EE214 ELECTRONIC CIRCUITS LABORATORY
TERM PROJECT
ANALOG BASEBAND TUNABLE RECEIVER

1. INTRODUCTION

In radio communications, utilizing the limited resources has been the most important challenge, and it is predicted to be ever important in the future due to the increasing demand for communications all around the world. Being able to use high frequency devices was a critical step in this utilization because it enabled embedding different information into the same waveform, effectively reducing both the number of antennas required and also their sizes.

Embedding information into a waveform is, in general, called as “modulation”. There are lots of different modulation techniques in the literature and lots of research is being conducted in this area. Collecting different modulated (information carrying) signals into a single waveform is called “multiplexing”. One method of multiplexing is “frequency division multiplexing”, which exploits the fact that different frequency sinusoids are orthogonal to each other. This simply means that by tuning a receiver to a specific frequency, signals with different frequencies can be suppressed, within some physical constraints. This method is widely used in modern communications, and FM radio is a good example. Within the pre-specified FM radio frequency band (around 88 to 108 MHz), up to 100 different information carrying signals (channels) are collected together, therefore 200 kHz bandwidth is allocated for each channel. A well-tuned receiver then adjusts its center frequency and bandwidth so that it receives only the desired channel and suppresses others. After receiving a channel with 200 kHz bandwidth, receivers “demodulate” the information in this channel to get the correct sound signal, whose largest frequency component is generally assumed to be 15 kHz.

2. ABOUT FREQUENCY DOMAIN ANALYSIS

It is known that any periodic waveform can be decomposed into sinusoidal signals with different frequencies and amplitudes. In other words, linear combinations of all orthogonal sinusoidal waveforms cover the whole space of periodic waveforms, and the coefficients of these linear combinations are unique for each periodic waveform. Then, it is possible to represent periodic waveforms with their “frequency-domain” coefficients. Such representations are done via the well-known Fourier Series expansion. A generalization of this expansion (when period goes to infinity, frequency difference goes to zero) which can be applied to aperiodic waveforms as well, is Fourier Transform.

Consider $x(t)$ is a time-domain waveform. Time (t) is its independent variable, and for each time instance, $x(t)$ is its value/amplitude. If you apply Fourier Transform to $x(t)$, another waveform is created, say, $X(f)$. Now, the frequency is its independent variable, and for each frequency, $X(f)$ can be considered to be corresponding to relative amount of signals with frequency f inside $x(t)$. If $x(t)$ does not include and signal with frequency f, $X(f)$ becomes 0. If $x(t)$ includes only N sinusoidal signals with different frequencies, $X(f)$ includes N impulses at those frequencies. If $x(t)$ includes not only discrete-frequency sinusoids but also aperiodic waveforms, $X(f)$ includes both impulses and continuous parts. The continuous frequency axis is usually called frequency spectrum, and $X(f)$ represents the place $x(t)$ occupies in the frequency spectrum. It should be noted that $X(f)$ is, in general, a complex-valued function.

Since the devices in the laboratory are digital, they sample and quantize the waveforms and process them as discrete-time waveforms. The Fourier transform of discrete-time signals is called Discrete-Time Fourier Transform (DTFT). The DTFT of a discrete-time signal is innately periodic in frequency axis, therefore in general, a single period of its DTFT is enough to describe a discrete-time signal. It should be noted that even if the time-domain signal is discrete-time, its DTFT is continuous in frequency axis.

The digital devices cannot process continuous signals, therefore, DTFT of a signal must be quantized in order to be processed digitally. In general, sampled version of DTFT is called Discrete Fourier Transform (DFT) (quantization errors are inherent in sampling, but thanks to Nyquist theorem, it is not necessary to have infinite sampling rate to get zero quantization error). DFT is discrete in both time and frequency axes, so that digital devices can work with DFT. A fast and accurate method to calculate DFT is Fast-Fourier Transform (FFT) algorithm.

The oscilloscopes in the Basic Electronics Laboratory have the capability of measuring FFT, via Math → FFT. They can measure FFT of both channels, as well as a pre-defined math function of the channels (like the sum of them). Only one FFT's magnitude can be plotted on the screen.

Lots of information on Fourier Analysis can be found on the internet. A couple of introductory materials can be found in https://www.thefouriertransform.com/ and https://youtu.be/spUNpyF58BY.

3. PROJECT DESCRIPTION

In this project, you are supposed to make a transmitter which produces different frequency tones (pure sinusoids) and collect them together, and a receiver which can be tuned to these frequencies. The received tone will be played with a speaker.

3.1. TRANSMIT UNIT

Your transmitter unit will produce sinusoidal signals with frequencies 1 kHz, 2 kHz, 3 kHz, 5 kHz, 6 kHz, 7 kHz, 9 kHz, 10 kHz, 11 kHz, 13 kHz, 14 kHz and 15 kHz (4 kHz and its multiples, 8 kHz and 12 kHz are the military frequencies that you are not allowed to generate or transmit for this project). In other words, 12 different tones are going to be produced and collected into a single waveform. The magnitudes of these sinusoids are **not** required to be the same, since in a real scenario, different channels almost always have different magnitudes. However, since you are also supposed to receive the tones, their magnitudes must be large enough to be distinguished from the noise. You are **not** supposed to modulate the signal, producing and collecting the sinusoids are enough. Modulation-demodulation parts in the real scenario will be assumed to be made perfectly. Similarly, you are **not** supposed to radiate the signal through an antenna, the RF radiation-transmission in the real scenario will be assumed to be made perfectly. In other words, the communication channel will be a copper wire, not the air.

3.2. RECEIVER UNIT

Your receiver unit will receive each of the tones separately. The receiver is supposed to be tunable, meaning that changing the circuit connections for selecting different channels will be forbidden. Tuning can be made via a single potentiometer, or multiple ones (keep in mind that as the degrees of freedom increase, tuning becomes harder). It is expected that co-channel interference is suppressed well (at least 10 dB).

Hint: For the filter design you may look up for **"modified"** KHN (Kerwin–Huelsman–Newcomb) filter.

3.3. SPEAKER UNIT

After your receiver unit selects the tone, you will play the tone with a speaker circuit. Since, in general, your circuit will have high output resistance (KΩ-MΩ) compared to the speaker's resistance (8Ω - 50Ω), the speaker can receive only a small portion of the signal. Therefore, you need to design an amplifier to drive your speaker. The amplifier should not saturate and amplifies the signal as linear as possible. You can use BJT amplifiers covered in lab sessions or refer to your research.

4. RULES AND REGULATIONS

4.1 ALLOWED COMPONENTS

You are allowed to use +/-25 V output of DC power supply and you may use any types of resistors, capacitors, inductors, diodes, op-amps, transistors, potentiometers. Speaker must not contain any embedded circuit so that you will drive the speaker with your OWN circuit. You are NOT allowed to use the signal generator in laboratories. You can use the signal generator to get partial credit only when your transmitter unit does not work.

4.2 DESIGN SPECIFICATIONS

In the transmit unit 12 different frequencies (1 kHz, 2 kHz, 3 kHz, 5 kHz, 6 kHz, 7 kHz, 9 kHz, 10 kHz, 11 kHz, 13 kHz, 14 kHz and 15 kHz) are generated.

In the receive unit, the magnitude difference of the desired channel to other channels should be at least 10 dB. (This specification will be checked out using the FFT property of the dıgıtal oscilloscope.)

The channel adjustment should be provided by adjusting at most 2 POTs (More than 2 POTs, depending on the number of POTs, would result in some point loss).

The speaker unit outputs the desired frequency channel at audible levels.

Further details about the project will be announced.

4.3 BONUS

Usıng AGC (Automatic Gain Control) in the speaker unit to provide the same output level for all frequency channels. (15 points)

Neat breadboard design (5 points)

Adjusting channels with using only 1 POT, while providing 10 dB difference from desired frequency channel to other channels. (10 points)

Any other Bonus will be announced later.

4.4 GROUPS

The project will be carried out in groups of two students. The students in the same group should be in the same laboratory session.

4.5 Important Dates

- April 13 : Project Announcement
- April 18-22 : First Project Session
- April 25 : Submission of the pre-report. (until 08:00 AM (morning time))
- June 6-20 : Second Project Session & Demonstrations
- June 25 : Submission of demo-videos (until 08:00 AM (morning time))
- June 25 : Submission of the Final Report (until 08:00 AM (morning time))

4.6 Documentation

You **must** submit two reports and a video for the term project.

4.6.1 Reports

As stated earlier, pre-report should include an introduction, pre-design of the project with circuit diagrams and overall circuit schematic, theory, formulations, simulation results and a conclusion.

Final report should also include all the parts in the first report for the overall design. In other words, it should explain the overall design with an introduction, a block diagram and circuit schematic, operation of each sub-block with theory, formulations, simulation and experimental results. Final report should also include analyses for the cost and power consumption of the project and you should justify the use of each component. Conclusion of the final report is very important since it reflects your understanding of the project and the experiences you gained during the overall process. The objectives, results and the experiences should be clearly presented. This does not necessarily mean a long report, but definitely a well-organized one.

Late submissions for both reports will lower your report grades as:

- %20 off for one-day late submission
- %50 off for two-day late submission
- %90 off for three-day late submission
- Zero credit for more than three-day late submission.

Pre report must be less than 5 pages. Each extra page may result in some point loss.

Final report must be less than 10 pages. Each extra page may result in some point loss.

4.6.2 Demonstration Video

You should prepare a 6-8 minute video where partners of each group present the project in a collaborative manner. The video should include the explanation of main blocks, why they are used and how they are designed. This video should be regarded as a formal presentation to the related assistant. Note that you should always appear in the video together with your presentation material.

4.7 Grading

- Pre Report : %15
- Final Report : %20
- Presentation Video : %5
- Design and Performance : %60 (partial credits are possible)
- Bonus : up to %30

4.8 REGULATIONS

- Attending the project demonstration is a must for both team members, otherwise, you will get zero from Design and Performance and Bonus.
- Fundamental knowledge of the students will be tested at the demonstration.
- Cheating is strongly forbidden and any indication of cheating will cause you to get zero credit from the project. You can collaborate with your friends by exchanging ideas, not copying the design details or the reports.
- Both members of the group are responsible for every single detail of their circuit.

For your further questions about the project, you can contact with Uğur Berkay Saraç (usarac@metu.edu.tr) and İlhan Can Avcu (avcucan@metu.edu.tr)

IMPORTANT NOTE: You need to buy your own circuit equipment (capacitors, breadboard, resistors, POTs, cables etc.) for the project. You can find these circuit components on the Internet or Konya Sokak in Ulus, Ankara. You can use the measurement devices (Digital Multimeter, Oscilloscope) and voltage generators (Signal Generator, DC Power Supply) found in the laboratory.