Submission guideline:

1. The weightage of this assignment is 60% of the total marks. The answer script should be typed (preferred) or written clearly (no pencil please), with your name on the first page.
2. Keep your assignment report within 12 pages including codes and all figures.
3. Please submit your assignment report (in a single pdf file only) to NTULearn by Sunday of week 15 (3 May 2020).
4. Please be informed that any form of cheating will be severely penalised.

In this assignment, students are required to go through the lecture notes, and research on relevant topics in the open literature on audio signal processing. Students are required to write a report answering the following questions, with proper discussions. Students are encouraged to come up with relevant examples, applications and Matlab codes for implementing the topics mentioned below. Codes and figures should be included into the assignment report.

1. (a) Global masking threshold has to be used for masking the non-audible components in audio compression. Describe the major steps that are carried out to compute the global masking threshold for a given signal segment.

   (b) Describe how a 12 bit PCM sample is represented by 8 bits with the floating point number system, and point out why quantization errors for large sample values are more likely to be produced than those for small sample values.

(50 Marks)

2. (a) Audio signal is typically time-varying because its frequency content is changing with time. To get more accurate spectrum information from the input audio signal, variable frequency (or temporal) resolution is needed by the audio encoder. Describe how this issue is addressed by MPEG-1 Layer 3 standard for improvement on the performance achieved by MPEG-1 Layers 1 and 2.

   (b) Bit reservoir concept is used to deal with temporal demand of high bit rate bursts of the coded audio data. Describe how this concept is implemented in MPEG-1 audio coding standards.

   (c) An audio coding system consists of a 16-channel filter bank and a modified discrete cosine transform (MDCT). The MDCT filter bank consists of a 24-point MDCT for steady-state signal or an 8-point MDCT for transient-like signal. If the sampling frequency of 24 kHz is used, compute the frequency and time resolution.

(50 Marks)

End of Assignment