Deadline submission: ***Saturday 9th of December, 2023***

1. Project statement

It is required to implement the MPEG audio **Layer 2**.

This part of the project is divided into two main sections as shown below figure:

- The MPEG audio Encoder Layer **2**
- The MPEG audio Decoder Layer **2**

Required Tasks

I. The MPEG audio Encoder Layer 2

1. At the beginning, you should read PCM-coded digital audio from standard Windows. WAV file.

2. **Then Design** the essentials of **time-to-frequency mapping and filter bank**. that break the input signal into thirty-two equally spaced frequency sub-bands depending upon the Nyquist frequency of the original signal. (For a more detailed **filter bank**, see chapter 4, Section 4.2 in the book [1])

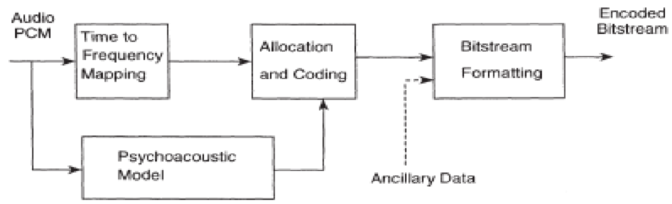

Figure 1. MPEG-1 Audio encoder basic building blocks

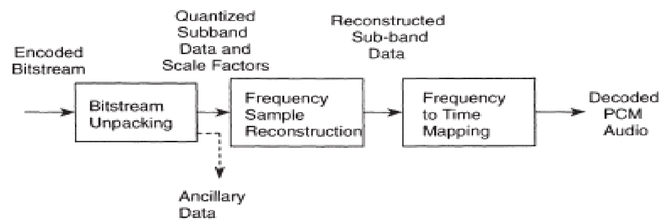

Figure 1: MPEG/Audio Encoder- Decoder

3. After the filter bank, you have to use the **downsampler** which reduces the amount of data required for each sub-band. However, the total amount of data required for the signal remains the same because there are thirty-two sub-bands in all. (For a more detailed **downsampler**, see chapter 4, Sections 1 & 2.2 in the book [1])
   *(For example, if there are 100 original audio samples, the filterbank creates thirty-two subbands, each with 100 samples. This increases the number of samples to 3200 (100 * 32). After downsampling, the number of samples is reduced to 100.)*

4. Simultaneously, a **psychoacoustic model** is applied to **the input signal** where the psychoacoustic analysis stage is performed with a 1024-point FFT (Layer II) to determine which frequency bands should be retained.
   The model takes advantage of **the masking properties** of the human auditory system to identify unneeded components present in the original signal. (see also chapter 11, Section 5 in book [1]) The psychoacoustic model provides data about these spurious components to the last block.
   (You can use the FFT command in MATLAB)

5. Finally, the last block (The Allocation and coding) uses the data provided by the model to determine how to encode the sub-band signals. To achieve data compression, the encoder ignores subbands that are completely masked by other signal components. Additionally, subbands that are partially masked are encoded with less accuracy than the dominant subbands. (see also chapter 11, Section 6 in book [1]) *(Recall that* MPEG Layer 2: Divides data into frames, each of them contains 384 samples, 12 samples from each of the 32 filtered sub-bands. then Use three frames in the filter *(before, current, next, a total of 1152 samples)*. This model has a little bit of temporal masking.)

II. The MPEG audio Decoder Layer 2

To interpret MPEG2 data frames and reconstruct the **original signal**,

6. You have to design the decoder that restores the quantized spectral components of the signal.
7. Finally, reconstructs the time-domain representation of the audio signal from its frequency representation. (Ideally, this signal should be perceptively identical to the original audio signal.)

8. Plotting:

You should provide those four plots on your code:

- Plot Spectral view of input audio signal
- Plot Spectral view of After Filterbank
- Plot Spectral view of After **Psychoacoustic model**
- Plot Spectral view of the reconstructed audio signal

9. **Performance measure**

You should calculate the Compression ratio using the following formulas, which is defined as the ratio of the original signal to the compressed signal.

$$\text{CR} = \frac{\text{Uncompressed Signal}}{\text{Compressed Signal}}$$

10. **Evaluation Test**:

The evaluation Test has both Objective and Subjective evaluation is used to measure the audio quality.

A. Mean Opinion Score (Subjective evaluation): The audience would listen to the sound files and see the sound quality and score it. It has to grades from 5 to 1.

MEAN OPINION SCORE

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly Annoying
2	Poor	Annoying
1	Bad	Very Annoying

2. Project Requirements:

A. Implement the needed **codes** for all the required tasks and include all the MATLAB files in your online submission.

B. The documentation is a **MUST** along with the code itself using MATLAB **Live Editor**.
This is a short video about the MATLAB live editor:
https://www.youtube.com/watch?v=bu4g8ID3aEk

Important Notes:

- ➢ Documentation of your MPEG2 design is a **mandatory** requirement in this part.
  (*The project submitted without documentation will not be accepted).*
- ➢ Documentation should be explained clearly all your used blocks and their corresponding inputs, outputs, internal variables, etc., and how they map to the implemented tasks.
- ➢ The code should be readable and all the variable names are meaningful.
- ➢ The code should be commented to explain the functionality of your code.
- ➢ the relevant outputs should be displayed in your live editor according to the above tasks.

C. Project Logistics:

1. **Each team consists of 3 ~ 5 students.**
2. Any **plagiarized reports or codes**, either fully or partially, will receive **ZERO points**.
   *(This applies to **both** the original and the copy)*
3. Any references used should be properly cited.
4. **No late submissions** are allowed.
5. **Coding Evaluation:** *(Total grade of this assignment is worth **10% of the course grade**)*
   5 points distributed among the 10 tasks mentioned above, with 0.5 point per task. Note that the grade per task includes the documentation of the task. The task grading will be on a level from 0-4.
   - **Level 0:** Task not done
   - **Level 1:** Done with low quality
   - **Level 2:** Done with fair quality
   - **Level 3:** Done with good quality
   - **Level 4:** Done with excellent quality
6. **Individual Discussion:** *(Total grade is worth **5% of the course grade**)*

D. Reference

[1] *INTRODUCTION TO DIGITAL AUDIO CODING AND STANDARDS*. . BOOK