Audio and Video Coding (2019/20)

Lab work Nº 2 - Due: 15 Nov. 2019

Audio Coding

1. Implement an entropy encoder using Golomb codes. Start by developing a class `Golomb`, where you should implement, at least, one method to encode numbers (signed integers) and another one to decode them. It should be possible to specify the parameter `m` of the Golomb code.

   You should consider the use of a class `BitStream`, to read/write bits from/to a file, as part of Golomb to read/write data in the encoded file. Recall that this class should have, at least, methods to write one bit, read one bit, write `n` bits and read `n` bits. The resulting file should be binary (not text) and take into consideration that the minimum amount of data that you can access in a file is one byte (8 bits). You can implement other methods that you think might be necessary (for example, methods to read and write strings, in binary). This class should be optimised, due to its extensive usage during compression/decompression.

2. Develop a lossless predictive audio codec, exploring temporal and channel redundancy, followed by Golomb encoding. You should try, at least, simple prediction based on the causal neighbours of the sample being encoded. Remember that the ultimate goal is to attain the largest compression ratio as possible, although compression/decompression time should be reasonable.

3. Include an option for lossy coding in the developed codec, based on residual quantization. You can take into consideration some of the software implemented in Lab work 1. The ultimate goal is to attain the largest compression ratio as possible with the minimum error introduced.

4. Elaborate a report, where you describe all the steps and decisions taken in all the items of the work. If appropriate, include measures of processing time, compression ratios and SNR (for the lossy case). Also, include histograms of both the original audio and of the prediction residuals, as they are important in understanding the properties of the audio signals and helping choosing the best parameters for lossless audio coding.

   The final mark will be calculated based on the best results of compression ratio, processing time and error introduced taking as reference the audio dataset available on elearning.