"I declare that I have read this report and in my opinion, it is suitable in terms of scope and quality for the purpose of awarding a Bachelor Degree of Electronic Engineering (Computer Engineering)"

Signature : ....
Supervisor : MUHAMMAD SYAHRIR BIN JOHAL
Date : 5 MAY 2006

DIGITAL RECEIVER

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This Report is Submitted in Partial Fulfillment of Requirements for The Bachelor Degree of Electronic Engineering (Computer Engineering)

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MAY 2006

"I admitted that this report is my own work except for sentences or phase that I have stated its sources".

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Special to my family, supervisor, lecturer and friends.

ABSTRACT

The purpose of this project is to design a digital receiver for digital audio broadcasting. DAB is currently being implemented and exploited in many parts of the world and in Asia, the first DAB service was launched in Nov 1999 by MediaCorp Radio Singapore. Some of the major international corporations developing DAB products include Sony, Blaupunct, Kenwood and Pioneer. This is because they foresee the large demand of these products by the public in the future. Until now, analogue radio signals have been subject to numerous kinds of interference on their way from the transmitter to radio. So, how about Malaysia? Stay with analogue broadcasting? No, Malaysia should not be left behind and speeds up to new technologies. So, the objective of this project is to design a digital receiver. The receiver will compare its performance in certain situation and will also be compared with analog receiver. At the end of this project, some of objectives have been achieved.

ABSTRAK

Projek ini bertujuan untuk merekabentuk satu penerima digital bagi penyiaran audio digital. Penyiaran audio digital telah dilaksanakan di banyak negara dan di Asia, penyiaran audio digital telah dilancarkan pada Nov 1999 oleh MediaCorp Radio Singapura. Beberapa syarikat antarabangsa juga sedang memajukan produk penyiaran audio digital ini seperti Sony, Blaupunct, Kenwood dan Pioneer. Ini kerana mereka melihat permintaan yang tinggi oleh pengguna terhadap produk ini pada masa hadapan. Sehingga sekarang, penyiaran audio analog bergantung kepada beberapa jenis gangguan yang wujud semasa proses penghantaran isyarat daripada penghantar kepada radio. Oleh itu, bagaimana dengan Malaysia? Adakah ,masih dengan penyiaran audio analog? Tidak, Malaysia tidak seharusnya ketinggalan dan maju ke arah teknologi baru ini. Oleh itu, tujuan projek ini adalah untuk merekabentuk satu penerima digital. Penerima ini akan dibandingkan prestasinya dalam beberapa keadaan termasuk dibandingkan dengan penerima analog. Di akhir projek ini, beberapa objektif telah dicapai.

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LIST OF ABBREVIATIONS

ADC	Analog To Digital Conversion
AM	Amplitude Modulation
AWGN	Additive White Gaussian Noise
BPSK	Binary Phase-Shift Keying
DAB	Digital Audio Broadcasting
DAC	Digital To Analog Conversion
DDR	Digital Drop Receiver
DDS	Digital Frequency Synthesizer
DSP	Digital Signal Processing
FFT	Fast Fourier Transform
FM	Frequency Modulation
FIR	Finite Impulse Response
IF	Intermediate Frequency
ISI	Inter-Symbol Interference
IFFT	Inverse Fast Fourier Transform
MPSK	M-ary Phase-Shift Keying
NCO	Numerically Controlled Oscillator
NRZ	Nonreturn-to-zero
OFDM	Orthogonal Frequency Division Multiplexing
QPSK	Quadrature Phase-Shift Keying
QAM	Quadrature Amplitude Modulation
SFNs	Single Frequency Networks
RF	Radio Frequency

CHAPTER I

INTRODUCTION

DAB is currently being implemented and exploited in many parts of the world. Many radio stations in DAB centers such as Toronto, Montreal and Vancouver are already broadcasting and promoting DAB right now. The first commercial DAB service in Asia was launched in Nov 1999 by MediaCorp Radio Singapore. Since then, many other broadcasters in the Asia-Pacific region have put DAB services on-air, including broadcasters from Australia, Brunei, China, Hong Kong, India, Korea and Taiwan. Although some of them are still under trial services, it will be just a matter of time before these broadcasters start their commercial DAB services, as the business case of DAB becomes more apparent. It is in a radio station's best interest to start getting up to speed with the technology now.

1.1 Problem Statement

Some of the major international corporations developing DAB products include Sony, Blaupunct, Kenwood, Bosch, Pioneer, Alpine, Grundig and so on. There are also many smaller companies that are developing DAB receivers because they foresee the large demand by the public. Governments of countries such as Singapore really push industry to develop low cost receivers for sale to the public. So, how about Malaysia? Stay with analogue broadcasting? No, Malaysia should not be left behind and speeds up to new technologies.

Until now, analogue radio signals have been subjected to numerous kinds of interference on their way from the transmitter to radio. These problems were caused by mountains, high-rise buildings and weather conditions. Listeners in rural areas know about the noisy reception that can happen when they try to tune into a radio station from any major city more than fifty kilometers away. DAB broadcasts stay crystal clear right to the end of the transmitter's coverage area. Listeners in the cities have not been immune from bad reception on the radio. Interference from lighting in offices, computers, and other electric and electronic devices cause noise on radios. Digital audio radio broadcasts rejects the interference that today's analog radio is susceptible to. So, DAB have better performance and quality than analogue broadcasting. To overcome the problems statement above and receive signal from DAB, we must design a digital receiver.

1.2 Objective

The objective of this project is to design a digital receiver. The receiver will receive signal from digital transmitter and will function in DAB system.

Second objective is to compare performance of digital receiver within the aspect of:

1. Without noise.
2. With noise (AWGN)
3. With noise (AWGN) and Rayleigh Fading.

Third objective is to compare performance between analog receiver and digital receiver.

1.3 Scope Of Work

The scopes of the project involve modeling and simulation of a digital receiver. That includes demodulation process, filtering and so on at the receiver. The scope will cover digital transmitter and receiver.

The scope also includes analog transmitter and receiver, which will be used to compare the performance between the digital receiver and analog receiver.

CHAPTER II

LITERATURE REVIEW

2.1 Historic Background

The new digital radio system DAB (Digital Audio Broadcasting) is very innovative and universal multimedia broadcast system which will replace the existing AM and FM audio broadcast services in many parts of the world in future. It was developed in the 1990s by the Eureka 147/DAB project. DAB is very well suited for mobile reception and provides very high robustness against multipath reception. It allows use of single frequency networks (SFNs) for high frequency efficiency.

Besides high-quality digital audio services, DAB is able to transmit programme-associated data and a multiplex of other data services (travel and traffic information, still and moving pictures). A dynamic multiplex management on the network side opens up possibilities for flexible programming.

In several countries in Europe and overseas broadcast organizations, network provider and receiver manufacturers are going to implement digital broadcasting services using the DAB system in pilot project and public services.

DAB work very differently from convention broadcasting systems. Most of the system component such as perceptual audio coding, channel coding and modulation, multiplex management or data transmission protocols are new solutions and typically not so familiar to the expert in existing analogue or digital broadcast system.

2.2 Analog Receiver

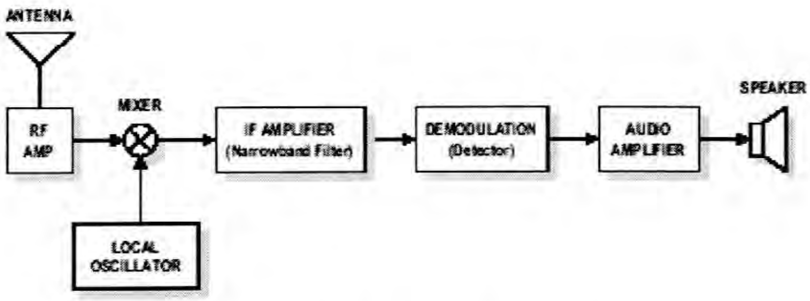

Figure 2.1: Analog Receiver Block Diagram

The conventional heterodyne radio receiver as seen in figure 2.1 has been in use for nearly a century. Let's review the structure of the analog receiver so comparison to the digital receiver becomes apparent. First the RF signal from the antenna is amplified, typically with a tuned RF stage, which amplifies a region of the frequency band of interest. This amplified RF signal is then fed into a mixer stage. The other input to the mixer comes from the local oscillator whose frequency is controlled by the tuning knob on the radio. In figure 2.2, the mixer translates the desired input signal to the intermediate frequency (IF). The IF stage is a bandpass amplifier which only lets one signal or radio station through. Common center frequencies for IF stages are 455 kHz and 10.7 MHz for commercial AM and FM broadcasts. The demodulator recovers the original modulating signal from the IF output using one of several different schemes. For example, AM uses an envelope detector and FM uses a frequency discriminator. In a typical home radio, the demodulated output is fed to an audio amplifier and then to a speaker.

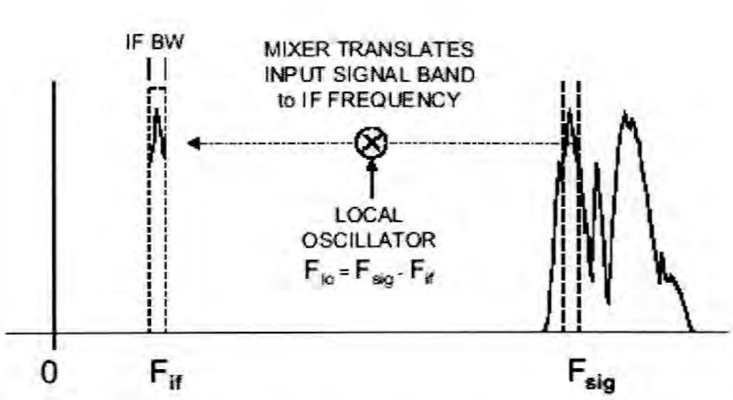

Figure 2.2: Analog Receiver Mixing

The mixer performs an analog multiplication of the two inputs and generates a difference frequency signal. The frequency of the local oscillator is set so that the difference between the local oscillator frequency and desired input signal (the radio station you want to receive) equals the IF. For example, if we wanted to receive an FM station at 100.7 MHz and the IF frequency is 10.7 MHz, you would tune the local oscillator to: 100.7 - 10.7 = 90 MHz. This is called "down conversion" or "translating" since a signal at a high frequency is shifted down to a lower frequency by the mixer. The IF stage acts as a narrowband filter which only passes a "slice" of the translated RF input. The bandwidth of the IF stage is equal to the bandwidth of the signal (or "station") that you are trying to receive. For commercial FM, the bandwidth is about 100 kHz and for AM it is about 5 kHz. This is consistent with channel spacing of 200 kHz and 10 kHz, respectively.

2.3 Digital Receiver

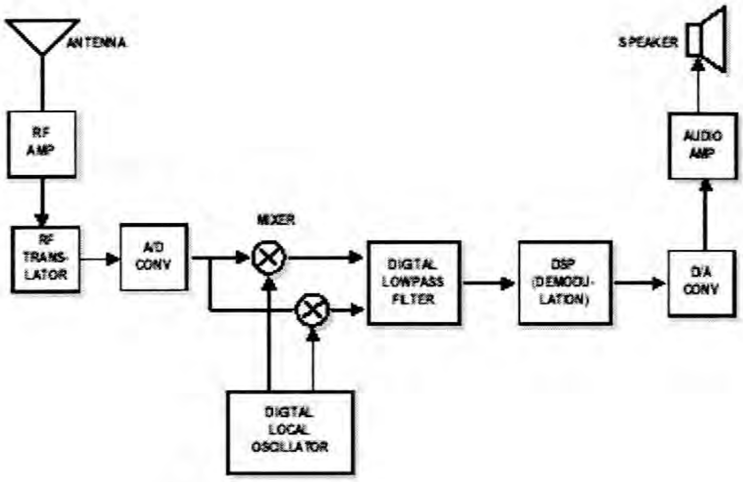

Figure 2.3: Digital Receiver Block Diagram

At the digital receiver, all of the basic principles of analog receivers still apply. Right after the RF amplifier and an optional RF translator stage, we use an analog-to-digital converter (ADC) to digitize the RF input into digital samples and all of the subsequent mixing, filtering and demodulation are performed using digital signal processing (DSP) elements. First, review a theorem fundamental to sampled data which lays the foundation for the ADC requirements.

Nyquist's Theorem:

"Any signal can be represented by discrete samples if the sampling rate is at least twice the bandwidth of the signal."

For example, if we use an ADC sampling at 70 MHz, then the bandwidth of the analog input must be less than 35 MHz. Now let's see what happens if we ignore Nyquist's criterion.

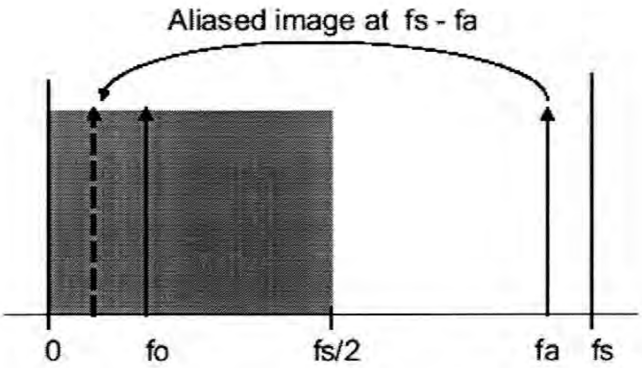

Figure 2.4: Aliasing

Figure 2.4 shows a frequency display of a system being sampled at frequency fs. For all input signals below fs/2, such as the one at fo, we fully meet the Nyquist criterion. In fact, any number of signals can be present in the shaded region and all will be correctly represented in the sampled data. But if we have a signal present at say, fa, which is above fs/2, the sampling process will generate an aliased image which will appear in the sampled data at fs - fa. This image cannot be distinguished from a true signal which might have been present at that same frequency. The point is that once an aliased image is created in the sampling process, no amount of further processing can distinguish between a true signal and an aliased signal. Therefore, it is imperative to prevent aliasing before it occurs.

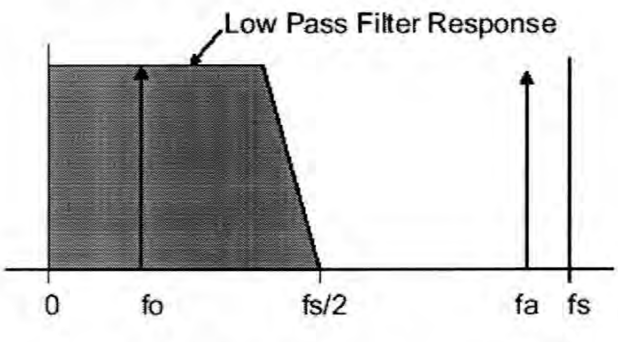

Figure 2.5: Anti-Aliasing Filter

The most straightforward way to prevent aliasing is to use a low pass filter before the ADC which removes all signals above fs/2. This filter is called an anti-aliasing filter. Now the signal at fa is blocked so the ADC never sees it.

Local Oscillator is a direct digital frequency synthesizer (DDS) sometimes called a numerically controlled oscillator (NCO). This device is implemented entirely with digital circuitry. The oscillator generates digital samples of two sine waves precisely offset by 90 degrees in phase, creating sine and cosine signals. It uses a digital phase accumulator and sine/cosine look-up tables.

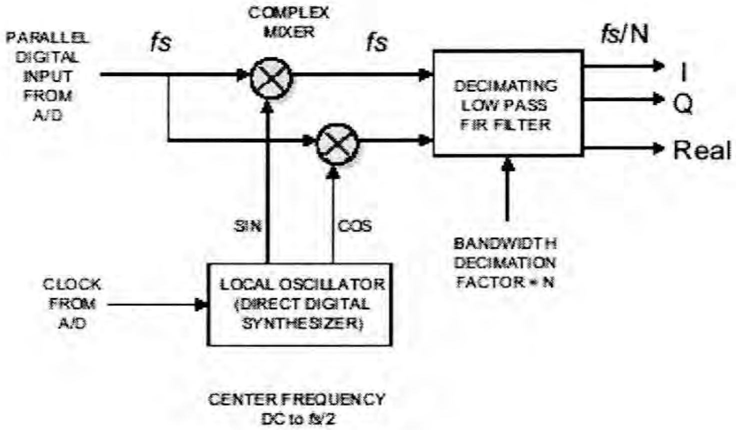

Figure 2.6: Digital Mixer

The Mixer actually consists of two digital multipliers. Digital input samples from the A/D are mathematically multiplied by the digital sine and cosine samples from the local oscillator. Note that the input A/D data samples and the sine and cosine samples from the local oscillator are being generated at the same rate, namely, once every A/D sample clock. Since the data rates into both inputs of the mixers are the A/D sampling rate, fs, the multipliers also operate at that same rate and produce multiplied output product samples at fs. The sine and cosine inputs from the local oscillator create I and Q (in-phase and quadrature) outputs that are important for maintaining phase information contained in the input signal. From a signal standpoint, the mixing produces a single-sideband complex translation of the real input.

Unlike analog mixers which also generate many unwanted mixer products, the digital mixer is nearly ideal and produces only two outputs: the sum and difference frequency signals.

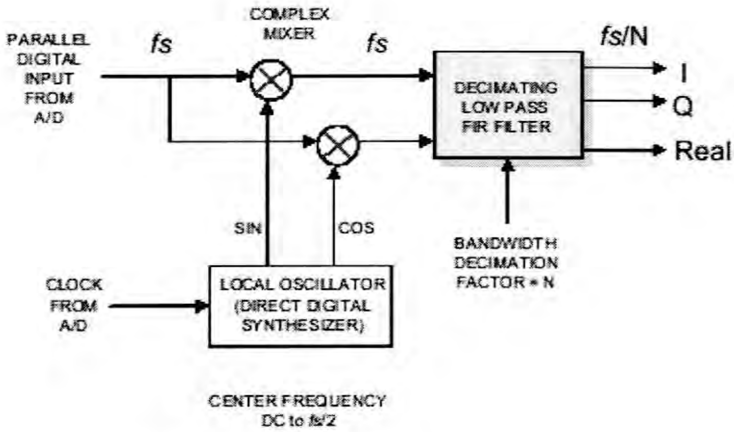

Figure 2.7: Decimating Low Pass Filter

The decimating low pass filter accepts input samples from the mixer output at the full A/D sampling frequency, *fs*. It utilizes digital signal processing to implement a FIR (finite impulse response) filter transfer function. The filter passes all signals from 0 Hz up to a programmable cutoff frequency or bandwidth, and rejects all signals above that cutoff frequency. This digital filter is a complex filter which processes both I and Q signals from the mixer. At the output you can select either I and Q (complex) values or just real values, depending on your system requirements.

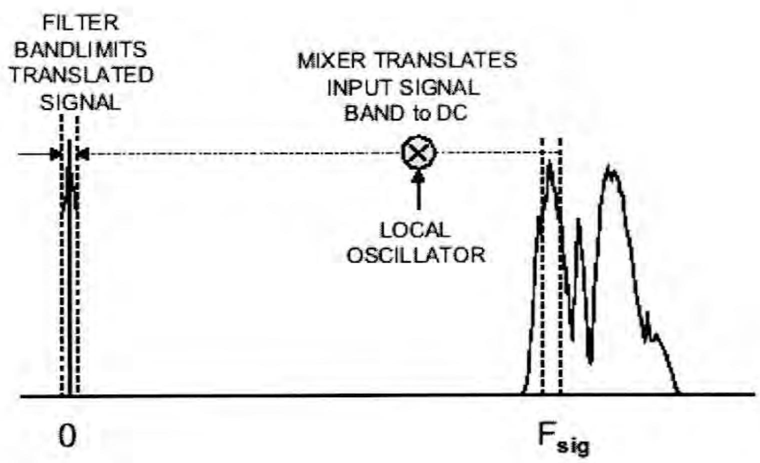

Figure 2.8: Decimating Filter Bandlimiting

Figure 2.8 shows a representation of the action of the filter in the frequency domain. The filter passes only signals from 0 Hz up to the filter bandwidth. All higher frequencies have been removed. But, the wideband input signal was translated down to DC by the mixer and positioned around 0 Hz by the tuning frequency of the local oscillator. Now at the filter output, we have effectively selected a narrow slice of the RF input signal and translated it to DC. Note that we have blocked all other signals above and below the band of interest.

The bandlimiting action of the filter is analogous to the action of the IF stage in the analog receiver except that the decimating low pass filter operates around DC instead of being centered at an IF frequency.

$$\textbf{Output Bandwidth} = \frac{\textbf{Input Sample Rate}}{\textbf{N}}$$

$$\textbf{Complex Output Samp. Rate} = \frac{\textbf{Input Sample Rate}}{\textbf{N}}$$

$$\textbf{Real Output Sample Rate} = \frac{\textbf{2 * Input Sample Rate}}{\textbf{N}}$$

Figure 2.9: Decimation Factor = N