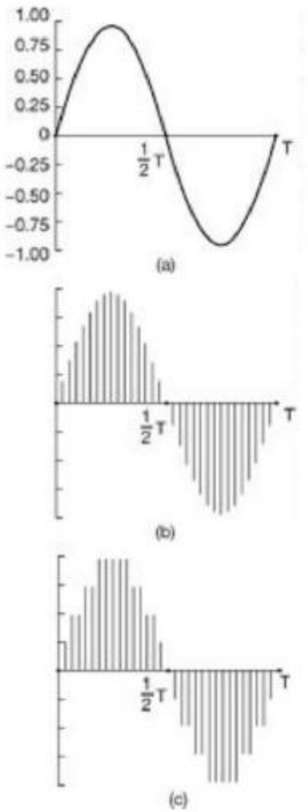
Speech Information Systems



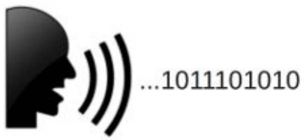
Lab 4

Sampling, Quantization, Speech Coding

Author: Bartalis Mátyás, Ali Raheem Mandeel, and Ibrahim Ibrahimov

Email: aliraheem.mandeel@edu.bme.hu

Academic Year: 2024/2025 – Spring

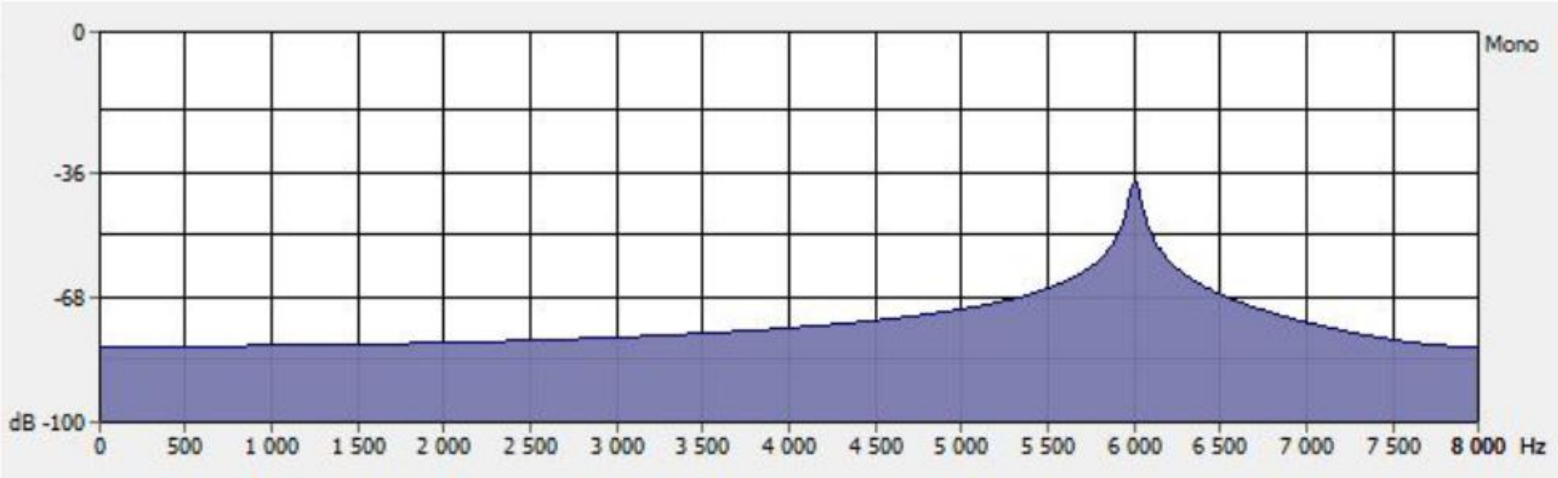


SAMPLING

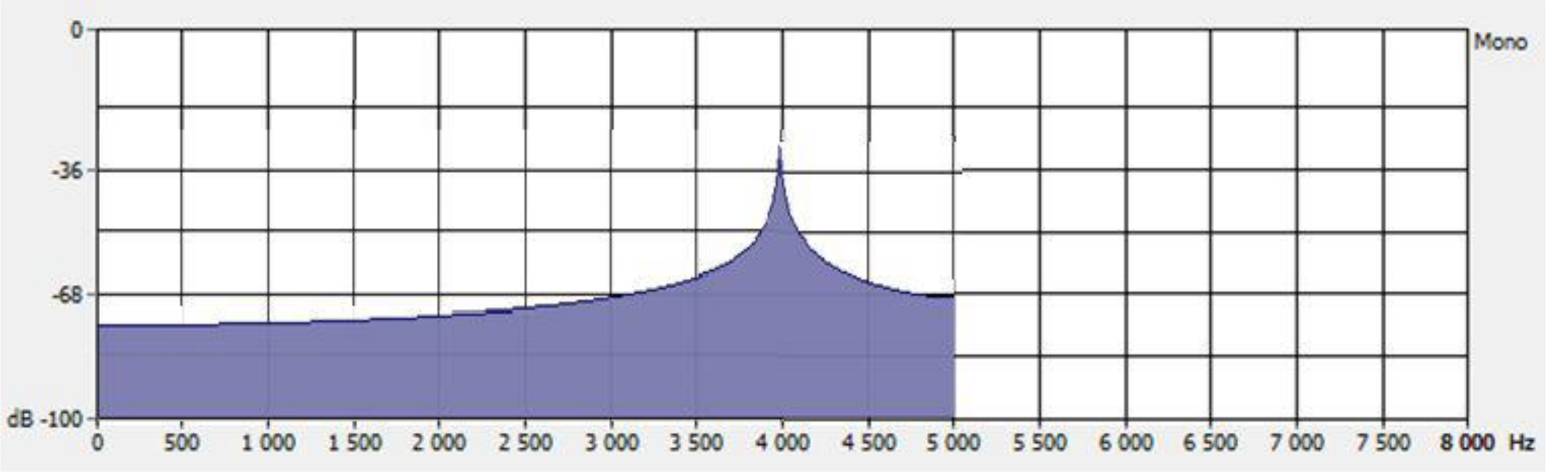
A **6 kHz sine wave** is sampled at **10 kHz** without filtering. Where does the original sine wave appear in the sampled signal?

- 4 kHz
- 16 kHz
- 5 kHz
- 3 kHz
- 2 kHz
- 1 kHz

a) Original Signal



b) Sampled Signal Without an Anti-Aliasing Filter



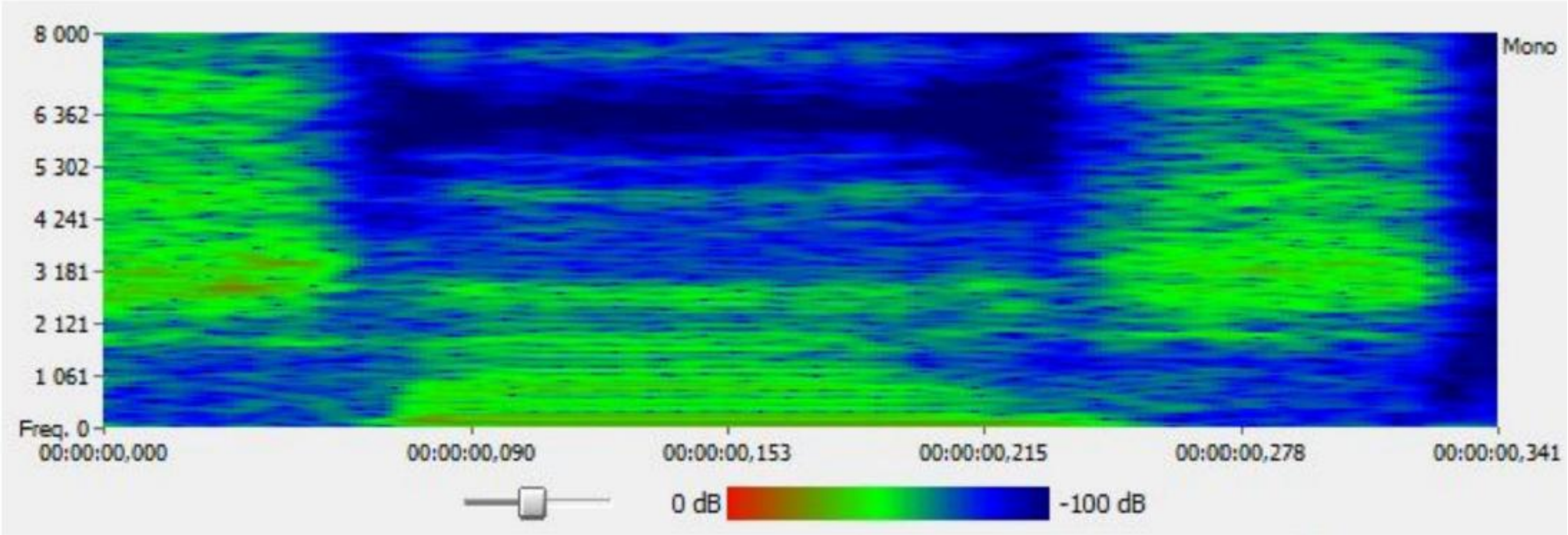
A **sampling-reconstruction system** operates with a **5.5 kHz sampling frequency** without an **anti-aliasing filter** at the input, but with a **high-quality low-pass filter with a cutoff frequency of 2.5 kHz** at the output.

Tasks:

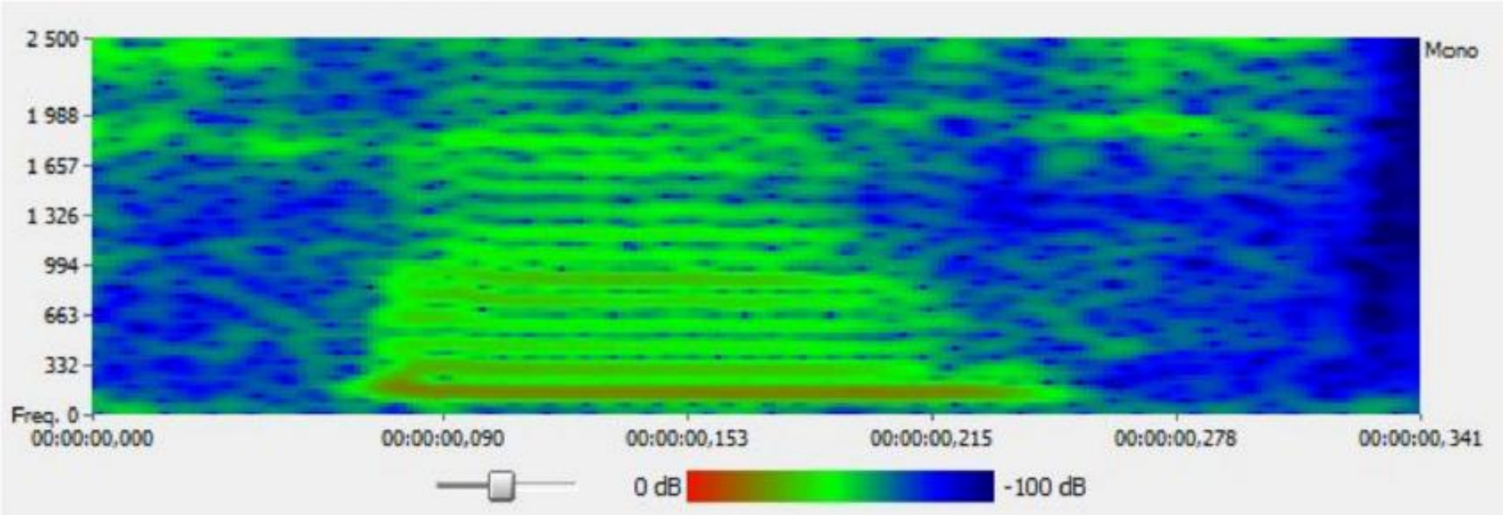
a) **Analyze the effect** of inputting the Hungarian word *"Sás"* (pronounced by an adult female speaker). Describe the characteristics of the output signal. Justify your answer with **numerical values**.

b) **Determine the optimal system parameters** (sampling frequency, input, and output filters) for **distortion-free transmission**.

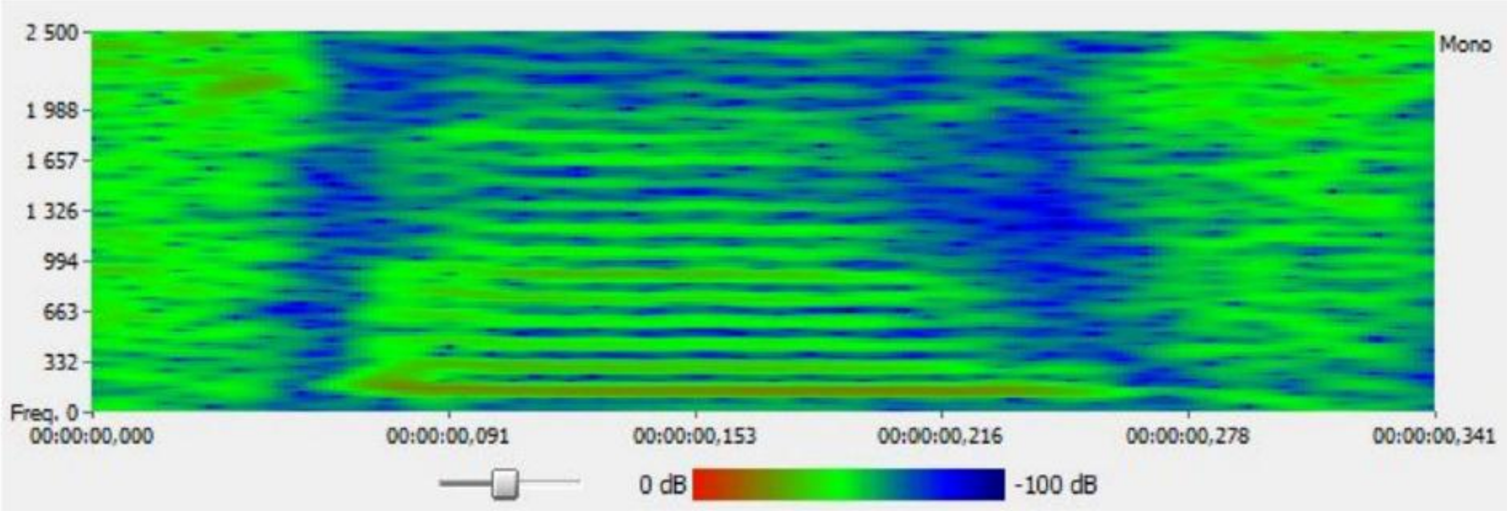
- “*Sás*”


8 000
6 362
5 302
4 241
3 181
2 121
1 061
Freq. 0
Mono
00:00:00,000
00:00:00,090
00:00:00,153
00:00:00,215
00:00:00,278
00:00:00,341
0 dB
-100 dB

a) Original Signal



b) Sampled Signal Without an Anti-Aliasing Filter

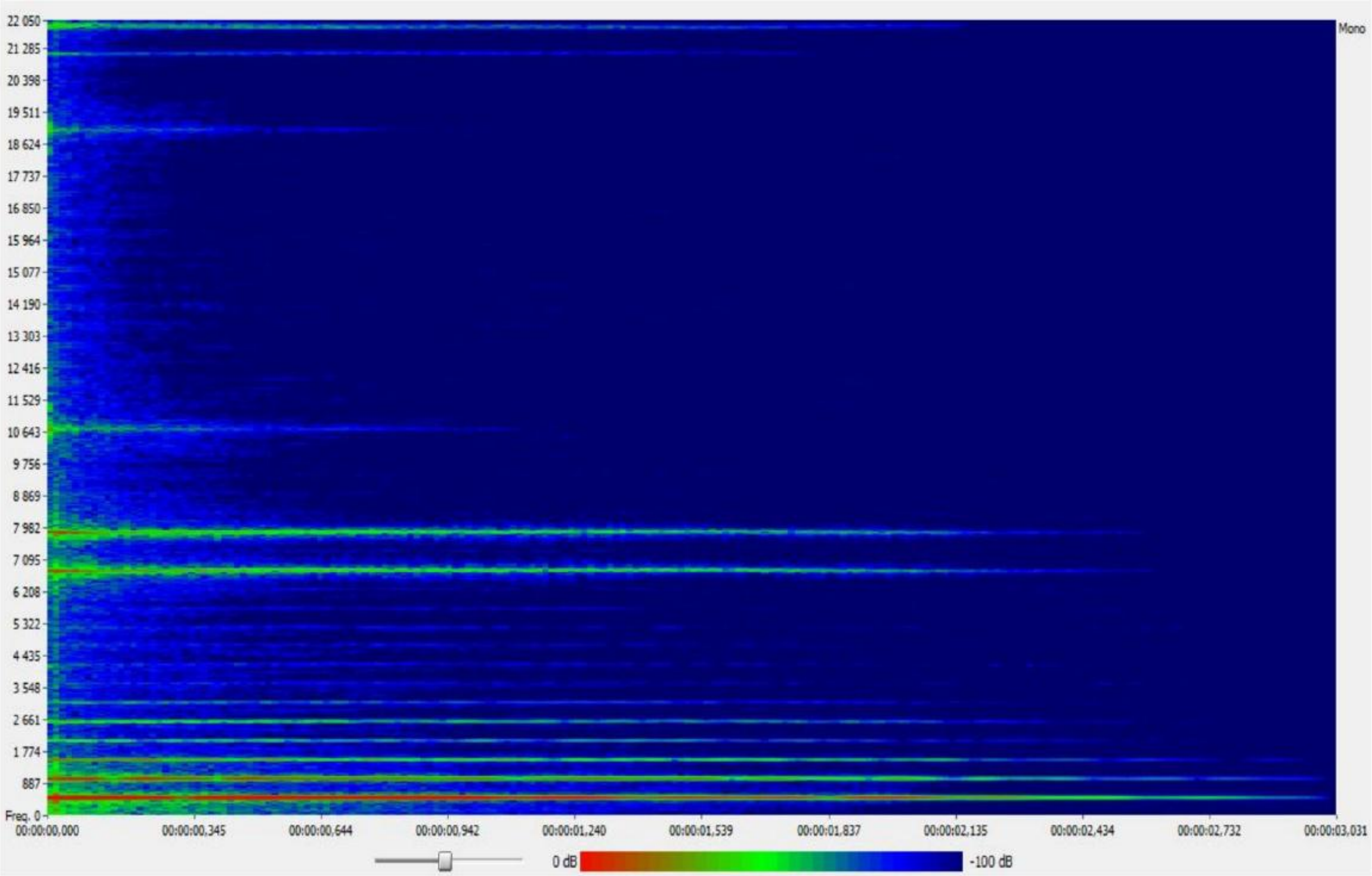


A **speech sampling system** operates with an **unknown sampling frequency** and **PAM-type reconstruction filter with H(f) characteristics**. The system alternately receives **speech signals and piano solos**. The **speech sounds intelligible**, but the **piano sound is distorted**.

Tasks:

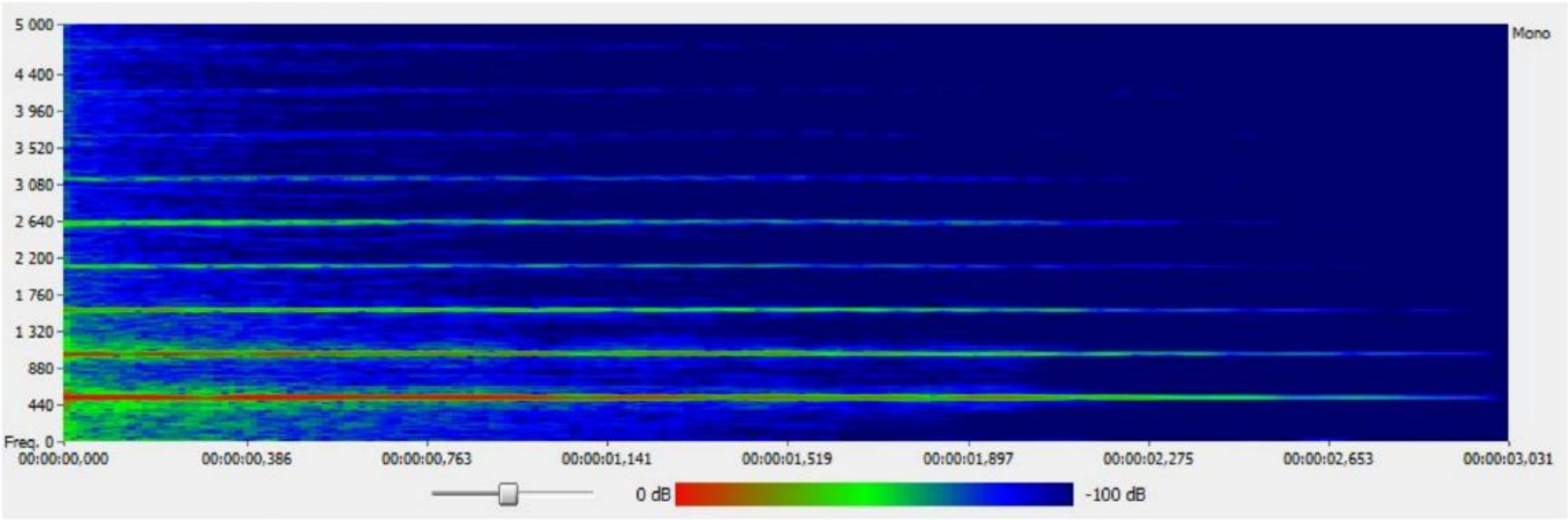
a) **Identify system parameters** that could cause this behavior. Multiple correct answers are possible.
b) **Propose a solution** that reconstructs the original signal with **reasonable complexity**.

Piano

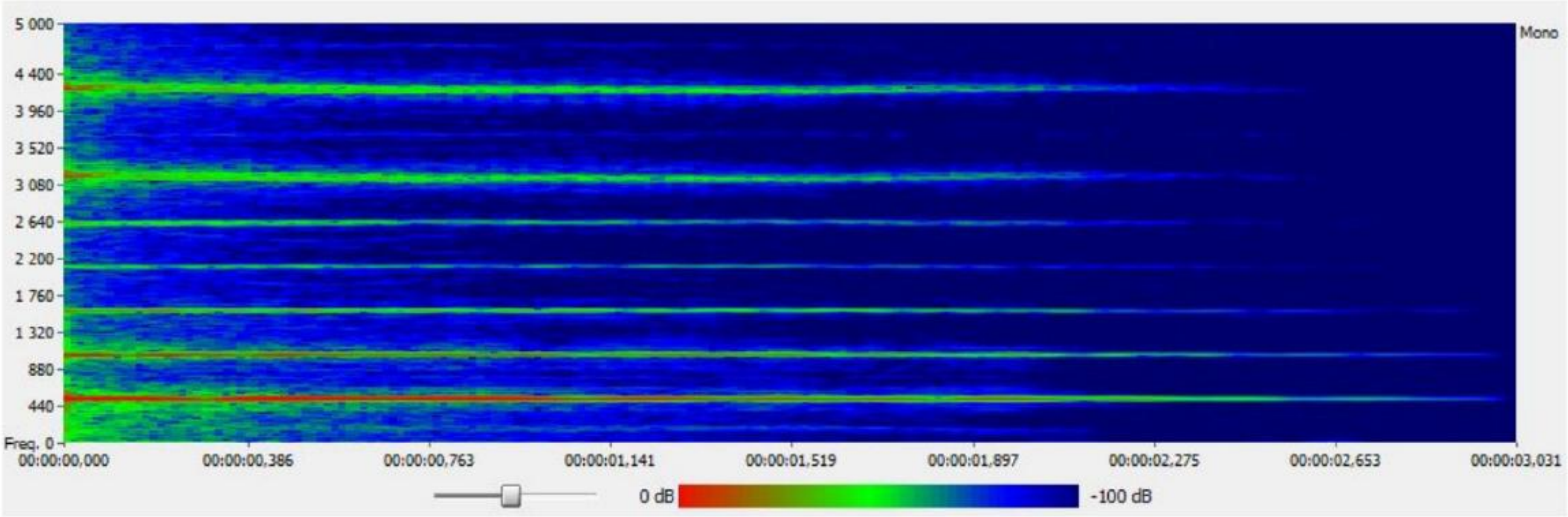


22 050
21 285
20 398
19 511
18 624
17 737
16 850
15 964
15 077
14 190
13 303
12 416
11 529
10 643
9 756
8 869
7 982
7 095
6 208
5 322
4 435
3 548
2 661
1 774
887
Freq. 0
Mono
00:00:00,000
00:00:00,345
00:00:00,644
00:00:00,942
00:00:01,240
00:00:01,539
00:00:01,837
00:00:02,135
00:00:02,434
00:00:02,732
00:00:03,031
0 dB
-100 dB

a) Original Piano Signal



b) Sampled Piano Signal Without an Anti-Aliasing Filter



- A text **(A4-sized, 50 lines, 80 characters per line)** is to be read aloud and **digitally stored**. Estimate the required **storage space** for the digitized audio using **44.1 kHz, 16-bit PCM format**, assuming a **female speaker**.

- **Outline the calculation steps** without a calculator. **Provide the final result** in **MB**.

Complete the following sentence with the correct statement:

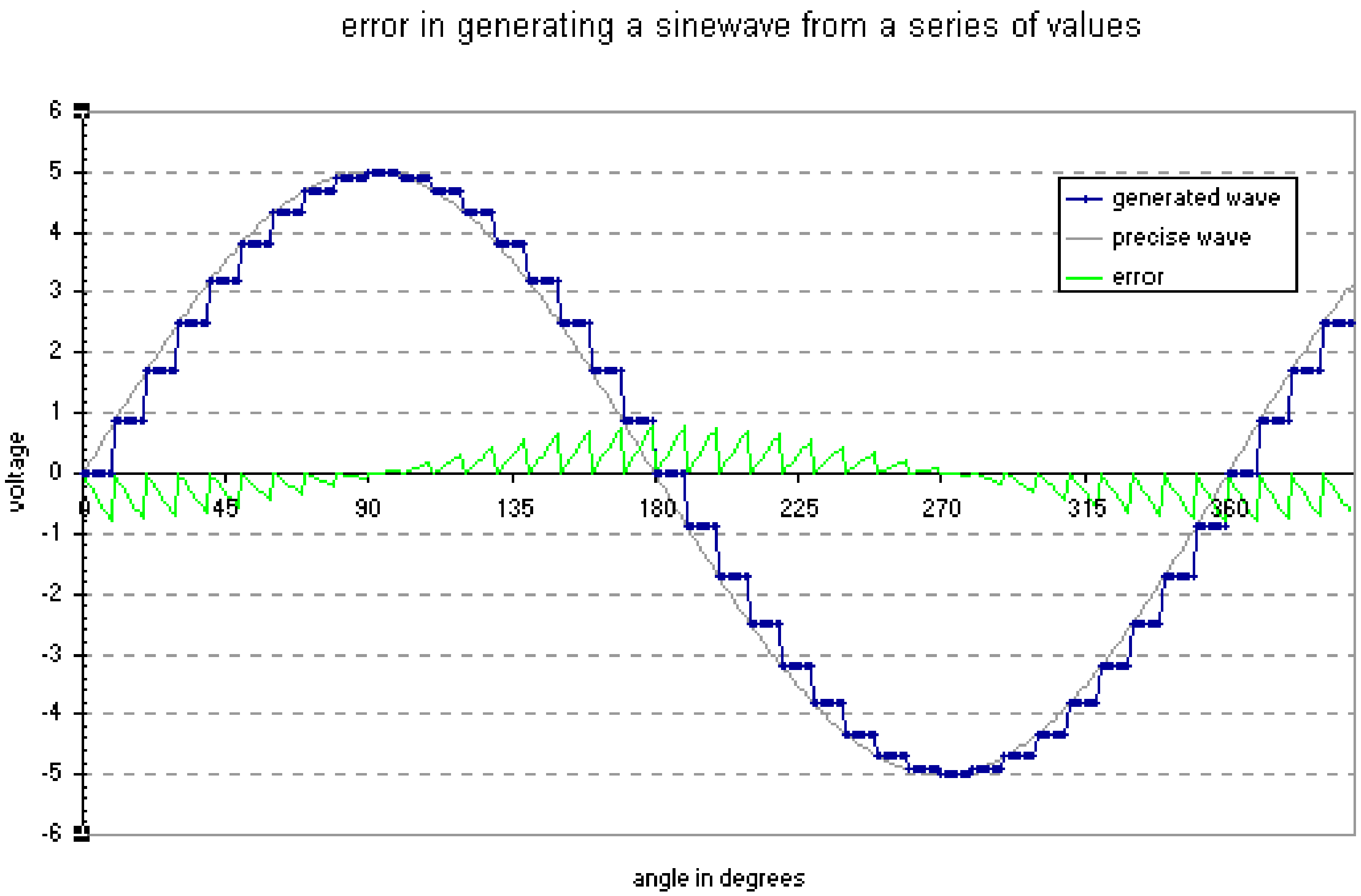
"When sampling speech at 10 kHz without an anti-aliasing filter and reconstructing it with a 4.5 kHz low-pass filter…"

- The audio quality will certainly be better than telephone quality.
- Aliasing is prevented.
- Speech components near 9 kHz are filtered out.
- The output signal will be completely unintelligible and unusable.
- The signal can always be perfectly reconstructed.
- The signal will be mostly intelligible but noisy.

QUANTIZATION

A **linear quantizer** has a **maximum signal-to-quantization noise ratio of 72 dB**.
How many **bits** does the quantizer use?

- **12 bits**
- **72 bits**
- **20 bits**
- **16 bits**
- **8 bits**
- **32 bits**



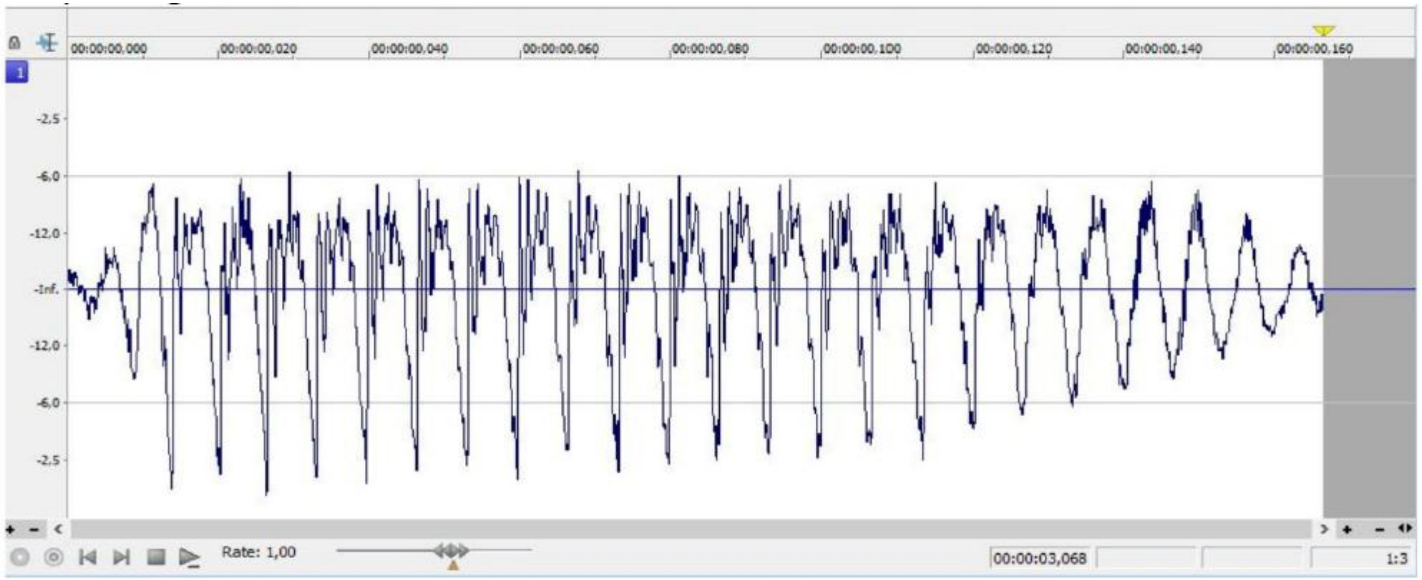
https://www.skillbank.co.uk/SignalConversion/quanterror.htm

Complete the following statement with the correct option:

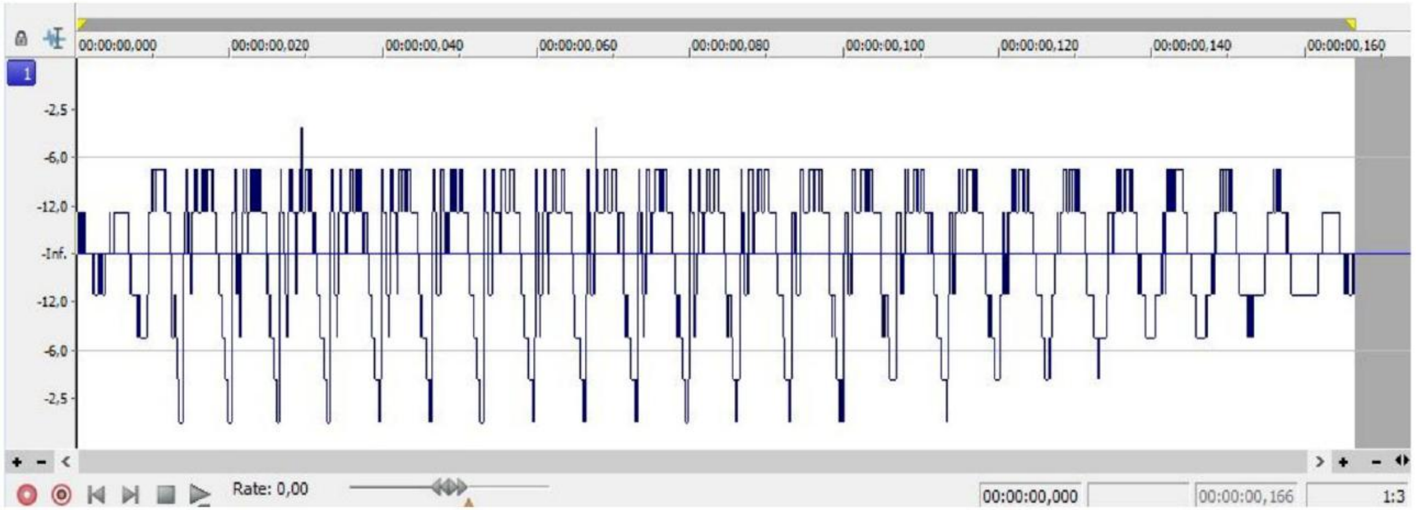
"Quantization step size…"

- Is independent of the sampling frequency.
- Depends on the sampling frequency.
- Can be perfectly reversed without loss.
- Does not affect audio quality.
- Is only used with uniform spacing for speech signals.
- Depends on the fundamental frequency of the speech.

a) Original Signal



b) Signal Quantized at a Low Bit Depth



SPEECH CODING

Rank the following **speech coding methods** by **perceived speech quality** and justify your ranking.

- 8 kHz, 8-bit A-law quantization
- 2 kHz, 32-bit linear quantization
- 13 kbps GSM full-rate coder

Is there a linear relationship between the bitrate of speech coders and speech quality?

A speech database of 15 hours of audio was recorded in the studio at a sampling rate of 44.1kHz in mono. Julie encoded the speech database with a 64 kbps MP3 encoder and then gave the material to John in MP3 format. John decoded the resulting audio and then re-encoded it with an encoder using the A-law PCM algorithm. The re-compressed audio took up about 412 MB on John's hard drive.

Task:

- Will the data that Julie gave to John fit in a 1 GB cloud storage?
- Is the audio compressed by John of sufficient quality for a telephone speech information system?
- Does John need to perform any signal processing steps before encoding?

- A **1-hour male speech sample** (8 kHz, 8-bit PCM) is **compressed using LPC-10**.
- The LPC-10 coder uses:
  - F0 values stored at 12 bits
  - Residual energy and LP coefficients stored at 10 bits each
  - Frame duration of 25 ms

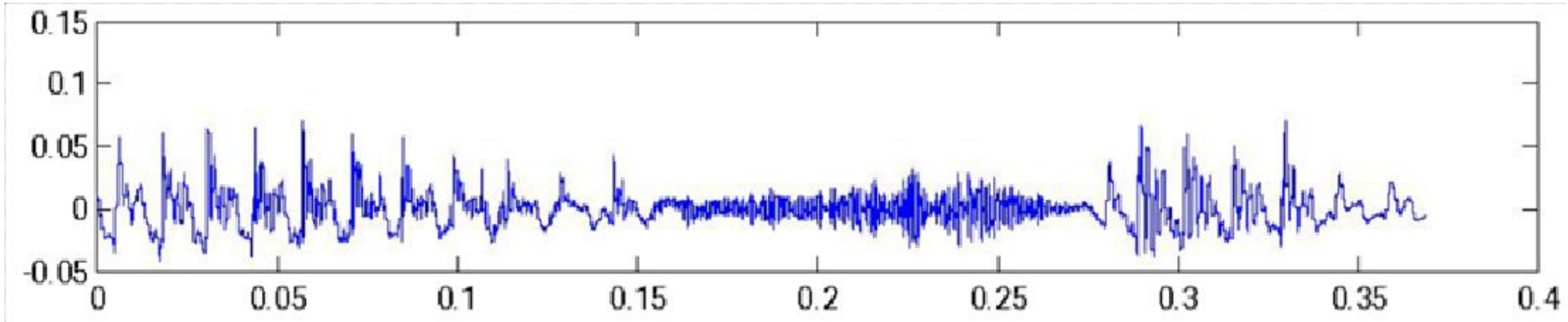
Tasks:

a) Calculate the compression ratio of the LPC-10 coder.

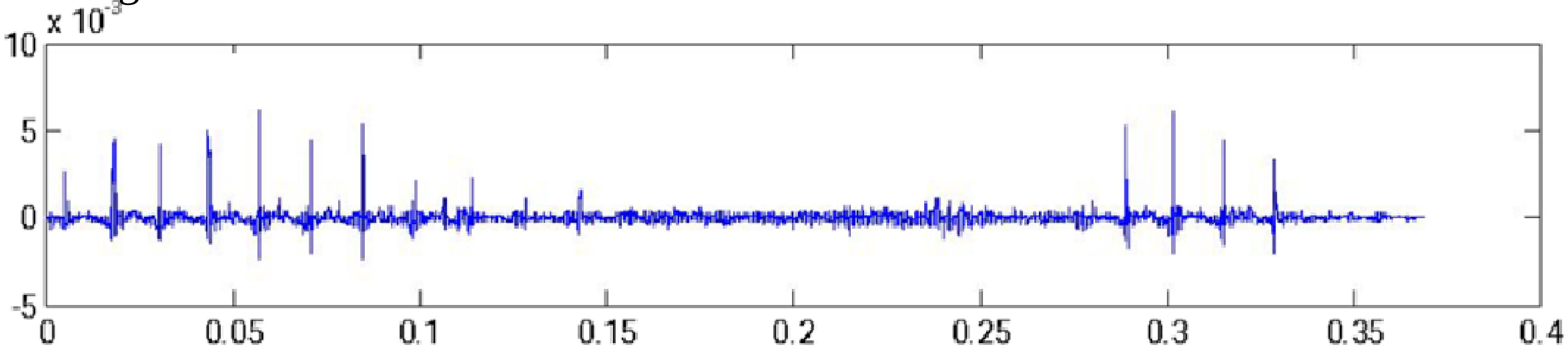
b) Describe how to synthesize monotone speech using LPC-10.

c) Explain the difference between whispered and normal speech and how to transform the LPC-10 speech to whispered speech.

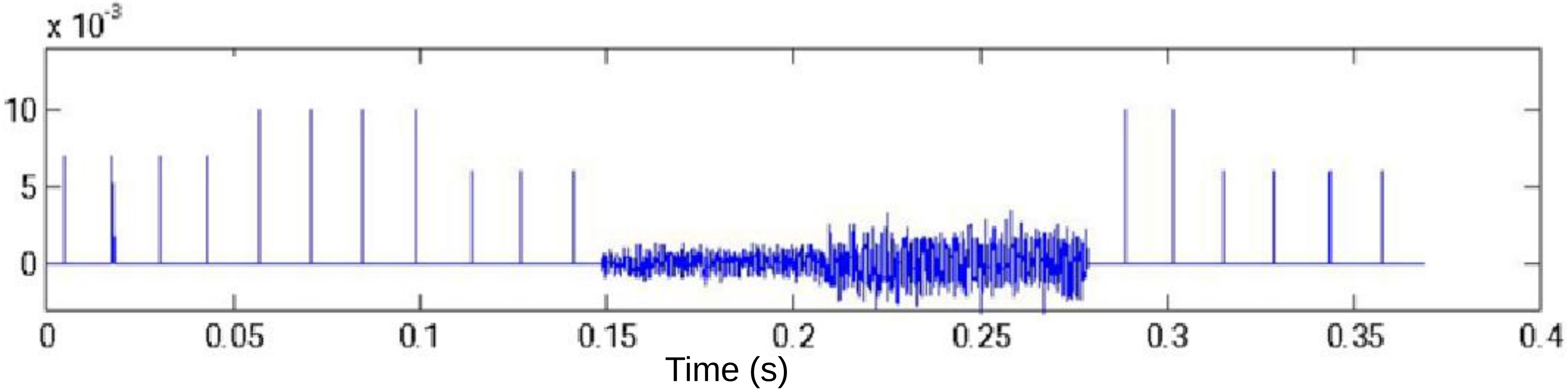
(a) Original Speech Signal



(b) residual signal



(c) LPC-10 excitation signal



a) Original Speech Signal

b) LPC-10 Compressed Speech

c) LPC-10 Monotone Speech

d) LPC-10 Whispered Speech

What is characteristic of the **residual signal** in **LPC analysis**?

- It contains signal components not captured within the analysis window.
- Its energy is greater than the original signal.
- It can be discarded without affecting reconstruction.
- It requires fewer bits for quantization than the original signal.
- It contains formant frequency information.
- LPC synthesis can be performed without the original residual signal.

BONUS TASK

- A **clean speech and music recording** is digitized at **24 kHz, 16-bit** with an **8 kHz low-pass filter** at the input.
- Playback uses:
  - 16-bit DAC (0-5V output range)
  - 24 kHz playback sampling rate
  - 8 kHz low-pass output filter

Tasks:

a) How much does the signal-to-noise ratio degrade when using an 8-bit linear DAC instead of a 16-bit DAC?
b) What would change if the input filter cutoff frequency were doubled?

Sampling, quantization, speech coding