CHAPTER 9
CONCLUSIONS AND
FUTURE WORK
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In the first part of research work, we have observed that the more sections we average, lower the variance of the result. However the signal length limits the number of sections possible. To obtain more sections, we can break the signal into overlapping sections. Figure 4 shows the power spectrum for 128 sample overlap. In this case, the sections are statistically dependent, resulting in higher variance. Thus we can conclude that there is trade-off between the number of sections and the overlap rate. In addition to this, a better technique for spectrum estimation of broadband signals from noisy measurements has been proposed. The simulation results have shown that the proposed model based spectrum estimation method give better results in case of broadband signals.

Moreover, we discussed, examined, and compared the performance of the AR spectral estimation methods by simulation. In the introduction part and in the second part, we gave the theory of estimators. After that, simulation of the methods and the best one are discussed. In contrast to other AR spectrum estimation techniques, the modified covariance method appears to give statistically stable spectrum estimate with high resolution.

In the second part of research work, a new digital sample-rate converter has been described. In this sample-rate converter, the output samples are obtained by simply taking over samples from the input and repeating a certain amount of them. The control source of this sample-rate converter is a sigma-delta modulator, which shapes time errors introduced by the conversion process to the higher frequency region.

In this part, a theoretical model has been derived with which the spectral properties of the digital sample-rate converter can be explained. It appears that the simulation results done with this sample-rate converter correspond to the theoretical considerations.

It appears that the performance of the digital sample-rate converter in terms of signal-to-noise ratio is sufficient to achieve a 16-bit performance for the output signal. The conversion factor of
the sample-rate converter lies between 0.59 and 3.33. Within this range, any real number may be chosen.

The sample-rate converter based on shaping of time errors requires an upsample filter which is much simpler than those used in conventional digital sample-rate converters. The upsampling factor can be lower and the stopband attenuation can therefore be smaller. The computational complexity needed is much smaller than for conventional sample-rate conversion methods. This gives a reduction in needed hardware.

This digital sample-rate converter is ideally suited for the application in an oversampled D/A converter. The upsample filter which is needed is already present in such a D/A converter. The sample-rate converted output samples can be fed to a noise shaper and a D/A section, which generates an analog output signal. This means that the downsample filter is not needed.

Such a D/A converter can handle input signals with a large range of input sampling frequencies. The digital sample-rate converter synchronizes the output sample rate and shapes the time jitter to out-of-band frequencies (out of the audio baseband). The phase errors are minimized by the digital PLL. The jitter components present in the incoming clock signal are filtered with a cut-off frequency of 0.5 Hz (clock recovery).

**Future Work:**

During the course of this investigation, many observations were made about the estimation of power spectrum of random signals as it applies to estimating the characteristics of random signals. In addition to continuing the work on predicting the power spectrum for the purpose of estimating the characteristics of random signals, the methods should be applied to other applications in the future.

In the second part, we have proposed a new method of sample-rate conversion, that is able to deal with all the requirements of audio applications. Furthermore, possible analysis of new method of sample-rate conversion is presented. Determining in detail their performances
differences is part of our future work. It would be useful to study the combination of the digital sample-rate converter and an oversampled D/A converter. The performance and the increase in IC-area should be examined and compared to traditional D/A converters. Furthermore, the possibility of applying the sample-rate converter in the A/D side of an audio system can be investigated. It would be useful to study the discrete spectral peaks in the output spectrum of sigma-delta modulators, because they can, when they fold back into the audio base band, cause a performance degradation of the input signal.