EECS-495 FINAL PROJECT: SOFTWARE RADIO

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Introduction

FM radios and Digital Signal Processors are typically implemented in analog hardware. But, by using the following design methods, they can be implemented using digital components as well.

Background

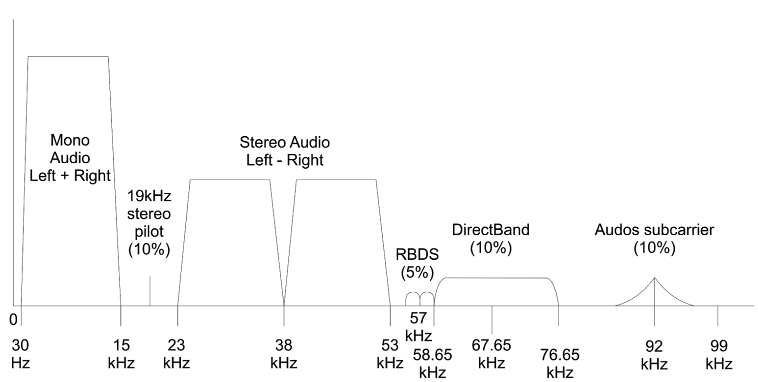
Sampling: A continuous signal can be converted to a discrete-time signal by sampling the input. This is the process of reading the input at certain time intervals. Some data may be lost due to sampling. When the signal is converted back, there may be some variations to the input signal. The sampling rate determines the amount of data in the discrete time signal.
Aliasing is an effect that causes different signals to become indistinguishable (or aliases of one another) when sampled. It also refers to the distortion or artifact that results when the signal reconstructed from samples is different from the original continuous signal.
Sampling theorem: A continuous time signal can be represented in its samples and can be recovered back when sampling frequency fs is greater than or equal to the twice the highest frequency component of message signal.

Quantization: In mathematics and digital signal processing, quantization is the process of mapping input values from a large set (often a continuous set) to output values in a (countable) smaller set, often with a finite number of elements. Rounding and truncation are typical examples of quantization processes. Quantization is involved to some degree in nearly all digital signal processing, as the process of representing a signal in digital form ordinarily involves rounding. Quantization also forms the core of essentially all lossy compression algorithms.
The difference between an input value and its quantized value (such as round-off error) is referred to as quantization error. A device or algorithmic function that performs quantization is called a quantizer. An analog-to-digital converter is an example of a quantizer.
Fixed point numbers have three components: a sign bit, integer bits, and fractional bits. As expected, the sign bit determines if the number is positive or negative, the integer bits determine the value of the mantissa, and the fractional bits determine the value of the fraction. Fixed point numbers can be quantized by left shifting by the number of fractional bits and dequantized by shifting right by the number of fractional bits. The total bit width of the number and the fractional bit width determine the overall accuracy of quantized fixed point operations, as well as the overall hardware cost to support said bit widths.

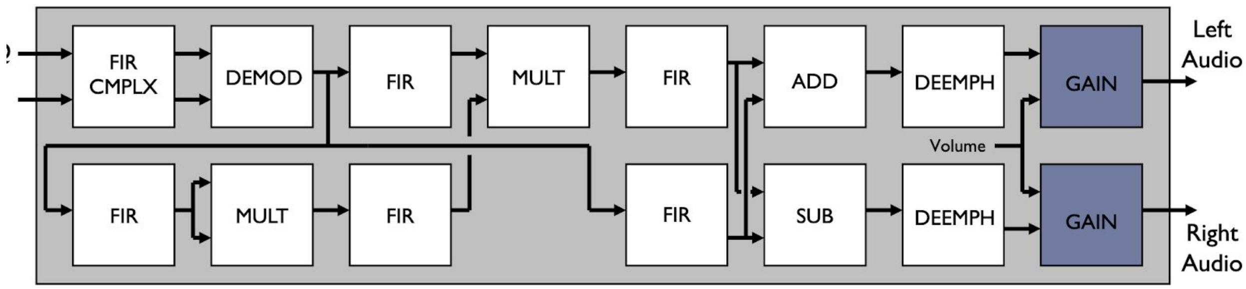
Filtering: A filter is a device or process that removes some unwanted components or features from a signal. Filtering is a class of signal processing, the defining feature of filters being the complete or partial suppression of some aspect of the signal. Most often, this means removing some frequencies or frequency bands. However, filters do not exclusively act in the frequency domain; especially in the field of image processing many other targets for filtering exist. Correlations can be removed for certain frequency components and not for others without

having to act in the frequency domain. Filters are widely used in electronics and telecommunication, in radio, television, audio recording, radar, control systems, music synthesis, image processing, and computer graphics.
Common filters include low pass, high pass, and band pass. Low pass filters filter out data at a frequency higher than some cutoff frequency and keep the low values. High pass filters are the opposite, keeping high frequency data and removing low frequency data. Band pass filters are a combination of low and high pass filters, used to select a range of frequencies. Typically, a filter would be implemented with a combination of capacitors, resistors, and inductors, but it is possible to implement them digitally.
Finite impulse response filters comprise of a history of input samples, each scaled by some coefficient, then summed to produce an output. Digitally, the input history can be created using a shift register. FIR filters can be used to simulate low pass, high pass, and band pass filters.
Infinite impulse response filters are similar to FIR filters, but they also keep a history of previous outputs as feedback. The sum of the previous inputs and previous outputs form the new output. IIR filters can be used to deemphasize a signal, improving the signal-to-noise ratio, or SNR. The transfer function used to deemphasize in this design is $H(s) = 1/(1+s)$.

FM Radio:
Frequency modulation or FM is a form of modulation which conveys information by varying the frequency of a carrier wave; the older amplitude modulation or AM varies the amplitude of the carrier, with its frequency remaining constant. With FM, frequency deviation from the assigned carrier frequency at any instant is directly proportional to the amplitude of the input signal, determining the instantaneous frequency of the transmitted signal. Because transmitted FM signals use more bandwidth than AM signals, this form of modulation is commonly used with the higher (VHF or UHF) frequencies used by TV, the FM broadcast band, and land mobile radio systems.
The maximum frequency deviation of the carrier is usually specified and regulated by the licensing authorities in each country. For a stereo broadcast, the maximum permitted carrier deviation is invariably ±75 kHz, although a little higher is permitted in the United States when SCA systems are used. For a monophonic broadcast, again the most common permitted maximum deviation is ±75 kHz. However, some countries specify a lower value for monophonic broadcasts, such as ±50 kHz.



Design Architecture: The design is a streaming architecture consisting of ten different types of functional units and fifteen units in total. Each functional unit is connected by a FIFO to buffer the data, bringing the overall component count to thirty-five. The components can be grouped into four broad sections: input, pilot determination, left branch, and right branch. Each arrow in the diagram indicates a FIFO



Input: The input section involves reading, splitting, filtering, and demodulating samples to set up for the rest of the design. The functional units are an IQ-reader, complex FIR, and a demodulator respectively. The IQ reader takes input samples and splits them into an I and Q component. The complex FIR treats each I/Q pair as a complex number then filters them. The demodulator is used to find the phase change between I/Q samples to determine the original frequency, which in turn determines pitch of the sound. The output of the demodulator is piped to the other three sections of the design for further processing.

Pilot Determination: This is a small section used to determine the pilot tone. The demodulated signal is passed through a 32-tap FIR band pass filter to isolate the 19 kHz tone. If the tone exists, it indicates that stereo data is available in the 23 kHz to 53 kHz range. The tone can be squared to use as a reference point at 38 kHz, splitting the stereo range into two components. The act of squaring also creates a 0 Hz tone that is removed with a second FIR as a high pass filter.

Left Branch:

The left branch is used to generate the left component of stereo audio. The branch begins with a 32-tap FIR band pass filter to isolate the 23 kHz to 53 kHz stereo audio range. The band is multiplied by the squared pilot tone to shift the frequencies down by 38 kHz, thereby centering the band on 0 Hz and making further calculations on the band easier to perform without offset. The result is passed through a decimating low pass FIR to isolate the stereo component from the rest of the data and to reduce the overall sampling rate by a factor of the decimating constant. The stereo data is added to the mono sound data generated by the right branch at an adder-subtractor component that joins the two branches temporarily. The addition isolates the left audio, which is deemphasized by passing it through an IIR filter. The filter result is passed through a final gain component, which multiplies the signal by a constant to change the volume level.

Right Branch:
The right branch is similar to the left branch. The demodulated input signal is put through a decimating 32-tap low pass FIR filter to isolate the mono sound. The stereo sound from the left branch is subtracted the mono sound to isolate the right audio. As done in the left branch, the audio is pushed through IIR and gain modules to deemphasize the signal and scale by the proper volume.

Design: A software description of the design was provided. The software demonstrated how each component functioned as well as the basic relationships between components. These components were converted to hardware, maintaining equivalent or better functionality. FIFOs were added to facilitate the flow of data throughout the design, like any streaming architecture.

Optimizations: A few basic optimizations were added to the design to gain benefits over the original software implementation.
Streaming
The design had a streaming architecture. Each data sample is piped through each of the components, one after the other, as opposed to calculating the result of each component on the entire data set, which is typical of software. Streaming allows the design to perform using fewer cycles overall. The software implementation uses at least N cycles per component to calculate the result for N samples, while the streaming implementation uses N closer to N cycles total, plus some setup and drain cycles.
Parallelism
The hardware implementation allows for easier parallelism in design. As mentioned before, the output of the demodulator component is piped to three different branches. Processing on the demodulator output in each branch can be performed concurrently in the hardware implementation and only serially in the software implementation. Parallelism also reduces the overall cycle count.
Resource Utilization:

Analysis & Synthesis Status	Successful - Mon Mar 18 20:36:59 2019
Quartus Prime Version	18.1.0 Build 625 09/12/2018 SJ Lite Edition
Revision Name	radio
Top-level Entity Name	radio
Family	Cyclone IV E
Total logic elements	28,087
Total combinational functions	28,010
Dedicated logic registers	6,483
Total registers	6483
Total pins	104
Total virtual pins	0
Total memory bits	81,920
Embedded Multiplier 9-bit elements	532
Total PLLs	0

Throughput: The throughput of the entire design is about 35-40 Mbps on average.

Bottlenecks:
As the device is not linear, it is prone to experiencing bottlenecks. The overall path from demodulator through pilot to the left branch is significantly longer than the path directly from the demodulator to the right path. Without any form of optimization this leaves the right branch waiting a while for the left branch to catch up before data can be sent through to the output. Unfortunately, the demodulator itself complicates the design, as it contains a divider, which does not run for a set number of cycles. Depending on the input to the demodulator, the divider could run from about 3 to 8 cycles. This variation causes increased waiting in the latter components of the design.

Result: The FM Radio was designed and implemented; the throughput was calculated and verified.