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AN AUDIO ENGINEERING SOCIETY PREPRINT

Direct Digital Processing of "Super Audio CD" Signals

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Efficient structures for directly processing 'Super Audio CD' type, signals are presented. We will first review the problems of current approaches. The advantages and disadvantages of processing signals this way are discussed. The structures are then explained and the effects of coefficient accuracy are discussed. Examples of the method performing equalising transfer functions are then presented.

0. INTRODUCTION

Analogue-to-digital (A/D) and digital-to-analogue (D/A) converters often use an intermediate sigma-delta modulating stage to convert signal inputs and outputs into a simple digital form for high quality conversion. In the case of A/D converters, a multibit representation of the signal is achieved with a decimating filter. Similarly, D/A converters employ interpolators to increase the sampling rate and to remove images of the baseband audio signal that are created by oversampling. The well-documented sigma-delta modulating technique [1-4], employing Nth order noise shaping, is then used to create a highly quantised two level signal. This one bit signal is a perfectly valid representation because it contains all the audio band information. This one bit signal is a perfectly valid representation because it contains all of the audio band information and is used as the information carrier in the new "Super Audio CD" format.

Processing the one bit signal directly offers an alternative approach to signal processing and removes the decimating or interpolating requirements in an analogue interface. It also allows a simpler system structure because the interconnections are naturally serial with no implied framing. Also, because the signal is heavily oversampled, the system characteristics can approach those of high quality analogue processors. Both phase response, and distortion effects, are preserved while retaining the advantages of digital processing techniques.

This paper describes filter structures that operate on one-bit signals and develops an efficient structure for realising efficient IIR filters with no multipliers.

1. FILTER STRUCTURES

A one bit signal processor must contain two distinct sections, one to filter the audio signal and a second that converts the resulting multibit signal back to one bit one. This latter section will comprise of a quantiser and a noise shaping filter.

1.1 One Bit FIR Filters

A one bit FIR filter is simple to implement. The basic system is shown in figure 1. The one bit output from the sigma-delta modulator is presented to a filter that has a traditional form comprising L taps (each taps comprises one delay unit and a coefficient multiplier) where L increases if a sharper filter cut-off is required. Filter coefficients are identical to those chosen for multibit PCM at the higher sampling rate. The signal produced at the output of the filter is a multibit signal with bit length depending on the filter coefficient length. The signal will have to be recoded so that the final output is one bit. It has already been established that low pass filtering a one bit signal will produce multibit PCM; the technique is used in A/D conversion. Therefore, provided the FIR filter is low pass or bandpass, high frequency shaped noise will be removed so that the output is PCM and can be requantised with a digital noise shaping modulator. Note that bandpass filtering with an upper cut-off at or just above the audio band edge will remove most of the high frequency noise to produce PCM and will have the effect of a normal highpass filter.

An alternative FIR filter structure incorporates requantisation into the filter as shown in figure 2 for third order noise shaping. The feedback required by the noise shaper may now affect the response of the audio filter and the forward path coefficients must be adjusted to cancel the poles that are introduced. The noise shaping characteristics are not affected by the forward path coefficients. It should be noted that there is little advantage in choosing such an FIR filter structure except for the cost of saving N delay units where N is the order of the noise shaper.

The main problem with one bit FIR filters are that they will be longer than the equivalent PCM one by the oversampling ratio for the same filter response. However there are two ways of implementing them more efficiently. The first way recognises that we don't care about the out of band frequency response and so can subsample the taps [9]. This means that in principle we can have miss out one less than the oversampling ratio of coefficients between each tap. This dramatically improves the computational efficiency. However in practice more taps than the minimum would be required to suppress the high frequency noise before recoding. A better technique due to Heylan [10,11] is to use a combination of cascaded integrators and a sparse tap FIR filter to achieve the desired response. It has the advantage of being efficient and also removes the high frequency noise.

1.2 One Bit IIR Filters

An obvious but expensive system to realise a second order filter is shown in figure 3 where audio filter and requantiser are placed in series. In this structure, the feedback signal is multibit. The audio filter section would have the same coefficient values as its multibit PCM equivalent and, to realise a higher order filter structure, all the second order structures could be placed before a single noise shaping modulator. The noise power introduced by this system depends on the choice of noise shaping modulator and on the wordlength of the signal feedback into the audio filter.

Johns and Lewis [5] have developed a structure without multibit multipliers, figure 4. Their filter is based on the biquad structure with integrators rather than simple delays but places a noise shaping modulator of arbitrary order after each integrator. To create a higher order filter extra integrators and sets of coefficient multipliers are added with feedback taken from after the final modulator, rather than stringing second order filters together with feedback taken from the end of each stage.

The designers of this filter concentrate on minimising the number of components and the system cost. Their final design using third order Butterworth noise shaping in a second order filter section would have 10 delay units, 9 multibit adders and 12 single bit coefficient multipliers. This compares to 5 delay units, 6 adders and 11 multipliers in the previous filter suggesting that this new realisation will be cheaper. Johns and Lewis point out that the signal at nodes (A) and (B) in figure 4 can take on only 4 and 8 values respectively. Subsequently the adders and multipliers directly before the two nodes can be replaced by a 4 input and 8 input multiplexer or ROM. This will have a significant cost saving. Unfortunately the noise performance of this structure is poor.

An alternative structure is shown in figure 5, where the quantiser is replaced by a noise shaping modulator as shown in figure 5 for third order noise shaping. All inputs to the audio filter coefficient multipliers are one bit as the signal is fed back from a point after the quantiser. This feedback signal not only contains the desired output audio components but also the quantisation noise. This means that the quantisation noise will be shaped by the poles of the audio filter resulting in a noise peak in the middle of the audio band. In fact, the comparator is overloaded by this high magnitude/low frequency signal component for any useful input level and becomes unstable. This effect can be avoided by adding extra zeros to the noise shaping transfer function to cancel the poles added by the audio filter, as shown in figure 6. However these structures still require multi-bit multipliers in the noise shaper.

A more practical topology known as the biquad structure combines both the audio and noise shaping filters using a series of integrators and a minimal number of multibit multipliers [6-8]. In this new realisation, illustrated in figure 7, both the noise shaping filter and the audio filter unavoidably share poles and what would be a second order audio filter in a multibit PCM processor becomes

a $(2+N)^{\text{th}}$ order system where N is the order of the noise shaping filter. The system has extra coefficient multipliers in the forward path to provide zeros in the audio filter to exactly cancel the poles introduced by the noise shaper. The poles in the noise shaper introduced by the audio filter should also be cancelled as in the structure of figure 6. This structure can be efficiently implemented as it is possible to implement the NTF zero coefficient via the use of small combinations of powers of two. The main problem with this structure is that the wordlengths required for the coefficients are high (typically 32 bits for 64Fs). The internal dynamic range is also high. An improved version of this filter has been developed by Kershaw [9] and is shown in figure 8. The main difference of this structure is the presence of power of two coupling coefficients between the stages which has the effect of reducing the internal dynamic range, each stage now has similar numerical requirements. The coefficient accuracy required is also reduced to typically 16 bits for a 64Fs system.

2. IMPLEMENTATION

Although these filter structures can seem complicated their implementation is straightforward. Clearly a general purpose DSP approach, while possible, is not to be recommended. Instead one is looking at an FPGA, or custom LSI, approach. The fact that the preferred structures require no multiplies is an advantage. In fact the processing is readily implemented as a series of adders whose inputs are controlled by the input and feedback bit patterns. These can be implemented using a four input multiplexer, or as a double adder structure, one to add the input and one to add the feedback. In practice both require a similar chip area for implementation. However the four input multiplexer has a slight advantage when carry times are considered but the double adder structure is easier to interpolate. This is likely, in practical applications, to favour the double adder approach. The binary powers of two feedforward coefficients can be hardwired, as can the power of two NTF zeros. Given these implementation methods, and the well-behaved scaling of the final IIR structure it is possible to implement this type of processing efficiently in FPGA or LSI technology.

3. SOME RESULTS

Tone control equaliser or shelving filter design is best done with 1st order filters. For the bass cut/boost control, the analogue prototype transfer function is:

$$H(s) = \frac{s + K\omega_c}{s + \omega_c} \quad (23)$$

where K controls the low frequency gain and ω_c controls the bandwidth of frequencies which are boosted or attenuated.

It can be seen that the low frequency gain, $H(0) = K$ and the high frequency gain, $H(\infty) = 1$ as required by a bass control. Regalia and Mitra [12] implement the equaliser by using an all-pass filter.

Figure 9 shows the predicted bass control transfer functions for a corner frequency of 1kHz and various values of gain ranging from -40dB to +40dB

Figure 10 shows the STF and NTFs of the bass control by simulation using gain values of -20dB, -6dB, 6dB and 20dB.

4. CONCLUSION

Filter structures that can process one bit signals directly have been developed and presented. The advantages of these systems over multibit signal processors are that many component savings could be made at the price of faster clocking hardware and marginally higher coefficient wordlengths. The magnitude response of the filters can be as good as that of multibit systems and that the phase response at high frequencies are an improvement on systems that clock at the Nyquist rate. One bit digital filtering shows promise as a technique for achieving high quality system effective signal processing.

5. REFERENCES

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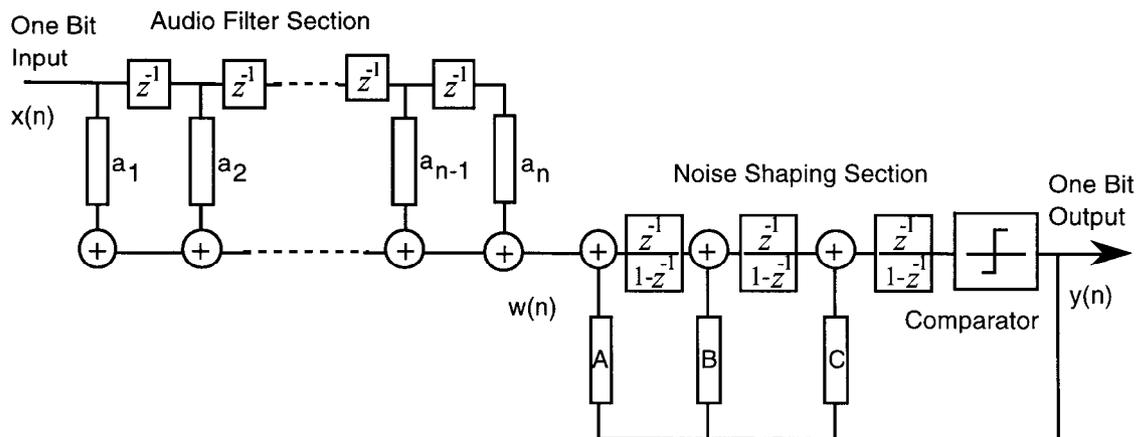


Fig 1: An Mth order one bit FIR filter with audio filter and requantiser in series.

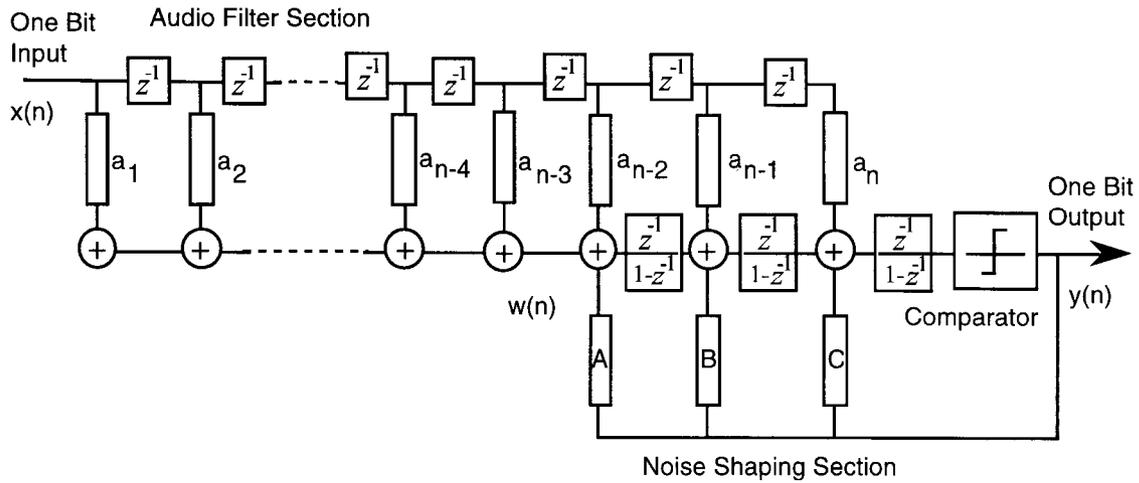


Fig 2: An Mth order one bit FIR filter with embedded third order requantiser.

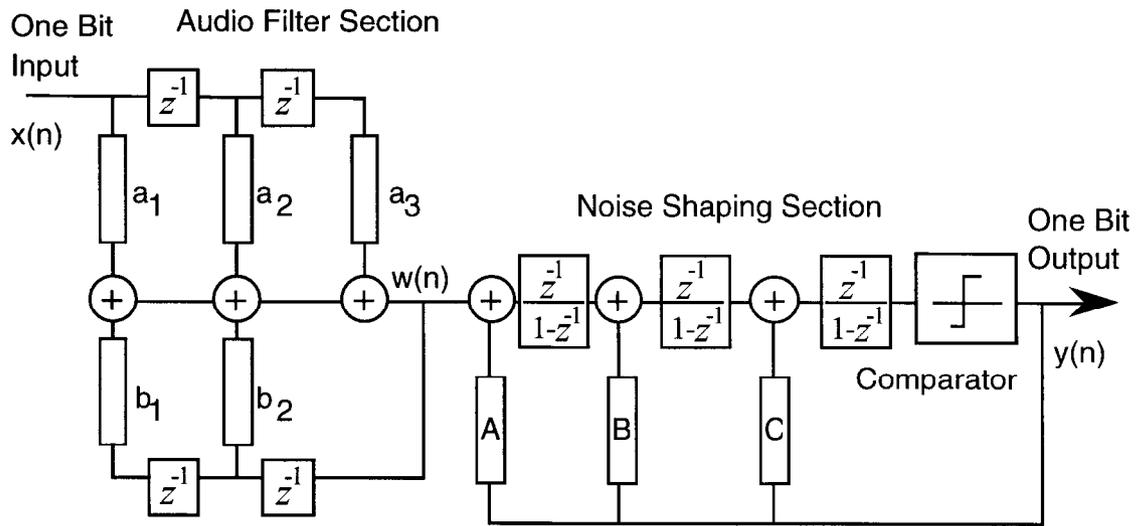


Fig 3: A second order one bit IIR filter consisting of an audio filter cascaded with a requantiser.

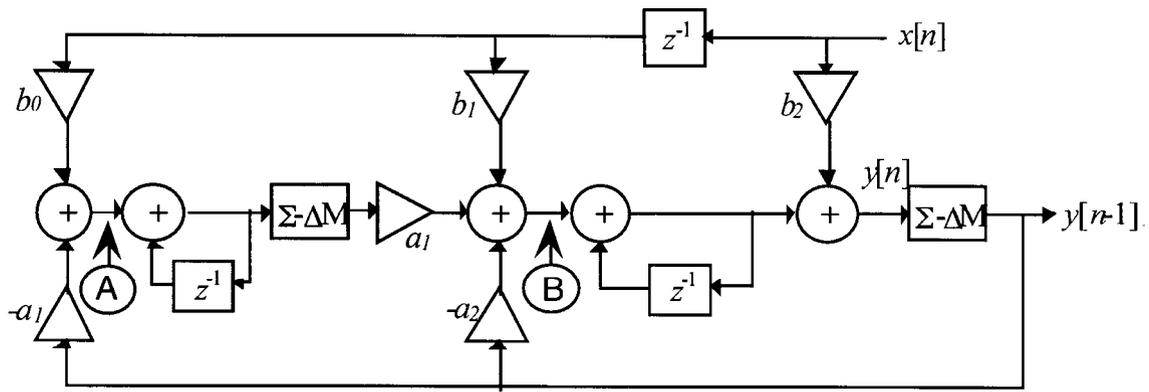


Fig 4: The Johns and Lewis one bit recursive filter with no multibit multipliers.

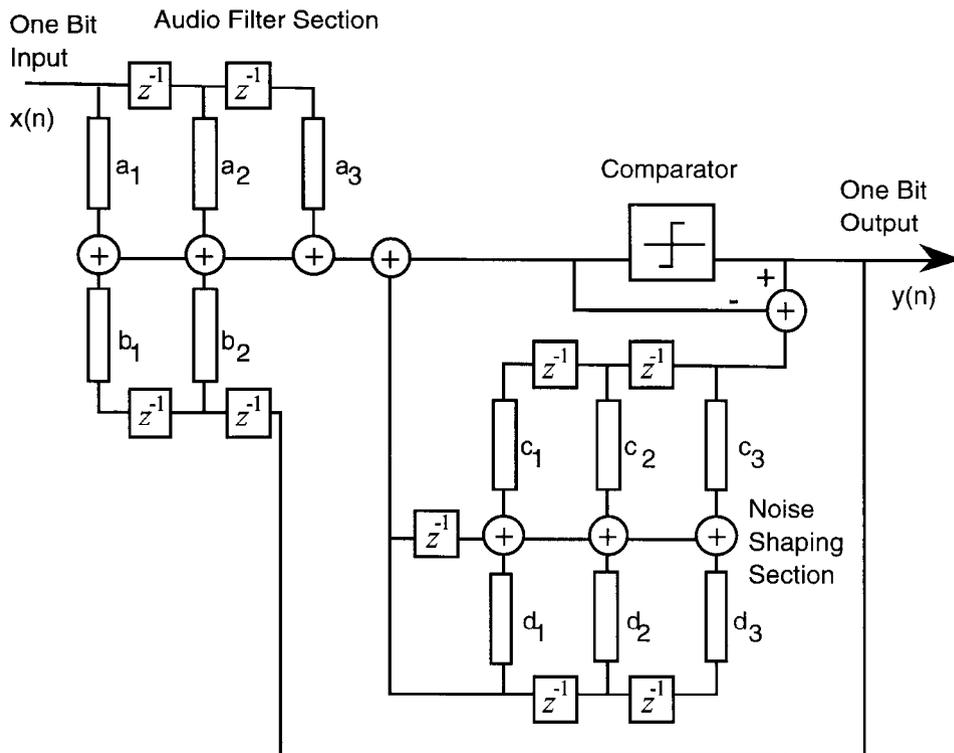


Fig 5: A one bit IIR filter with one bit inputs to the audio filter section.

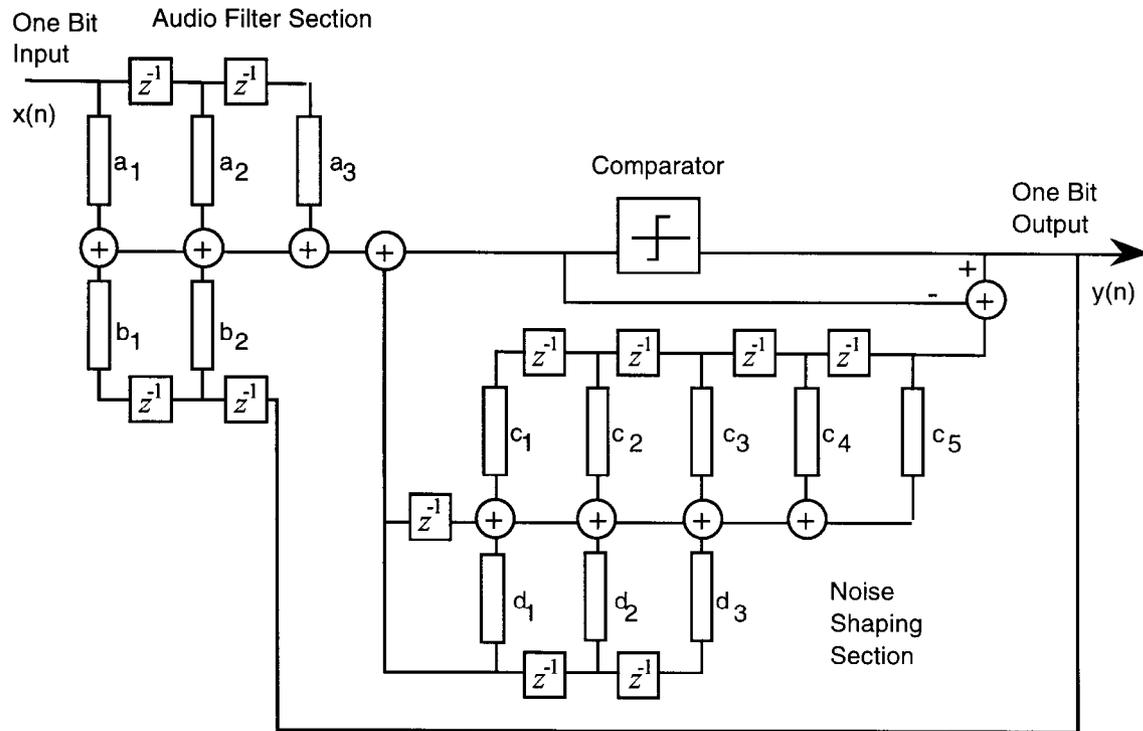


Fig 6: A one bit IIR filter with extra cancelling zeros in the noise shaper.

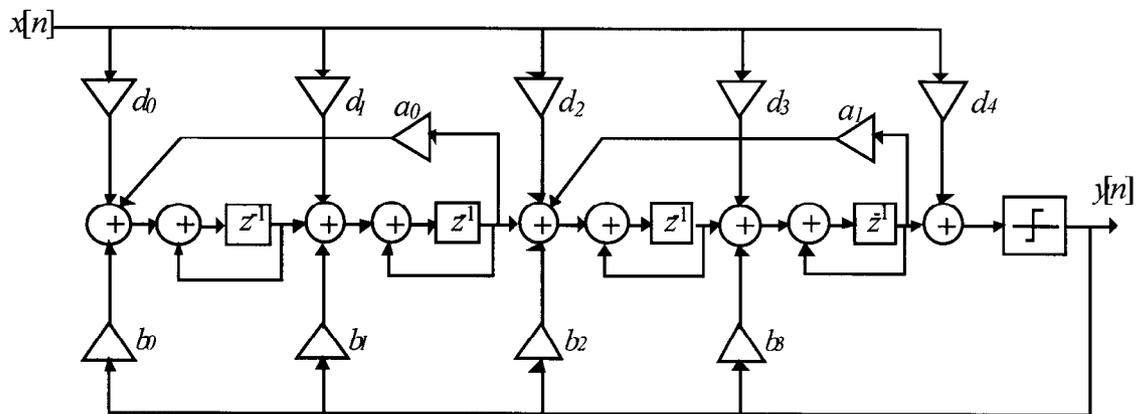


Fig 7: A biquad one bit IIR filter realisation where audio filter and noise shaping filter share poles.

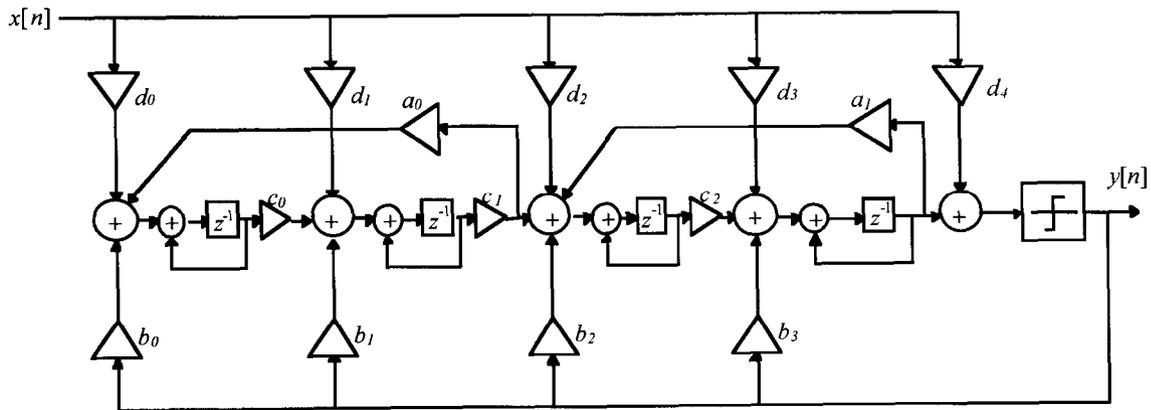


Fig 8: A modified combined structure with integral power of two NTF zero and coupling coefficients.

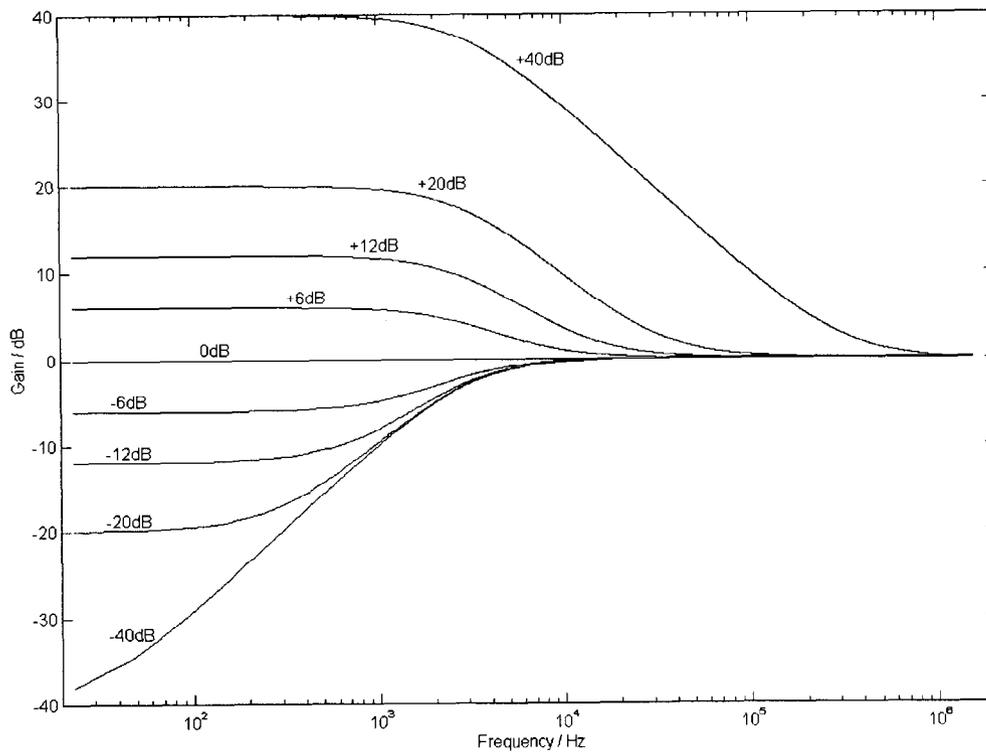


Figure 9 Predicted bass control transfer functions for a modified combined structure structure.

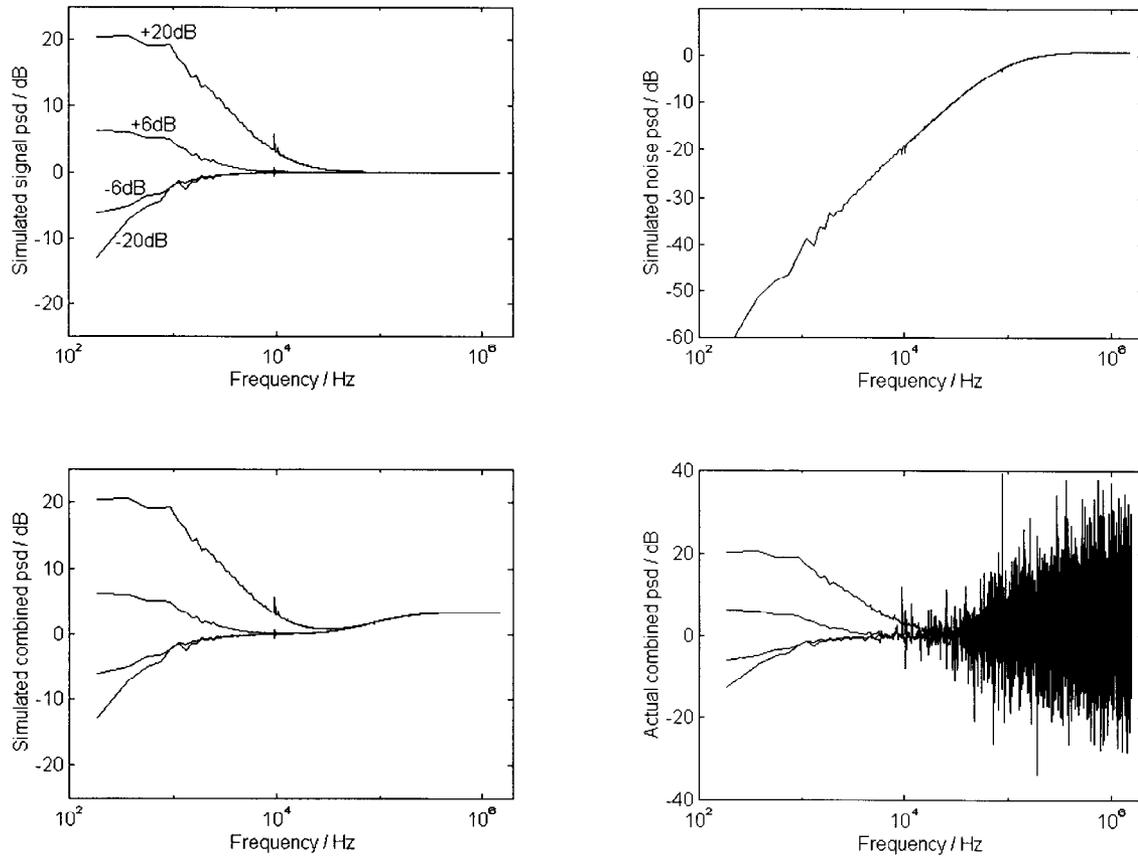


Figure 10 Output spectra of one-bit a modified combined structure bass control filter and noise shaper for various gain values.