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VLSI Architecture for Zero Frequency Filter

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ABSTRACT

Now a days everyone is using mobile phones. Mobile phones play an important role in communication. While using mobile phones, people face the problem of noise signals. These noise signals are affecting the quality of speech signals. These noise signals maybe produced from the materials or sources such as echoes, crowded places, etc. For example, when we speak over the mobile phone or in any crowded place, the external noise adds up like the noise of the subway, train, car, etc. In real time to remove external noise is a very crucial task. A lot of research has been done in cancelling the background noise reduction. Acoustic noise is a type of noise which can produce unwanted sound and it can be reduced by selecting a higher switching frequency (at the cost of higher inverter losses). So a new technique was introduced called passive noise cancellation that suppresses higher frequency acoustic noise. In order to suppress the lower frequency, the passive techniques require materials that is too bulky and heavy. For these materials, an alternative method is required known as active noise cancellation. This technique does not come without its change as it is used to actively cancel the noise of the world around you to make your audio come clearer. Active noise cancellation separates noise signal and the speech signal is chosen. In order to use active noise cancellation, we use zero frequency filter technique for the cancellation of noisy speech signals. It is used for detecting the regions of glottal activity and in estimating the strength of excitation n each glottal cycle. The main advantage of active noise cancellation is to cancel the random sounds due to repetition in the waveform. ANC is used in the reduction by using a power source. It is best suited for low frequencies. VLSI architecture for zero frequency filter can be used as a voice processor in mobile applications. The existing system has produced a time delay of 220ps. In this paper, we are reducing the time delay into 120ps by using tanner tool software.

Keywords: Active noise cancellation; Zero frequency filtering; Acoustic noise.

1.0 Introduction

Noise is an unwanted signal. In communication systems, noise is an error or undesired random disturbance of a useful information signal. Noise is the summation of unwanted energy from natural and sometimes man-made sources. There are several ways to classify noise but conveniently noise is classified as external noise and internal noise. External noise is defined as the noise that is generated externally due to communication. These are classified as atmospheric noises, industrial noises and terrestrial noises. Internal noise is generated within the communication system or in the receiver. These can be classified as low frequency, high frequency and thermal noises. Zero frequency filter

is the signal that has zero frequency. DC signal is filtered by using zero frequency filter which is used in active noise cancellation method. Here we are using adders to decrease DC noises. Some of the DC noises are thermal, shot, flicker, burst and transit time noises.

Active noise cancellation or active noise control or active noise reduction is a method for reducing the unwanted sound specifically designed to cancel the first. It works by using microphones to pick up low frequency noise and neutralise it before it reaches the ear. In this project we have taken active noise cancellation method as it can reduce noise that has low frequencies as well as high frequencies, whereas passive noise cancellation can reduce only high frequencies.

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The main aim of active noise cancellation is to generate an anti- noise signal from ambient noise to reduce the noise level of the original signal.

2.0 Literature Survey

Speech is the output of time varying local tract system excited by a time varying excitation. In resulting speech signal, the information of speech production system is embedded as relations in the sequence of values of signal at different instants of sampling signal. The main objective of speech signal processing is to extract information of time varying characteristics of speech production system [1]. The information is represented in the form of parameters or features derived from the signal. Clock tree synthesis is a fundamental step in physical design. If targeting global zero skew it would cost in area or power and limit achievable operating frequency. Data path optimization is not sufficient to handle timing violations [2].

A very straight forward approach in noise cancelling is the use of LMS algorithm [3] which was developed by Windrow and Hoff. This algorithm uses a gradient descent method that finds a minimum, if it exists by taking steps in the direction negative of gradient and it does by adjusting filter coefficients in order to minimize error. The speech signals collected simultaneously using pair of spatially separated microphones in real room environment. The spatial separation results in fixed time delay of arrival of speech signals from given speaker at a pair of microphones [4]. The time delays are estimated for exploiting impulse like characteristic of excitation during speech production. Zero frequency filtering is a technique used in the characterization and analysis of glottal activity from speech signals. IIR filters followed by two successive finite impulse response filters [5]. The integrated variation analysis technique is considered for the effects of both systematic and random variation in interconnect and devices to get better perspective on the impact of variation [6]. By exploiting useful clock skew, it can help reduce timing violations rapidly. Multi-corner multimode design flow can analyse over MCMM scenarios concurrently. It employs new analysis data model which advantages data duplication across various timing graphs. To solve this MCMM problem, chip level clock tree synthesis is used to reduce divergence between IPS

for different corners. All papers [7-11] all are related to same topic of clock skew optimisation.

Engineering change order (ECO) is always used after detail routing in order fix violations of time by incremental adjustment [11]. clock scheduling is a problem in several work studies in [12-14]. But in all case the problem is clock optimisation period. Both this work is considering the delay but not the speeding up of the clock arrival. A novel clock tree re-synthesis methodology is presented to trickle all these issues. Instead of estimating clock schedule at the leap level registers, the author considered offsets in clock arrival at the clock tree driver pins of any designs which are already synthesized and routed clock tree.

In adaptive filters various algorithms are available such as LMS, NLMS for active noise cancellation [16-17]. Namely the glottal flow is the major source of excitation to convey the useful complementary information. Glottal flow is the air flow passing through the folds at the glottis. Hilbert envelop method, group delay method are used for extracting the glottal activity from speech signals in the papers such as [18-21].

Significant excitation of the vocal track system during production of speech is the Epoch the most challenging task is the extraction of epoch from speech signals.

3.0 Existing System

In this Zero frequency filtering technique is used to filter out the necessary instants of excitation those are not affected by time varying resonances from vocal track. Voiced and Non-Voiced detection involves identifying the region of speech if there is a significant glottal activity [23]. The speech signal is the combination of both voiced and non-voiced or the background voice.

In this it target the zero skew which limits the achievable operating frequency to the maximum data path delay in the circuit. Useful clock skew means the clock is skewed intentionally then that is called as useful clock skew.

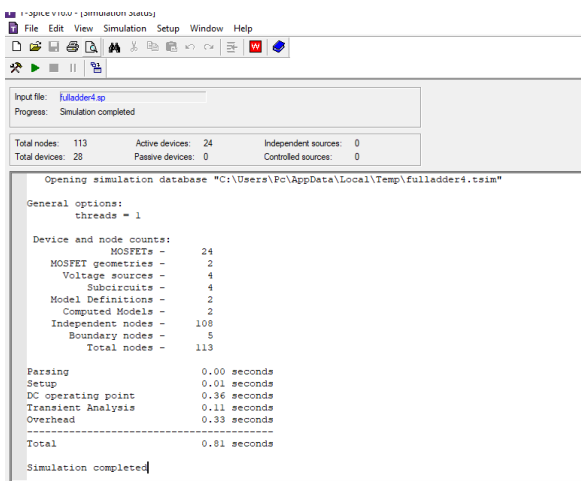
The timing slack is increased on the critical path, the timing can be met with impossible timing constraints, such as the input delay that is more than the time of a clock cycle. This can be done by skewing the receiving flip-flop as it does not create a

Figure 4: Input and Output Components Used in the Filter



Figure 4, shows the details about the input and output components details these details are produced in the simulation time only.

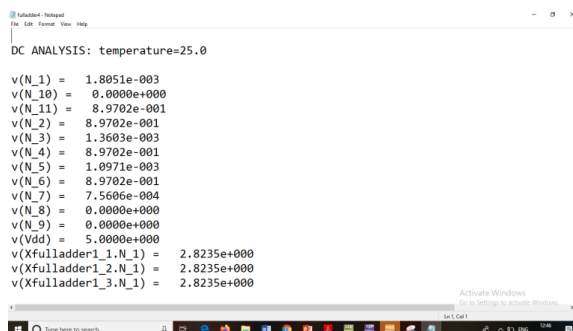
Figure 5: During Simulation Components Details



In all the components used in the circuit details can be obtained. And number components are used also obtained and delay results of every component in the circuit diagram

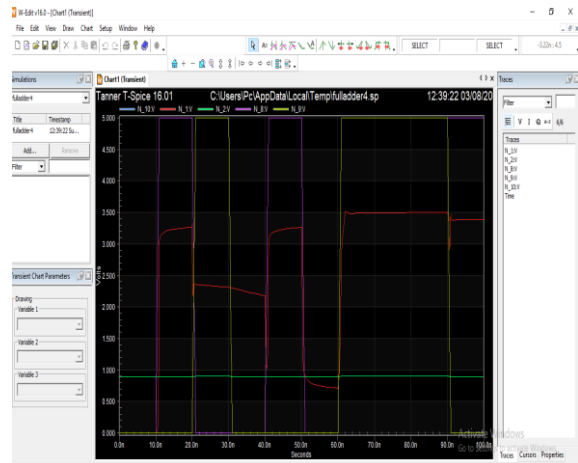
5.0 Results

Figure 6: Dc Analysis



The figure 6, shows the results of dc analysis, it can shows the temperature of each component and whole circuit also. These all results are produced in another folder.

Figure 7: Output Waveforms



The output waveforms are generated with decreasing of delay with approximately 125ps delay and the input frequency 800MHz.

Figure 8: Power Analysis

q1nv	25.2909T	31.1980T	170.4420E-030	31.191T
qs	-12.6475f	-21.6262f	-270.6274e-033	-21.630f
* VOLTAGE SOURCES				
	24	25	26	27
	VW4	VW1	VW2	VW3
VOLTAGE	5.0000	0.	0.	0.
CURRENT	-6.3322	8.0493f	3.1585	16.7834
POWER	-31.6609	0.	0.	0.

The Voltage, Current and power can be obtained in the folder after simulation process can be completed.

6.0 Conclusions

In the speech processing the most frequent problem is the consequence of interference noise in speech signals. To tackle this issue, an active noise cancellation architecture based on Zero frequency filter is realized and non voiced classification can be done. The functionality is verified in the Tanner tool with an operating frequency of 800MHz. In this Physical design process delay is achieved through

using the adder circuits. This adders circuit resulted the delay overhead of about approximately 125ps.

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