Communication Systems Project Report

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# Introduction

**Task1: What is the difference between a traditional radio and a software-defined radio (SDR)?**

A traditional or typical radio, besides the classic demodulation, performs three other operations:

(1) carrier frequency tuning to select the desired signal,

(2) filter to separate it from others received, and

(3) amplification to compensate transmission losses.

Moreover, an amplification step is commonly placed before the demodulation block to carry the signal to an acceptable level for the demodulator circuitry. After the signal enters through the antenna, it is typically amplified by an RF stage that operates only in the frequencies of interest region. Then, the signal is passed to the mixer which receives the local oscillator contribution by its other input. The local oscillator's frequency is set by the radio's tuning control. The mixer is in charge of translating the signal to the Intermediate Frequency (IF). Typically, the oscillator's frequency is set to a value that ensures that its difference from the desired signal's frequency is equal to the IF. The operation is known as down conversion. The next stage is a bandpass filter that attenuates every signal except a specific portion of the spectrum. The bandwidth of this stage limits the band width of the signal that's being received. At the end, the demodulator recovers the original modulating signal from the IF amplifier's output

A software defined radio differs from a traditional radio in several ways. The biggest difference is in how RF is detected and demodulated. An SDR uses a quadrature sampling detector (QSD) that divides the incoming waveform into an in-phase and quadrature signal. The in-phase signal is the first 90º of the RF sine wave and the quadrature signal is the second 90º segment of the RF sine wave. Advantages of SDR:

* DSP code is not "fixed" in firmware.
* DSP Code is Open source. It is not proprietary.
* DSP hardware can be upgraded easily.
* New radio or operating features are easily implemented with a software upgrade Radio is constantly being improved.
* It never becomes obsolete Single step or conversion from RF to baseband audio
* Low noise due to eliminated multiple IF conversions
* Low distortion - distortion is introduced at every conversion stage
* Does not require roofing filters to improve performance
* 99% of the signal path is entirely in the digital domain SDR receiver.

**Task 2: Block Diagram of RTL-SDR Receiver**

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**ESD protection diode:**

Electrostatic discharge (ESD) is the sudden release of electricity from one charged object to another when the two objects come into contact. If ESD protection is not present in a system, the high voltage of an ESD strike via an interface connection would cause a large current spike to flow directly into the IC, causing damage. To protect sensitive circuitry from electrical overstress failures, ESD protection diodes are connected to each signal line between the interface connector and the IC. In the event of an ESD strike, the ESD diode would breakdown and create a low impedance path that limits the peak voltage and current by diverting the current flow to ground, thereby protecting the IC.

**Variable gain LNA:**

As the first active function block in a receiver front-end design, a low-noise amplifier (LNA) is a key component in that system. High gain and low noise figure for the LNA are essential for receiving and processing weak incoming signals. A variable-gain LNA (VG-LNA) can not only prevent a receiver from entering saturation conditions for relatively large-amplitude input signals, but also can mitigate the linearity requirement of the following mixer and maximize the dynamic range of the overall system.

**Tracking RF filter:**

A passive band pass tracking filter tracks the frequency of an RF input signal received within a radio receiver's tracking range of RF frequencies. A tracking control signal is selectively applied to change the capacitance of the LC circuits in order to shift the filter's frequency characteristics/profile as the filter tracks through the tracking band. The filter attenuates half-IF, receiver IF, and image spurious signals. In particular, the filter substantially attenuates the image spurious signal throughout the tracking frequency range.

**RF Mixer:**

a mixer, or frequency mixer, is a non-linear electrical circuit that creates new frequencies from two signals applied to it. In its most common application, two signals are applied to a mixer, and it produces new signals at the sum and difference of the original frequencies.

**Local oscillator:**

 a local oscillator (LO) is an electronic oscillator used with a mixer to change the frequency of a signal. This frequency conversion process produces the sum and difference frequencies from the frequency of the local oscillator and frequency of the input signal.

**IF filter (intermediate):**

 an intermediate frequency (IF) is a frequency to which a carrier wave is shifted as an intermediate step in transmission or reception. The intermediate frequency is created by mixing the carrier signal with a local oscillator signal  resulting in a signal at the difference of the two.

**Variable Gain IF filter:**

This circuit is a flexible, frequency agile, direct conversion IF-to-baseband receiver. conversion gain reduces the cascaded noise figure. Variable baseband gain is used to adjust the signal level. The bandwidth of this filter can be dynamically adjusted as the bandwidth of the input signal changes. This ensures that the available dynamic range of the ADC that this circuit drives is fully used.

**ADC (8-bit):**

Analog to Digital Converter (ADC) is an electronic integrated circuit used to convert the analogue signals such as voltages to digital or binary form consisting of 1s and 0s. Most of the ADCs take a voltage input as 0 to 10V, -5V to +5V, etc. and correspondingly produces digital output as some sort of a binary number. An 8-bit ADC can assign 2^8=256 levels. The sample rate for an ADC is defined as the number of output samples available per unit time and is specified as samples per second (SPS).

**DSP (Digital Signal Processing):**

The output from the ADC is shifted and divided using mixers including the in-phase component and the quadrature component. These signals are then passed through low pass (LP) filters to remove unwanted high frequency components. the next stage is the down sampling where signal sampling is reduced for easier and efficient management of the data. In other words, down sampling is the process of reducing the sampling rate of a signal. This is usually done to reduce the data rate or the size of the data.

**USB Control:**

DSP output is fed into the USB control Unit which interfaces the hardware part of the radio with the software component. This is fed into a pc where further processing takes place. The USB has a fixed clock which processes at a frequency of 48MHz. this is tuned with the frequency of the pc.

**Receiver Management:**

There is an external unit called the receiver management that looks out for the synchronization. It has its own clock tuned with the control data to ensure smooth data transfers. This reduces the number of induced errors.

**References:**

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<https://patents.google.com/patent/US6453157B1/en>

<https://en.wikipedia.org/wiki/Frequency_mixer>

<https://en.wikipedia.org/wiki/Local_oscillator>

<https://en.wikipedia.org/wiki/Intermediate_frequency>

<https://www.elprocus.com/analog-to-digital-adc-converter/>

<https://m.eet.com/media/1158578/c0895pt2.pdf>

**Task 3: Spectrum analysis using RTL-SDR**

**FM radio**

The radio spectrum is the part of the electromagnetic spectrum with frequencies from 30 hertz to 300 GHz. Electromagnetic waves in this frequency range, called radio waves, are widely used in modern technology, particularly in telecommunication. To prevent interference between different users, the generation and of radio waves is strictly regulated by national laws, coordinated by an international body, the International Telecommunication Union (ITU)

**DAB radio**

DAB is generally more efficient in its use of spectrum than analogue FM radio, and thus can offer more radio services for the same given bandwidth. However, the sound quality can be noticeably inferior if the bitrate allocated to each audio program is not sufficient. DAB is more robust with regard to noise and multipath fading for mobile listening, although DAB reception quality degrades rapidly when the signal strength falls below a critical threshold, whereas FM reception quality degrades slowly with the decreasing signal, providing effective coverage over a larger area.

**Digital TV**

Digital television (DTV) is the transmission of television audio-visual signals using digital encoding, in contrast to the earlier analogue television technology which used analogue signals. At the time of its development it was considered an innovative advancement and represented the first significant evolution in television technology since color television in the 1950s. Modern digital television is transmitted in high definition (HDTV) with greater resolution than analogue TV. It typically uses a widescreen aspect ratio (commonly 16:9) in contrast to the narrower format of an analogue TV. It makes more economical use of scarce radio spectrum space; it can transmit up to seven channels in the same bandwidth as a single analogue channel, and provides many new features that analogue television cannot.

**GSM**

GSM (Global System for Mobile communications) is an open, digital cellular technology used for transmitting mobile voice and data services. GSM differs from first generation wireless systems in that it uses digital technology and Time Division Multiple Access (TDMA) transmission methods. GSM is a circuit-switched system that divides each 200kHz channel into eight 25kHz timeslots. GSM operates in the 900MHz and 1.8GHz bands in Europe and the 1.9GHz and 850MHz bands in the US. The 850MHz band is also used for GSM and 3GSM in Australia, Canada and many South American countries. GSM supports data transfer speeds of up to 9.6 Kbit/s, allowing the transmission of basic data services such as SMS (Short Message Service). Another major benefit is its international roaming capability, allowing users to access the same services when travelling abroad as at home. This gives consumers seamless and same number connectivity in more than 210 countries. GSM satellite roaming has also extended service access to areas where terrestrial coverage is not available.

GSM uses a form of Modulation known as GMSK. Gaussian Minimum Shift Keying, GMSK is a form of modulation based on frequency shift keying that has no phase discontinuities and provides efficient use of spectrum as well as enabling high efficiency radio power amplifiers*.* To generate GMSK modulation filter the modulating signal using a Gaussian filter and then apply this to a frequency modulator where the modulation index is set to 0.5.

Reference:[**https://www.electronics-notes.com/articles/radio/modulation/what-is-gmsk-gaussian-minimum-shift-keying.php**](https://www.electronics-notes.com/articles/radio/modulation/what-is-gmsk-gaussian-minimum-shift-keying.php)

90MHZ FM Radio:

A screenshot of a computer

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100MHZ FM Radio:

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101.4MHZ FM Radio:

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220MHz DAB Radio:

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230MHz DAB Radio:

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480 MHz Digital TV:

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720 MHz Digital TV:

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750MHz Digital TV:

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820MHz GSM mobile:

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870 GSM Mobile:

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950 MHz GSM Mobile:**A screenshot of a computer

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960 GSM Mobile:**A screenshot of a computer

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**Airplane Tracking Using ADS-B Signals**

Automatic dependent surveillance—broadcast (ADS–B) is a surveillance technology in which an aircraft determines its position via satellite navigation and periodically broadcasts it, enabling it to be tracked. The information can be received by air traffic control ground stations as a replacement for secondary surveillance radar, as no interrogation signal is needed from the ground. It can also be received by other aircraft to provide situational awareness and allow self separation. This type of broadcasting is done using pulse position modulation (PPM).

Pulse position modulation is done in accordance with the pulse width modulated signal. Each trailing of the pulse width modulated signal becomes the starting point for pulses in PPM signal. Hence, the position of these pulses is proportional to the width of the PWM pulses.

References:

<https://www.mathworks.com/help/comm/examples/airplane-tracking-using-ads-b-signals.html>

<https://www.mathworks.com/matlabcentral/fileexchange/73286-pulse-width-modulation-and-pulse-position-modulation>

A close up of a map

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ADS-B is a cooperative surveillance technology for tracking aircraft. This technology enables an aircraft to periodically broadcast its position information (altitude, GPS coordinates, heading, etc.) using the Mode-S signalling scheme.

Using this process, we managed to detect a total of 12 aircrafts, 5 of which had transmitted their detailed statistics. These stats included the aircraft ID, the flight ID, the exact latitude and longitude< the altitude, the speed, the climb rate, the orientation and the exact time of detection. using this surveillance technology, we were also able to capture the exact location of the aircraft on a very intuitive map of the region.

**Task 4: FM receiver designing using RTL-SDR and Simulink**

In this task we simulated an FM receiver on MATLAB to catch radio waves using an RTL-SDR Receiver. We set the gain to 45dB and varied the central frequency to catch different broadcasting channels. The receiver passed on the data to the FIR Decimator. The FIR Decimation block resamples the discrete-time input at a rate *K* times slower than the input sample rate, where *K* is the integer value you specify for the Decimation factor parameter. To do so, the block implements a polyphase filter structure and performs the following operations:

1. Filters the data in each channel of the input using a direct-form FIR filter.
2. Down-samples each channel of filtered data by discarding *K*–1 consecutive samples following each sample that is retained.

The FM signal is demodulated and converted into audio output.

Reference: [https://www.mathworks.com/help/dsp/ref/firdecimation.html#:~:targetText=Description,for%20the%20Decimation%20factor%20parameter.](https://www.mathworks.com/help/dsp/ref/firdecimation.html#:~:targetText=Description,for%2520the%2520Decimation%2520factor%2520parameter.)

Block diagram:A screenshot of a social media post

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Signal Received in Communication Systems Lab was distorted due to the presence of signal jammers.A screenshot of a computer

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Samples were then taken elsewhere, and the following results were produced,

Frequency = 88MHZ, gain = 45dBA picture containing indoor

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Frequency = 96MHZ, gain = 45dBA picture containing indoor

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Frequency = 106MHZ, gain = 45dBA picture containing indoor

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**Task 5: QPSK Receiver designing.**

In this task we designed and simulated a QPSK transceiver using MATLAB.

**QPSK Receiver Block Diagram:A screenshot of a cell phone

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The following excerpt details how the transmitter and receiver operate.

**Transmitter**

**Bit Generation**:

Generates the bits for each frame.

**QPSK Modulator**:

Modulates the bits into QPSK symbols.

**Raised Cosine Transmit Filter**:

Uses a roll-off factor of 0.5, and up-samples the QPSK symbols by two.

**AWGN Channel with Frequency Offset and Variable Delay:**

The AWGN Channel with Frequency Offset and Variable Delay subsystem first applies the frequency offset and a pre-set phase offset to the transmit signal. Then it adds a variable delay with a choice of the following two types of delay to the signal:

* Ramp delay: This type of delay is initialized at Delay Start samples and increases linearly at a rate of Delay Step samples in each frame. When the actual delay reaches one frame, the delay buffer is full, and it maintains a delay of one frame.
* Triangle delay: This type of delay linearly changes back and forth between Min Delay samples and Max Delay samples at a rate of Delay Step samples in each frame

The use of multiple delay characteristics allows you to investigate their effects on receiver performance, particularly on the Symbol Synchronizer block. The delayed signal is processed through an AWGN Channel.

**Receiver**

**Raised Cosine Receive Filter**

The Raised Cosine Receive Filter provides matched filtering for the transmitted waveform with a roll-off factor of 0.5.

**AGC**

The received signal amplitude affects the accuracy of the carrier and symbol synchronizer. Therefore, the signal amplitude should be stabilized to ensure an optimum loop design. The AGC output power is set to a value ensuring that the equivalent gains of the phase and timing error detectors keep constant over time. The AGC is placed before the Raised Cosine Receive Filter so that the signal amplitude can be measured with an oversampling factor of two, thus improving the accuracy of the estimate.

The Coarse Frequency Compensation subsystem corrects the input signal with a rough estimate of the frequency offset. The following diagram shows the subsystem, in which the frequency offset is estimated by averaging the output of the correlation-based algorithm of the Coarse Frequency Compensator block. The compensation is performed by the Phase/Frequency Offset block. There is usually a residual frequency offset even after the coarse frequency compensation, which would cause a slow rotation of the constellation. The Carrier Synchronizer block compensates for this residual frequency.

The accuracy of the Coarse Frequency Compensator decreases with its maximum frequency offset value. Ideally, this value should be set just above the expected frequency offset range.

**Symbol Synchronizer**

Resamples the input signal according to a recovered timing strobe so that symbol decisions are made at the optimum sampling instants

The timing recovery is performed by a Symbol Synchronizer library block, which implements a PLL. The timing error detector is estimated using the Gardner algorithm, which is rotationally invariant. In other words, this algorithm can be used before or after frequency offset compensation. The input to the block is oversampled by two. On average, the block generates one output symbol for every two input samples. However, when the channel timing error (delay) reaches symbol boundaries, there will be one extra or missing symbol in the output frame. In that case, the block implements bit stuffing/skipping thus the output of this block is a variable-size signal. The Damping factor, Normalized loop bandwidth, and Detector gain parameters of the block are tuneable.

**Carrier Synchronizer**

The fine frequency compensation is performed by a Carrier Synchronizer library block, which implements a phase-locked loop (PLL), to track the residual frequency offset and the phase offset in the input signal. The PLL uses a Direct Digital Synthesizer (DDS) to generate the compensating phase that offsets the residual frequency and phase offsets. The phase offset estimate from DDS is the integral of the phase error output of a Loop Filter. The Damping factor and Normalized loop bandwidth parameters of the block are tuneable.

**Preamble Detector and Frame Synchronizer**

The location of the known frame header is detected by a Preamble Detector library block and the frame synchronization is performed by a MATLAB System block using a Frame Synchronizer System object™. The Preamble Detector block uses the known frame header (QPSK-modulated Barker code) to correlate against the received QPSK symbols in order to find the location of the frame header. The Frame Synchronizer block uses this location information to align the frame boundaries. It also transforms the variable-size output of the Symbol Synchronizer block into a fixed-size frame, which is necessary for the downstream processing. The second output of the block is a Boolean scalar indicating if the first output is a valid frame with the desired header and if so, enables the Data Decoding subsystem to run.

**Data Decoding**

The Data Decoding enabled subsystem performs phase ambiguity resolution, demodulation and text message decoding. The Carrier Synchronizer block may lock to the unmodulated carrier with a phase shift of 0, 90, 180, or 270 degrees, which can cause a phase ambiguity.  The Phase Ambiguity Correction & Demodulation subsystem rotates the input signal by the estimated phase offset and demodulates the corrected data. The payload bits are descrambled and printed out to the Simulink Diagnostic Viewer at the end of the simulation.

Reference: <https://www.mathworks.com/help/comm/examples/qpsk-transmitter-and-receiver-1.html>

**Spectrum Analysis:**A picture containing indoor, wall

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**Signal after Carrier Synchronization:**A picture containing wall

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**Signal after Symbol Synchronization:**