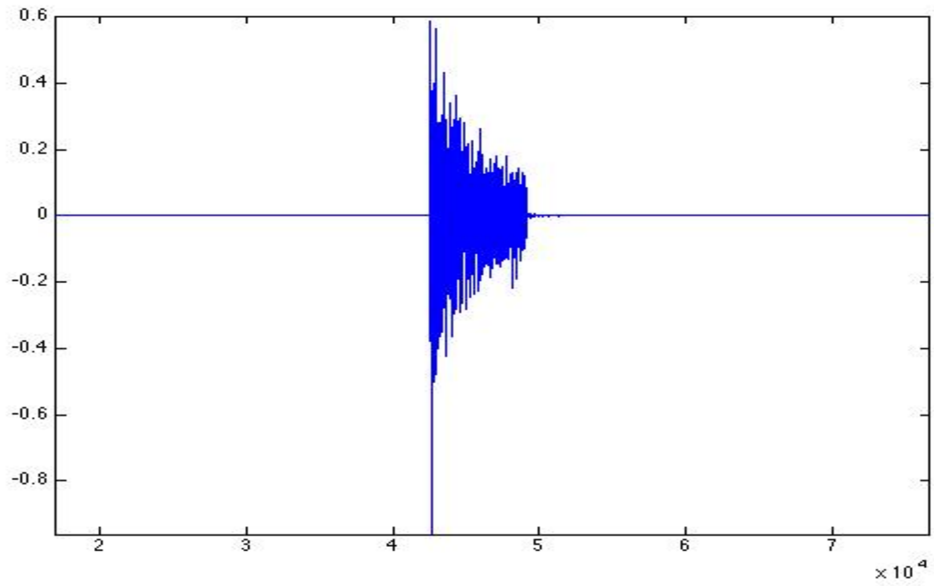

Fig 2a)
Time domain plot of a snare drum sample filtered with IIR filtering.

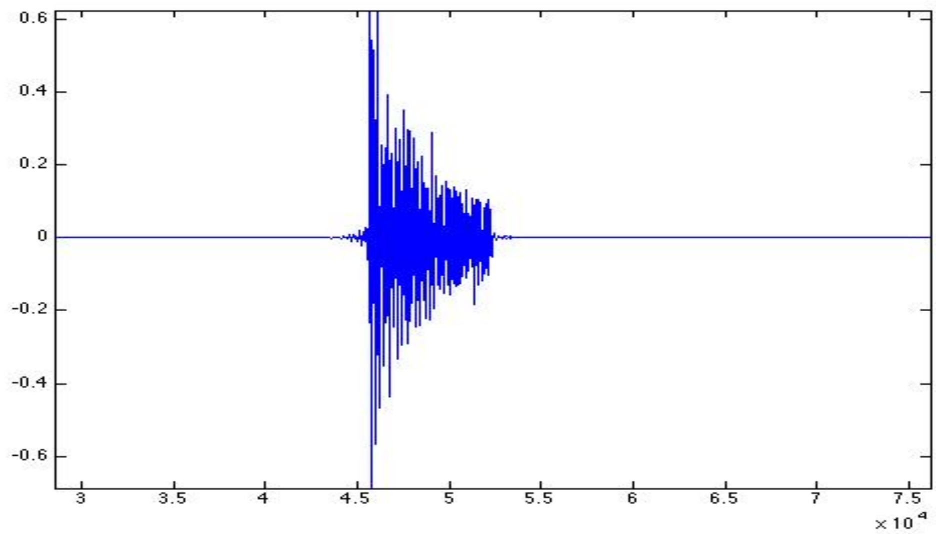

Fig 2b)
Time domain plot of a snare drum sample filtered with 2000 point FIR filter – **note the pre-ringing before the main transient.**

Appendix 2:

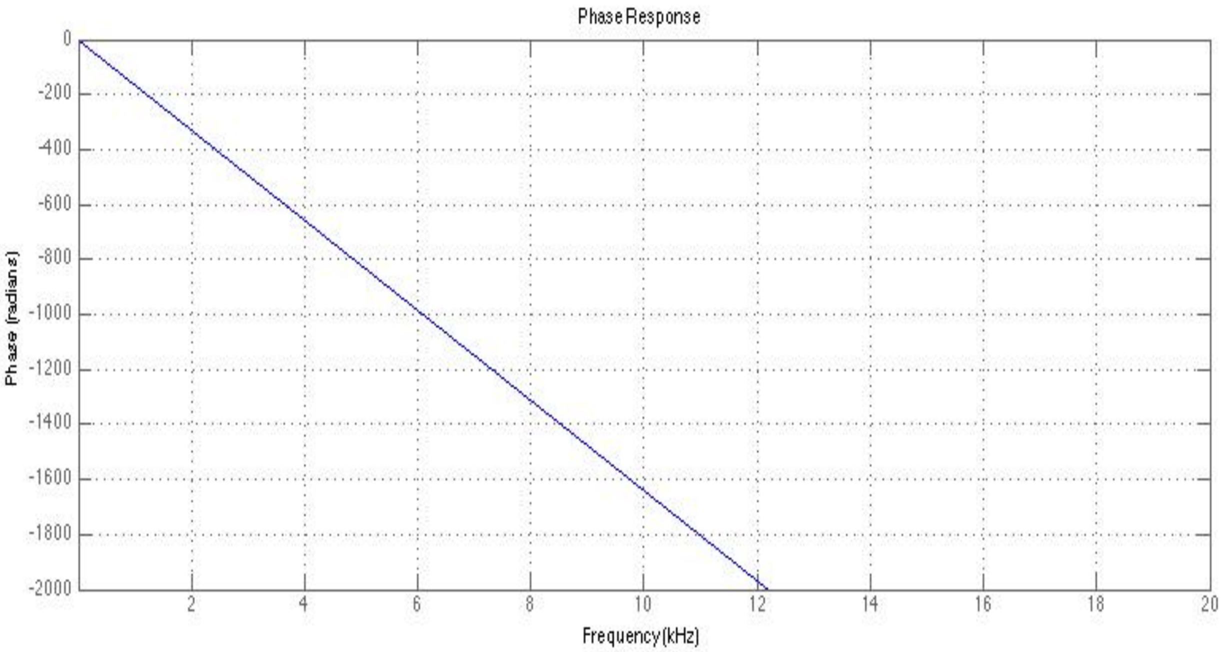

Fig 3

Phase response of the ‘zero’ filter choice setting (fL: 120Hz, fH: 600Hz) shows a linear relationship between phase and frequency for the system set to the ‘zero’ setting.

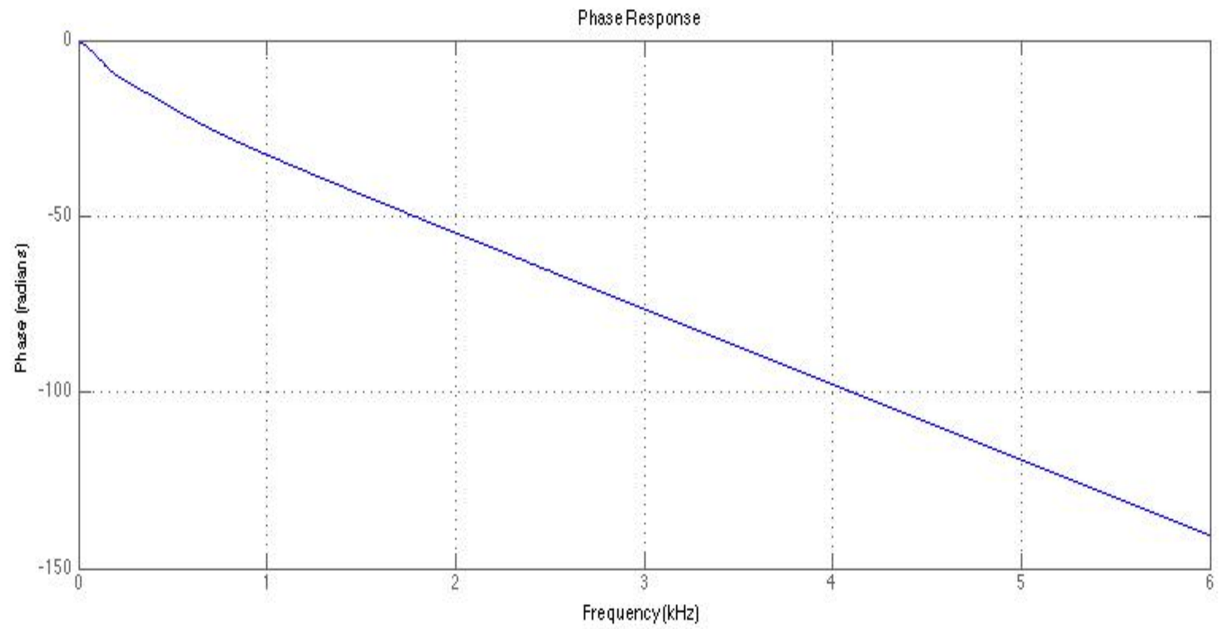

Fig 4

Phase response of the ‘min’ filter choice setting (fL120Hz, fH: 600Hz) showing the non-linear relationship between phase and frequency for the system set to the ‘min’ setting.

Work Cited:

P. Dutilleux & U. Zolner, 2002, Ch 2 'Filters', Ch 5 'Nonlinear Processing' in U. Zolner (ed.), DAFX – Digital Audio Effects, John Wiley & Sons Ltd, Chichester, p. 48, p.95

M. Holters, 'm.file 4.2' in DAFX – Digital Audio Effects 2nd edition, U. Zolner (ed), John Wiley & Sons Ltd, Chichester.

S Smith, 1997, Ch 20 'Chebyshev Filters', Ch 21 'Filter Comparison' in The Scientist and Engineers Guide to Digital Signal Processing, California Technical Publications, California, p. 333, p. 348