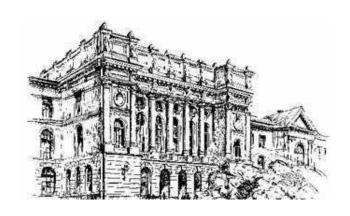
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Программирование

Отчет по курсовой работе Приложение "Sun Radio"

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1 Проектирование приложения

В современном мире музыка является неотъемлемой частью жизни многих и многих людей. Музыка часто сопровождает нас на работе, в дороге, во время досуга. Порой музыка может играть долгие часы. Настолько долгие, что за время прослушивания солнце может изменить свое положение относительно прослушивающего. Со временем суток так же меняется и настроение, и запас сил. И как было бы здорово, если бы музыка подстраивалась под наше самоощущение, дополняла и обогащала его.

1.1 Задание

Разработать приложение, позволяющее пользователям автоматически изменять тональность и громкость воспроизводимой музыки в соответствии с уровнем освещения.

1.2 Концепция

Изменения в музыкальные файлы вносятся с помощью преобразования Фурье. Информация об освещенности поступает с фоторезистра.

1.3 Минимально работоспособный продукт

Консольное приложение, получающее на вход музыкальный файл и воспроизводящее его в соответствии с текущим уровнем освещенности.

1.4 Решаемые задачи

В процессе проектирования приложения было выделено три основных задачи.

• Получение спектра файла

Для удобства работы используются .wav файлы. Удобство в том, что в каждом фрейме хранятся значения амплитуд, то есть имеем зависимость амплитуды от времени, что является входными данными для преобразования Фурье. С помощью этого преобразования получим спектр файла — зависимость амплитуды от частоты. Но не всё так просто. При последовательном чтении фреймов накапливаются ошибки, поэтому необходимо использовать оконную функцию и читать данные с перекрытием. Эмпирически получено наилучшее перекрытие — в одну шестнадцатую от количества читаемых фреймов при условии, что читаем по 2048 фреймов. Число фреймов, равное степени двойки, выбрано не случайно: это задел на будущее. Для быстрого преобразования Фурье необходимо число фреймов, равное степени двойки. Используемая оконная функция — функция Блэкмана-Наталла. Выбрана она за минимальный размер боковых лепестков и удовлетворительное "растяжение"спектра. При использовании входного фильтра необходим выходной фильтр. Выходной фильтр должен быть таким, чтобы сумма произведений входного и выходного фильтров в каждой точке была равна единице. Для этого здесь я просто нормализую входную оконную функцию.

• Корректные преобразования

Необходимо изменить тон и громкость воспроизведения. Для изменения тона необходимо растянуть файл в N раз, интерполируя значения амплитуды и фазы между точками, а затем ускорить воспроизведение в N раз. Таким образом получим изменение тона при сохранении скорости воспроизведения.

• Воспроизведение файла

Воспроизведение файла производится средствами javax.sound.sampled.*

1.5 Выводы

В данном разделе рассмотрен процесс проектирования приложения для модуляции звука в зависимости от уровня освещенности. Выделены основные задачи и предложены варианты их решения.

2 Реализация приложения

2.1 Среда разработки

Операционная система: Windows 8.1 Среда разработки: IntelliJ IDEA 2016.3.4 Компилятор: javac, JDK 1.8.0_102

2.2 Выделенные классы

В приложении были выделены следующие классы:

- SunRadio главный класс. Осуществляет основную работу приложения.
- Complex класс для работы с комплексными числами. Взят из открытого источника.
- WavFile класс для работы с .wav файлами. Взят из открытого источника.
- WavFileException исключения для класса WavFile. Взят из открытого источника.
- LightLevel класс, с помощью которого можно получить текущий уровень освещенности.
- АМ класс, содержащий средства для амплитудной модуляции.
- DFTStraight реализует прямое дискретное преобразование Фурье.
- DFTInverse реализует обратное дискретное преобразование Фурье.
- Filter реализует оконную функцию.
- Interpolation реализует линецную интерполяцию по двум точкам.
- Scale реализует масштабирование в заданных пределах.
- ToneModulation реализует модуляцию тона.
- ToneModulationException исключения для класса ToneModulation.

2.3 Выводы

В данном разделе были описаны все классы, выделенные в процессе работы над проектом.

3 Процесс обеспечения качества и тестирование

3.1 Тестирование

Для проверки работы библиотеки использовались автоматические тесты, покрывающие основную функциональность ядра. Также в процессе разработки приложения проводилось ручное тестирование программы.

3.2 Выводы

В данном разделе описан процесс тестирования программы.

4 Выводы

В результате работы над курсовым проектом было реализовано приложение, предназначенное для изменения музыки в соответствии с уровнем освещенности. Изучено: преобразование Фурье, оконные функции, амплитудная модуляция, линейная интерполяция. Так же приобретены навыки разработки приложения на языке Java, а также навыки разработки тестов на языке Groovy.

5 Приложение 1

Листинг 1: Complex.java

```
/*

* This work by W. Patrick Hooper is free of known copyright restrictions.

* The work is in the public domain.

* Author's website: <a href="http://wphooper.com">http://wphooper.com</a>.

*/

package com.external;
```

```
10 /**
11
    * This class stores a complex number, and allows the user to do arithmetic
12
   * with these numbers.
13
    \ast Note that our complex numbers are immutable. That is, once they are
14
    \ast constructed, they will not change. In particular, all our algebraic
15
    * operations create a new complex number rather than updating the current one.
16
17
18
   * @author W. Patrick Hooper
19
   public final class Complex {
20
21
       // The number stored is x+I*y.
       final private double x, y;
22
23
       // I don't want to allow anyone to access these numbers so I've labeled
24
       // them private.
25
26
       /** Construct a point from real and imaginary parts. */
27
       public Complex(double real_part, double imaginary_part) {
28
           x=real_part;
29
           y=imaginary_part;
30
31
       /** Construct a real number. */
32
33
       public Complex(double real part) {
34
           x=real part;
35
           y = 0;
36
37
38
       // A static constructor.
39
40
       /** Construct a complex number from the given polar coordinates. */
       public static Complex fromPolar(double r, double theta) {
41
42
           return new Complex (r*Math.cos(theta), r*Math.sin(theta));
43
44
45
       // Basic operations on Complex numbers.
46
47
       /** Return the real part. */
       public double re(){
48
49
           return x;
50
51
52
       /** Return the imaginary part. */
53
       public double im(){
54
           return y;
55
56
57
       /** Return the complex conjugate */
58
       public Complex conj() {
59
           return new Complex (x, -y);
60
61
62
       /** Return the square of the absolute value. */
63
       public double absSquared() {
64
           return x*x+y*y;
65
66
67
       /** Return the absolute value. */
68
       public double abs() {
69
              The java.lang.Math package contains many useful mathematical functions,
           // including the square root function.
70
71
           return Math.sqrt(absSquared());
72
73
74
       // ARITHMETIC
75
76
       /** Add a complex number to this one.
77
78
        * @param z The complex number to be added.
79
        * @return A new complex number which is the sum.
80
81
       public Complex add(Complex z) {
82
           return new Complex (x+z.x, y+z.y);
83
84
85
       /** Subtract a complex number from this one.
```

```
86
 87
         * @param z The complex number to be subtracted.
88
         * @return A new complex number which is the sum.
 89
90
        public Complex minus (Complex z) {
91
            return new Complex (x-z.x, y-z.y);
92
93
94
        /** Negate this complex number.
95
96
         * @return The negation.
97
98
        public Complex neg() {
99
            return new Complex(-x, -y);
100
101
102
        /** Compute the product of two complex numbers
103
         st @param z The complex number to be multiplied.
104
105
         * @return A new complex number which is the product.
106
107
        public Complex mult (Complex z) {
            \mathbf{return} new Complex (x*z.x-y*z.y, x*z.y+z.x*y);
108
109
110
        /** Divide this complex number by a real number.
111
112
         * @param q The number to divide by.
113
         * @return A new complex number representing the quotient.
114
115
116
        public Complex div(double q) {
            return new Complex (x/q, y/q);
117
118
119
        /** Return the multiplicative inverse. */
120
        public Complex inv() {
121
122
            // find the square of the absolute value of this complex number.
123
            double abs_squared=absSquared();
            return new Complex(x/abs_squared, -y/abs_squared);
124
125
126
127
        /** Compute the quotient of two complex numbers.
128
129
         * @param z The complex number to divide this one by.
130
         * @return A new complex number which is the quotient.
131
132
        public Complex div (Complex z) {
133
            return mult(z.inv());
134
135
        /** Return the complex exponential of this complex number. */
136
137
        public Complex exp() {
138
            return new Complex (Math.exp(x)*Math.cos(y), Math.exp(x)*Math.sin(y));
139
140
141
142
        // FUNCTIONS WHICH KEEP JAVA HAPPY:
143
144
        /** Returns this point as a string.
145
         * The main purpose of this function is for printing the string out,
         * so we return a string in a (fairly) human readable format.
146
147
        // The optional override directive "@Override" below just says we are
148
        // overriding a function defined in a parent class. In this case, the
149
150
        // parent is java.lang.Object. All classes in Java have the Object class
151
        // as a superclass.
        @Override
152
153
        public String toString() {
            // Comments:
// 1) "" represents the empty string.
154
155
156
            // 2) If you add something to a string, it converts the thing you
157
            // are adding to a string, and then concatentates it with the string.
158
159
            // We do some voodoo to make sure the number is displayed reasonably.
            if (y==0) {
160
                return ""+x;
161
```

```
162
163
             if (y>0) {
                return ""+x+"+"+y+"*I";
164
165
            // otherwise y < 0.
return ""+x+"-"+(-y)+"*I";
166
167
168
169
170
        /** Return true if the object is a complex number which is equal to this complex number.
        @Override
171
172
        public boolean equals(Object obj) {
               Return false if the object is null
173
174
            if (obj == null) {
175
                 return false;
176
             // Return false if the object is not a Complex number
177
178
            if (!(obj instanceof Complex)) {
                return false;
179
180
181
            // Now the object must be a Complex number, so we can convert it to a
182
             // Complex number.
183
184
            Complex other = (Complex) obj;
185
186
             // If the x-coordinates are not equal, then return false.
187
            if (x != other.x) {
188
                 return false;
189
             // If the y-coordinates are not equal, then return false.
190
191
            if (y != other.y) {
                return false;
192
193
194
            // Both parts are equal, so return true.
195
            return true;
196
197
        // Remark: In Java, we should really override the hashcode function
198
199
        // whenever we override the equals function. But, I don't want to
200
        // get into this for a light introduction to programming in java.
201
        // Hash codes are necessary for various of Java's collections. See HashSet for instance.
202
        // The following was generated by Netbeans.
203
        @Override
204
        public int hashCode() {
            int hash = 3;
205
            hash = 83 * hash + (int) (Double.doubleToLongBits(this.x) ^ (Double.doubleToLongBits(
206
        \hookrightarrow this.x) >>> 32);
207
            hash = 83 * hash + (int) (Double.doubleToLongBits(this.y) ^ (Double.doubleToLongBits(
        \hookrightarrow this.y) >>> 32);
208
            return hash;
209
210
```

Листинг 2: WavFile.java

```
package com.external;
3
  import java.jo.*:
  import javax.sound.sampled.AudioFormat;
 6
  import javax.sound.sampled.AudioSystem;
7
8
9
10
   * Wav file IO class
   * http://www.labbookpages.co.uk
11
12
   * File format is based on the information from
13
   * http://www.sonicspot.com/guide/wavefiles.html
14
   * http://www.blitter.com/~russtopia/MIDI/~jglatt/tech/wave.htm
15
16
17
   * @author A. Greensted
18
   * @version 1.0
19
   */
20 public class WavFile
21 | {
```

```
22
     private enum IOState {READING, WRITING, CLOSED};
^{23}
     private final static int BUFFER SIZE = 4096;
24
25
     private final static int FMT CHUNK ID = 0x20746D66;
     26
27
28
29
30
     private File file;
                                       // File that will be read from or written to
     private IOState ioState;
31
                                         // Specifies the IO State of the Wav File (used for snaity
       // Number of bytes required to store a single sample // Number of frames within the data section
32
     private int bytesPerSample;
33
     private long numFrames;
     private FileOutputStream oStream; // Output stream used for writting data
34
                                            // Input stream used for reading data
// Scaling factor used for int <-> float conversion
35
     private FileInputStream iStream;
     private double floatScale;
36
     private double floatOffset; // Offset factor used for int <-> float conversion private boolean wordAlignAdjust; // Specify if an extra byte at the end of the data chunk
37
38
       \hookrightarrow is required for word alignment
39
     // Wav Header
40
                                          // 2 bytes unsigned, 0x0001 (1) to 0xFFFF (65,535) // 4 bytes unsigned, 0x00000001 (1) to 0xFFFFFFFF
     private int numChannels;
41
     private long sampleRate;
42
       \hookrightarrow (4,294,967,295)
43
                                // Although a java int is 4 bytes, it is signed, so need to use a
       \hookrightarrow long
                                          // 2 bytes unsigned, 0\,x\,0\,0\,0\,1 (1) to 0\,x\,\mathrm{FFFF} (6\,5\,,5\,3\,5)
44
     private int blockAlign;
                                         // 2 bytes unsigned, 0x0002 (2) to 0xFFFF (65,535)
     private int validBits;
45
46
47
     // Buffering
                                         // Local buffer used for IO
48
     private byte[] buffer;
     private int bufferPointer;
                                            // Points to the current position in local buffer
49
                                          // Bytes read after last read into local buffer
50
     private int bytesRead;
51
                                           // Current number of frames read or written
     private long frameCounter;
52
     // Cannot instantiate WavFile directly, must either use newWavFile() or openWavFile()
53
     private WavFile()
54
55
       buffer = new byte[BUFFER SIZE];
56
57
58
59
     public int getNumChannels()
60
61
       return numChannels;
62
63
64
     public long getNumFrames()
65
66
       return numFrames;
67
68
69
     public long getFramesRemaining()
70
71
       return numFrames - frameCounter;
72
73
74
     public long getFrameCounter() {
75
       return frameCounter;
76
77
78
     public long getSampleRate()
79
80
       return sampleRate;
81
82
83
     public int getValidBits()
84
85
       return validBits;
86
     public int getblockAlign()
87
88
89
       return blockAlign;
90
91
92
     public AudioFormat getAudioFormat() throws Exception
93
```

```
return AudioSystem.getAudioInputStream(file).getFormat();
95
      }
96
      public static WavFile newWavFile (File file, int numChannels, long numFrames, int validBits,
97
        \hookrightarrow \ long \ sampleRate) \ throws \ IO\,Exception \,, \ WavFileException
98
        // Instantiate new Wavfile and initialise
99
100
        WavFile wavFile = new WavFile();
101
        wavFile.file = file;
102
        wavFile.numChannels = numChannels;
103
        wavFile.numFrames = numFrames;
104
        wavFile.sampleRate = sampleRate;
        wavFile.bytesPerSample = (validBits + 7) / 8;
105
106
        wavFile.blockAlign = wavFile.bytesPerSample * numChannels;
107
        wavFile.validBits = validBits;
108
109
         / Sanity check arguments
110
        if (numChannels < 1 || numChannels > 65535) throw new WavFileException("Illegal_number_of_
        111
        if (numFrames < 0) throw new WavFileException("Number_of_frames_must_be_positive");
112
        if (validBits < 2 || validBits > 65535) throw new WavFileException("Illegal_number_of_

→ valid_bits,_valid_range_2_to_65536");
        if (sampleRate < 0) throw new WavFileException("Sample_rate_must_be_positive");
113
114
115
        // Create output stream for writing data
116
        wavFile.oStream = new FileOutputStream(file);
117
118
        // Calculate the chunk sizes
        long dataChunkSize = wavFile.blockAlign * numFrames;
119
120
        long mainChunkSize = 4 + // Riff Type
                       8 + // Format ID and size
121
                       16 + // Format data
8 + // Data ID and size
122
123
124
                       dataChunkSize;
125
        // Chunks must be word aligned, so if odd number of audio data bytes
126
127
          adjust the main chunk size
128
        if (dataChunkSize % 2 == 1) {
129
          mainChunkSize += 1;
130
          wavFile.wordAlignAdjust = true;
13\,1
132
        else {
          wavFile.wordAlignAdjust = false;
133
134
135
136
        // Set the main chunk size
        putLE(RIFF CHUNK ID, wavFile.buffer, 0, 4);
137
        putLE (mainChunkSize, wavFile.buffer, 4, 4);
138
139
        putLE(RIFF_TYPE_ID, wavFile.buffer, 8, 4);
140
        // Write out the header
141
        wavFile.oStream.write(wavFile.buffer, 0, 12);
142
143
144
        // Put format data in buffer
        long averageBytesPerSecond = sampleRate * wavFile.blockAlign;
145
146
147
        putLE (FMT CHUNK ID,
                                    wavFile.buffer, 0, 4);
                                                                 // Chunk ID
                                                           // Chunk Data Size
148
        putLE(16,
                                wavFile.buffer, 4, 4);
                                  wavFile.buffer \ , \quad 8 \ , \quad 2) \ ;
149
        putLE(1,
                                                             // Compression Code (Uncompressed)
                                                               // Number of channels
                                    wavFile.buffer, 10, 2);
wavFile.buffer, 12, 4);
150
        putLE (numChannels,
                                                                // Sample Rate
        putLE (sampleRate,
151
        putLE(averageBytesPerSecond, wavFile.buffer, 16, 4); // Average Bytes Per Second putLE(wavFile.blockAlign, wavFile.buffer, 20, 2); // Block Align
152
153
                                                               // Valid Bits
154
        putLE (validBits,
                                    wavFile.buffer, 22, 2);
155
156
        // Write Format Chunk
157
        wavFile.oStream.write(wavFile.buffer, 0, 24);
158
159
        // Start Data Chunk
                                                                   // Chunk ID
        putLE (DATA_CHUNK ID,
160
                                      wavFile.buffer, 0, 4);
161
        putLE (dataChunkSize,
                                      wavFile.buffer, 4, 4);
                                                                   // Chunk Data Size
162
163
        // Write Format Chunk
164
        wavFile.oStream.write(wavFile.buffer, 0, 8);
165
166
        // Calculate the scaling factor for converting to a normalised double
```

```
167
        if (wavFile.validBits > 8)
168
          // If more than 8 validBits, data is signed
169
170
          // Conversion required multiplying by magnitude of max positive value
171
          wavFile.floatOffset = 0;
172
          wavFile.floatScale = Long.MAX VALUE >> (64 - wavFile.validBits);
173
174
        else
175
          // Else if 8 or less validBits, data is unsigned
176
177
          // Conversion required dividing by max positive value
178
          wavFile.floatOffset = 1;
179
          wavFile.floatScale = 0.5 * ((1 << wavFile.validBits) - 1);
180
181
        // Finally, set the IO State
182
183
        wavFile.bufferPointer = 0;
184
        wavFile.bytesRead = 0;
        wavFile.frameCounter = 0;
185
        wavFile.ioState = IOState.WRITING;
186
187
188
        return wavFile;
189
      }
190
      public static WavFile openWavFile (File file) throws IOException, WavFileException
191
192
193
        // Instantiate new Wavfile and store the file reference
        WavFile wavFile = new WavFile();
194
        wavFile.file = file:
195
196
197
        // Create a new file input stream for reading file data
198
        wavFile.iStream = new FileInputStream(file);
199
200
        // Read the first 12 bytes of the file
        int bytesRead = wavFile.iStream.read(wavFile.buffer, 0, 12);
201
        if (bytesRead != 12) throw new WavFileException("Not_enough_wav_file_bytes_for_header");
202
203
        // Extract parts from the header
204
        long riffChunkID = getLE(wavFile.buffer, 0, 4);
205
206
        long chunkSize = getLE(wavFile.buffer, 4, 4);
207
        long riffTypeID = getLE(wavFile.buffer, 8, 4);
208
209
        // Check the header bytes contains the correct signature
210
        if (riffChunkID != RIFF CHUNK ID) throw new WavFileException("Invalid_Wav_Header_data,_

→ incorrect_riff_chunk_ID");

        if (riffTypeID != RIFF TYPE ID) throw new WavFileException("Invalid_Wav_Header_data,_
211

    incorrect_riff_type_ID");

212
213
        // Check that the file size matches the number of bytes listed {f in} header
214
        if (file.length() != chunkSize+8) {
         throw new WavFileException("Header_chunk_size_(" + chunkSize + ")_does_not_match_file_
215
        \hookrightarrow size (" + file.length() + ")");
216
217
        boolean foundFormat = false;
218
219
        boolean foundData = false;
220
        // Search for the Format and Data Chunks
221
222
        while (true)
223
224
          // Read the first 8 bytes of the chunk (ID and chunk size)
          bytesRead = wavFile.iStream.read(wavFile.buffer, 0, 8);
225
226
          if (bytesRead == -1) throw new WavFileException("Reached_end_of_file_without_finding_

    format_chunk");
227
          if (bytesRead != 8) throw new WavFileException("Could_not_read_chunk_header");
228
229
          // Extract the chunk ID and Size
          long chunkID = getLE(wavFile.buffer, 0, 4);
230
231
          chunkSize = getLE(wavFile.buffer, 4, 4);
232
233
          // Word align the chunk size
          // chunkSize specifies the number of bytes holding data. However,
234
235
            the data should be word aligned (2 bytes) so we need to calculate
          // the actual number of bytes in the chunk
236
237
          long \ numChunkBytes = (chunkSize\%2 == 1) \ ? \ chunkSize+1 \ : \ chunkSize;
238
```

```
239
          if (chunkID == FMT CHUNK ID)
240
            // Flag that the format chunk has been found
241
242
            foundFormat = true:
243
244
            // Read in the header info
245
            bytesRead = wavFile.iStream.read(wavFile.buffer, 0, 16);
246
247
            // Check this is uncompressed data
248
            int compressionCode = (int) getLE(wavFile.buffer, 0, 2);
            if (compressionCode != 1) throw new WavFileException("Compression_Code_" +
249

→ compressionCode + "unotusupported");
250
251
            // Extract the format information
252
            wavFile.numChannels = (int) getLE(wavFile.buffer, 2, 2);
            wavFile.sampleRate = getLE(wavFile.buffer, 4, 4);
253
            wavFile.blockAlign = (int) getLE(wavFile.buffer, 12, 2);
254
255
            wavFile.validBits = (int) getLE(wavFile.buffer, 14, 2);
256
257
            if (wavFile.numChannels == 0) throw new WavFileException("Number_of_channels_specified

→ Jin Jheader Jis Jequal Jto Jzero");
258
            if (wavFile.blockAlign == 0) throw new WavFileException("Block_Align_specified_in_
        → header_is_equal_to_zero");
            if (wavFile.validBits < 2) throw new WavFileException("Valid_Bits_specified_in_header_
259
        \hookrightarrow is less than 2");
260
            if (wavFile.validBits > 64) throw new WavFileException("Valid_Bits_specified_in_header

→ Jis greater than 64, this is greater than a long can hold");

261
            // Calculate the number of bytes required to hold 1 sample
262
263
            wavFile.bytesPerSample = (wavFile.validBits + 7) / 8;
264
            if (wavFile.bytesPerSample * wavFile.numChannels != wavFile.blockAlign)
              throw new WavFileException ("Block_Align_does_not_agree_with_bytes_required_for_
265

→ validBits_and_number_of_channels");
266
267
            // Account for number of format bytes and then skip over
            // any extra format bytes
268
269
            numChunkBytes -= 16:
270
            if (numChunkBytes > 0) wavFile.iStream.skip(numChunkBytes);
271
272
          else if (chunkID == DATA CHUNK ID)
273
274
            // Check if we've found the format chunk,
            // If not, throw an exception as we need the format information
275
276
              before we can read the data chunk
            if (foundFormat == false) throw new WavFileException("Data_chunk_found_before_Format_
277
       \hookrightarrow chunk");
278
279
            // Check that the chunkSize (wav data length) is a multiple of the
280
            // block align (bytes per frame)
281
            if (chunkSize % wavFile.blockAlign != 0) throw new WavFileException("Data_Chunk_size_

→ is _ not _ multiple _ of _ Block _ Align ");
282
283
            // Calculate the number of frames
284
            wavFile.numFrames = chunkSize / wavFile.blockAlign;
285
286
            // Flag that we've found the wave data chunk
287
            foundData = true;
288
289
            break:
290
291
          else
292
293
            // If an unknown chunk ID is found, just skip over the chunk data
294
            wavFile.iStream.skip(numChunkBytes);
295
296
297
298
         / Throw an exception if no data chunk has been found
299
        if (foundData == false) throw new WavFileException("Did_not_find_a_data_chunk");
300
301
        // Calculate the scaling factor for converting to a normalised double
302
        if (wavFile.validBits > 8)
303
304
            If more than 8 validBits, data is signed
305
          // Conversion required dividing by magnitude of max negative value
306
          wavFile.floatOffset = 0;
```

```
307
          wavFile.floatScale = 1 \ll (wavFile.validBits - 1);
308
309
        else
310
            Else if 8 or less validBits, data is unsigned
311
312
          // Conversion required dividing by max positive value
          wavFile.floatOffset = -1;
313
          wavFile.floatScale = 0.5 * ((1 << wavFile.validBits) - 1);
314
315
316
        wavFile.bufferPointer = 0;
317
318
        wavFile.bytesRead = 0;
        wavFile.frameCounter = 0;
319
320
        wavFile.ioState = IOState.READING;
321
322
        return wavFile;
323
      }
324
      // Get and Put little endian data from local buffer
325
326
      private \ static \ long \ getLE (\ byte [] \ buffer \ , \ int \ pos \ , \ int \ numBytes)
327
328
329
        numBytes --;
330
        pos += numBytes;
331
332
        long val = buffer[pos] & 0xFF;
        333
334
335
        return val;
      }
336
337
      private static void putLE(long val, byte[] buffer, int pos, int numBytes)
338
339
340
        for (int b=0; b<numBytes; b++)
341
          buffer[pos] = (byte) (val & 0xFF);
342
343
          val >>= 8;
344
          pos ++;
345
346
      }
347
348
      // Sample Writing and Reading
349
350
      private void writeSample(long val) throws IOException
351
352
        for (int b=0; b<br/>bytesPerSample; b++)
353
          if (bufferPointer == BUFFER SIZE)
354
355
356
            oStream.write(buffer, 0, BUFFER SIZE);
357
            bufferPointer = 0;
358
359
          buffer[bufferPointer] = (byte) (val & 0xFF);
360
361
          val >> \equiv 8;
362
          bufferPointer ++;
363
364
      }
365
366
      private long readSample() throws IOException, WavFileException
367
368
        long val = 0;
369
370
        for (int b=0; b<bytesPerSample; b++)
371
          if (bufferPointer == bytesRead)
372
373
374
            int read = iStream.read(buffer, 0, BUFFER SIZE);
375
            if (read == -1) throw new WavFileException("Not_enough_data_available");
            bytesRead = read;
376
377
            bufferPointer = 0;
378
379
          int v = buffer[bufferPointer];
380
          if (b < bytesPerSample-1 || bytesPerSample == 1) v &= 0xFF;
381
382
          val += v << (b * 8);
```

```
383
384
          bufferPointer ++;
385
386
387
        return val;
388
389
390
         Integer
391
392
      public int readFrames(int[] sampleBuffer, int numFramesToRead) throws IOException,
        → WavFileException
393
        \textbf{return} \hspace{0.2cm} \texttt{readFrames(sampleBuffer, 0, numFramesToRead);} \\
394
395
396
      public int readFrames(int[] sampleBuffer, int offset, int numFramesToRead) throws
397
        → IOException, WavFileException
398
        if (ioState != IOState.READING) throw new IOException("Cannot_read_from_WavFile_instance")
399
400
401
        for (int f=0 ; f<numFramesToRead ; f++)
402
          if (frameCounter == numFrames) return f;
403
404
405
          \quad \textbf{for} \quad ( \text{ int } c = 0 \ ; \ c < num Channels \ ; \ c + +)
406
407
            sampleBuffer[offset] = (int) readSample();
408
             offset ++;
409
410
          frameCounter ++;
411
412
413
414
        return numFramesToRead;
415
      }
416
      public int readFrames(int[][] sampleBuffer, int numFramesToRead) throws IOException,
417
         → WavFileException
418
419
        return readFrames(sampleBuffer, 0, numFramesToRead);
420
      }
421
422
      public int readFrames(int[][] sampleBuffer, int offset, int numFramesToRead) throws
        \hookrightarrow IOException, WavFileException
423
        if (ioState != IOState.READING) throw new IOException("Cannot_read_from_WavFile_instance")
424
425
426
        for (int f=0; f<\text{numFramesToRead}; f++)
427
          if (frameCounter == numFrames) return f;
428
429
          for (int c=0; c<numChannels; c++) sampleBuffer[c][offset] = (int) readSample();
430
43\,1
432
          offset ++:
43\,3
          frameCounter ++;
434
435
436
        return numFramesToRead;
437
438
439
      public int writeFrames(int[] sampleBuffer, int numFramesToWrite) throws IOException,
        → WavFileException
440
441
        return writeFrames(sampleBuffer, 0, numFramesToWrite);
442
443
      public int writeFrames(int[] sampleBuffer, int offset, int numFramesToWrite) throws
444
        → IOException, WavFileException
445
        if (ioState != IOState.WRITING) throw new IOException("Cannot_write_to_WavFile_instance");
446
447
448
        for (int f=0; f<numFramesToWrite; f++)
449
450
          if (frameCounter == numFrames) return f;
```

```
451
452
           for (int c=0; c<numChannels; c++)
453
454
             writeSample(sampleBuffer[offset]);
455
             offset ++;
456
457
458
           frameCounter ++;
459
460
461
         return numFramesToWrite;
462
      }
463
46\,4
      public int writeFrames(int[][] sampleBuffer, int numFramesToWrite) throws IOException,
         → WavFileException
465
466
         return writeFrames (sampleBuffer, 0, numFramesToWrite);
467
      }
468
469
      public int writeFrames(int[][] sampleBuffer, int offset, int numFramesToWrite) throws
        \hookrightarrow \ \ IOException \ , \ \ WavFileException
470
         if (ioState != IOState.WRITING) throw new IOException("Cannot_write_to_WavFile_instance");
471
472
473
         for (int f=0; f<numFramesToWrite; f++)
474
475
           if (frameCounter == numFrames) return f;
476
           for (int c=0; c<numChannels; c++) writeSample(sampleBuffer[c][offset]);
477
478
479
           offset ++;
           frameCounter ++;
480
48\,1
482
483
         return numFramesToWrite;
484
      }
485
486
         Long
487
      public\ int\ readFrames(long[]\ sampleBuffer\ ,\ int\ numFramesToRead)\ throws\ IOException\ ,
488
         → WavFileException
489
         return readFrames(sampleBuffer, 0, numFramesToRead);
490
491
      }
492
493
      public int readFrames(long[] sampleBuffer, int offset, int numFramesToRead) throws
         → IOException, WavFileException
494
         if (ioState != IOState.READING) throw new IOException("Cannot_read_from_WavFile_instance")
495
496
497
         for (int f=0; f<numFramesToRead; f++)
498
           if (frameCounter == numFrames) return f;
499
500
           \label{eq:formula} \textbf{for} \hspace{0.2cm} (\hspace{0.1cm} \texttt{int} \hspace{0.2cm} c{=}0 \hspace{0.2cm} ; \hspace{0.2cm} c{<}\texttt{numChannels} \hspace{0.2cm} ; \hspace{0.2cm} c{+}{+})
501
502
503
             sampleBuffer[offset] = readSample();
504
             offset ++;
505
506
507
           frameCounter ++;
508
509
510
         return numFramesToRead;
511
512
      public int readFrames(long[][] sampleBuffer, int numFramesToRead) throws IOException,
513

→ WavFileException

514
515
         return readFrames(sampleBuffer, 0, numFramesToRead);
516
517
      public int readFrames(long[][] sampleBuffer, int offset, int numFramesToRead) throws
518
        \hookrightarrow \ \ IOException \ , \ \ WavFileException
519
```

```
520
        if (ioState != IOState.READING) throw new IOException("Cannot_read_from_WavFile_instance")
521
        for (int f=0; f<numFramesToRead; f++)
522
523
524
          if (frameCounter == numFrames) return f;
525
          for (int c=0; c<numChannels; c++) sampleBuffer[c][offset] = readSample();
526
527
528
          offset ++;
          frameCounter ++;
529
530
531
532
        return numFramesToRead;
533
534
      public int writeFrames(long[] sampleBuffer, int numFramesToWrite) throws IOException,
535
       → WavFileException
536
537
        return writeFrames(sampleBuffer, 0, numFramesToWrite);
538
539
      public int writeFrames(long[] sampleBuffer, int offset, int numFramesToWrite) throws
540
       \hookrightarrow IOException, WavFileException
541
        if (ioState != IOState.WRITING) throw new IOException("Cannot_write_to_WavFile_instance");
542
543
        for (int f=0; f<numFramesToWrite; f++)
544
545
          if (frameCounter == numFrames) return f;
546
547
          for (int c=0; c<numChannels; c++)
548
549
550
            writeSample(sampleBuffer[offset]);
551
            offset ++;
552
553
554
          frameCounter ++;
555
556
557
        return numFramesToWrite;
      }
558
559
560
      public int writeFrames(long[][] sampleBuffer, int numFramesToWrite) throws IOException,
       → WavFileException
561
        return writeFrames(sampleBuffer, 0, numFramesToWrite);
562
563
564
565
      public int writeFrames(long[][] sampleBuffer, int offset, int numFramesToWrite) throws
        → IOException, WavFileException
566
        if (ioState != IOState.WRITING) throw new IOException("Cannot_write_to_WavFile_instance");
567
568
        for (int f=0 ; f<numFramesToWrite ; f++)</pre>
569
570
571
          if (frameCounter == numFrames) return f;
572
          for (int c=0; c<numChannels; c++) writeSample(sampleBuffer[c][offset]);
573
574
575
          offset ++:
576
          frameCounter ++;
577
578
579
        return numFramesToWrite;
580
581
582
      // Double
583
      public int readFrames(double [] sampleBuffer, int numFramesToRead) throws IOException,
584
        → WavFileException
585
586
        return readFrames(sampleBuffer, 0, numFramesToRead);
587
      }
588
589
      public int readFrames(double [] sampleBuffer, int offset, int numFramesToRead) throws
```

```
→ IOException, WavFileException
590
        if (ioState != IOState.READING) throw new IOException("Cannot_read_from_WavFile_instance")
591
592
593
        for (int f=0; f<\text{numFramesToRead}; f++)
594
595
          if (frameCounter == numFrames) return f;
596
597
          for (int c=0; c<numChannels; c++)
598
599
            sampleBuffer[offset] = floatOffset + (double) readSample() / floatScale;
600
            offset ++;
601
602
603
          frameCounter ++;
604
605
606
        \textbf{return} \hspace{0.1in} \texttt{numFramesToRead} \hspace{0.1in} ; \\
607
      }
608
      public int readFrames(double [][] sampleBuffer, int numFramesToRead) throws IOException,
609

→ WavFileException

610
611
        return readFrames(sampleBuffer, 0, numFramesToRead);
612
613
      public int readFrames(double[][] sampleBuffer, int offset, int numFramesToRead) throws
614
        \hookrightarrow \ \ IOException \ , \ \ WavFileException
615
616
        if (ioState != IOState.READING) throw new IOException("Cannot_read_from_WavFile_instance")
617
618
        for (int f=0; f<numFramesToRead; f++)
619
620
          if (frameCounter == numFrames) return f;
621
          for (int c=0; c<numChannels; c++) sampleBuffer[c][offset] = floatOffset + (double)
622

    readSample() / floatScale;
623
624
          offset ++;
625
          frameCounter ++;
626
627
628
        return numFramesToRead;
629
630
      public int readFramesWithOverlap(double[] sampleBuffer, int numFramesToRead, int overlap)
631

→ throws IOException , WavFileException

632
        numFramesToRead = readFrames(sampleBuffer, 0, numFramesToRead);
633
        long coeff = frameCounter * overlap / numFramesToRead - overlap + 1;
634
        frameCounter = coeff * numFramesToRead / overlap;
635
636
637
        return numFramesToRead;
638
      }
639
640
      public int writeFrames(double[] sampleBuffer, int numFramesToWrite) throws IOException,
641
        → WavFileException
642
        return writeFrames(sampleBuffer, 0, numFramesToWrite);
643
644
      }
645
646
      public int writeFrames(double[] sampleBuffer, int offset, int numFramesToWrite) throws
        → IOException, WavFileException
647
648
        if (ioState != IOState.WRITING) throw new IOException("Cannot_write_to_WavFile_instance");
649
        for (int f=0; f<numFramesToWrite; f++)
650
651
          if (frameCounter == numFrames) return f;
652
653
654
          for (int c=0; c<numChannels; c++)
655
656
            writeSample((long) (floatScale * (floatOffset + sampleBuffer[offset])));
```

```
657
                 offset ++;
658
659
660
              frameCounter ++;
661
662
663
           return numFramesToWrite;
664
665
666
        public int writeFrames(double[][] sampleBuffer, int numFramesToWrite) throws IOException,

→ WavFileException

667
           return writeFrames(sampleBuffer, 0, numFramesToWrite);
668
669
670
        public int writeFrames(double[][] sampleBuffer, int offset, int numFramesToWrite) throws
671
          → IOException, WavFileException
672
           if (ioState != IOState.WRITING) throw new IOException("Cannot_write_to_WavFile_instance");
673
674
675
           for (int f=0; f<numFramesToWrite; f++)
676
              if (frameCounter == numFrames) return f;
677
678
679
              for (int c=0; c<numChannels; c++) writeSample((long) (floatScale * (floatOffset +

    sampleBuffer[c][offset])));
680
681
              offset ++;
              frameCounter ++;
682
683
684
685
           return numFramesToWrite;
686
687
688
689
        public void close() throws IOException
690
691
               Close the input stream and set to null
692
           if (iStream != null)
693
694
              iStream.close();
695
              iStream = null;
696
697
698
           if (oStream != null)
699
700
                Write out anything still in the local buffer
701
               \textbf{if} \hspace{0.1in} (\hspace{0.1em} \texttt{bufferPointer} \hspace{0.1em} > \hspace{0.1em} 0\hspace{0.1em}) \hspace{0.1em} \hspace{0.1em} \texttt{oStream.write} \hspace{0.1em} (\hspace{0.1em} \texttt{buffer} \hspace{0.1em}, \hspace{0.1em} 0\hspace{0.1em}, \hspace{0.1em} \hspace{0.1em} \texttt{bufferPointer}) \hspace{0.1em};
702
703
                'If an extra byte is required for word alignment, add it to the end
               \textbf{if} \hspace{0.1in} (\hspace{0.1em} \text{word} \hspace{0.1em} A \hspace{0.1em} \text{lig} \hspace{0.1em} n \hspace{0.1em} A \hspace{0.1em} \text{djust} \hspace{0.1em} ) \hspace{0.1em} \text{oStream.write} \hspace{0.1em} (\hspace{0.1em} 0\hspace{0.1em}) \hspace{0.1em} \hspace{0.1em} ; \\
704
705
706
              // Close the stream and set to null
707
              oStream.close();
708
              oStream = null;
709
710
711
           // Flag that the stream is closed
712
           ioState = IOState.CLOSED;
713
714
        public void display()
715
716
        {
717
           display (System.out);
718
719
720
        public void display (PrintStream out)
721
           out.\mathbf{printf}("File: \clim{1}{2}\%s \n", file);
722
           out.printf("Channels: \%d, \Frames: \%d\n", numChannels, numFrames);
723
           out.\mathbf{\hat{printf}}("IO \cup State: \cup \%s \setminus n", ioState);
724
           out. \, \textbf{printf}(\, "Sample\_Rate: \, \_\%d \,, \, \_Block\_Align: \, \_\%d \, \backslash \, n" \,\,, \,\, sampleRate \,, \,\, blockAlign \,) \,\, ;
725
           out. \textbf{printf}("Valid\_Bits:\_\%d,\_Bytes\_per\_sample:\_\%d \setminus n", validBits, bytesPerSample);\\
726
727
728
729
        public static void main (String [] args)
```

```
730
         if (args.length < 1)
731
732
           System.err.println("Must_supply_filename");
733
734
           System. exit(1);
735
736
737
         t r y
738
739
           for (String filename: args)
740
741
              WavFile readWavFile = openWavFile(new File(filename));
              readWavFile.display();
742
743
              long numFrames = readWavFile.getNumFrames();
744
              int \ num Channels = \ read Wav File. \, get Num Channels \, (\, ) \; ; \\
745
746
              int validBits = readWavFile.getValidBits();
747
              long sampleRate = readWavFile.getSampleRate();
748
749
              WavFile writeWavFile = newWavFile(new File("out.wav"), numChannels, numFrames,

→ validBits, sampleRate);
750
              final int BUF SIZE = 5001;
751
752
753
                int [] buffer = new int [BUF SIZE * numChannels];
                long[] buffer = new long[BUF SIZE * numChannels];
754
755
              double [] buffer = new double [BUF_SIZE * numChannels];
756
757
              int framesRead = 0;
758
              int framesWritten = 0;
759
760
              do
761
                \begin{array}{lll} framesRead &=& readWavFile.readFrames(buffer, BUF\_SIZE); \\ framesWritten &=& writeWavFile.writeFrames(buffer, BUF\_SIZE); \\ \end{array}
762
763
                System.out.printf("%d_%d\n", framesRead, framesWritten);
764
765
766
              while (framesRead != 0);
767
              read Wav File. close ();
768
769
              writeWavFile.close();
770
           }
771
772
           WavFile writeWavFile = newWavFile(new File("out2.wav"), 1, 10, 23, 44100);
           double\,[\,]\quad b\,u\,ff\,e\,r\ =\ new\ d\,ou\,ble\,[\,1\,0\,]\,;
773
774
           writeWavFile.writeFrames(buffer, 10);
775
           writeWavFile.close();
776
777
         catch (Exception e)
778
           System.err.println(e);
779
780
           e.printStackTrace();
781
782
```

Листинг 3: WavFileException.java

```
1
   package com.external;
 2
3
    * Exception for WavFile Class
 4
    * http://www.labbookpages.co.uk
6
 7
    * @author A. Greensted
8
    */
9
10
   public class WavFileException extends Exception
11
12
     public WavFileException()
13
14
       super();
15
16
17
     public WavFileException (String message)
18
```

```
19
        super(message);
^{20}
     }
21
     public WavFileException (String message, Throwable cause)
^{22}
^{23}
24
        super(message, cause);
^{25}
26
27
     public WavFileException (Throwable cause)
28
29
        super(cause);
30
     }
31
```

Листинг 4: LightLevel.java

```
package com.sunradio.core;
2
3
  import com. sunradio. math. Scale;
 4
5
  import static java.lang.Math.*;
 6
  import java.util.Random;
 7
8
9
   * LightLevel describes how we get an array of data with current light level
10
   * @author V. Kremneva
11
12
   public class LightLevel {
13
14
15
        * Get randomised light level
16
17
        * @param amount of how many measurement of light level we want
18
        * @return scaled to [0;1] array of light values
19
^{20}
       private static double [] getFakeLightLevel(int amount, Double minAmplitude, Double
       21
           //fake it 'till you make it
Integer[] lightLevel = new Integer[amount];
^{22}
23
24
           for (int i = 0; i < amount; i++)
               lightLevel[i] \ = \ (int) (sin(2 * PI * i / 100)*1000); \ // \textbf{for} \ smoothness
25
26
27
28
           return Scale.run(lightLevel, minAmplitude, maxAmplitude);
       }
29
30
31
32
        * Get light level from Arduino //TODO: specify
33
        \ast @param amount of how many measurement of light level we want
34
35
        * @return scaled to [0;1] array of light values
36
37
       static double [] getLightLevel (int amount, Double minAmplitude, Double maxAmplitude) {
38
           return getFakeLightLevel(amount, minAmplitude, maxAmplitude);
39
40
41
       /**
        * Get light level
42
        * @param values an array to whom get light level
43
        * @return light level for 'values
44
45
46
       public static double[] getLightLevel(double[] values) {
47
           double maxVal, minVal;
48
           maxVal = Double.MIN VALUE; minVal = Double.MAX VALUE;
49
           for (double val: values)
50
                if (maxVal > val) maxVal = val;
                if (maxVal < val) minVal = val;
51
52
           }
53
54
           return getLightLevel(values.length, minVal, maxVal);
55
56
57
       private static int getAverageFakeLevel() {
58
           Random random = new Random();
59
           return random.nextInt();
```

```
60 | 8 | 61 | 62 | static int getAverageLightLevel(double[] values) { 63 | return getAverageFakeLevel(); 64 | 8 | 65 | }
```

Листинг 5: SunRadio.java

```
package com. sunradio. core;
  1
 2
     import com. external. WavFile;
     import com.sunradio.math.DFTInverse;
  4
  5
     import com. sunradio. math. DFTStraight;
     import com. sunradio. math. Filter;
     import com.sunradio.math.ToneModulation;
  8
 9
     import java.io.File;
10
11
     import javax.sound.sampled.AudioInputStream;
     import javax.sound.sampled.AudioSystem;
12
13
     import javax.sound.sampled.Clip;
14
15
16
17
       * @author V. Kremneva
18
19
     public class SunRadio {
              private final int FRAMES = 2048; //amount of frames to read private final int OVERLAP = 16; //coefficient of overlap
20
21
^{22}
23
              private WavFile wavInput; //input file
24
              private WavFile wavOutput; //output file
25
              private \ int \ bufferIndAmount; \ //amount \ of \ indexes \ in \ 'buffer' \ array \ needed \ to \ read \ data 
26
27
              private int overlapIndAmount; //amount of indexes to work with overlap
              private int offset; //amount of new frames read in each step of cycle
2.8
29
              private int outputBufferIndAmount; //amount of indexes in am array to write
30
              private long wholeIndAmount; //amount of pieces to read in whole file
31
32
33
                * Open file and set some fields depending on it
34
35
                * @param inputPath path to file to open
36
              private void openWavFile(String inputPath) {
37
38
                      try {
39
40
                               wavInput = WavFile.openWavFile(new File(inputPath));
41
42
                      } catch (Exception e) {
                               System.err.println(e.toString());
43
44
45
46
                      int numChannels = wavInput.getNumChannels();
                      bufferIndAmount \ = \ FRAMES \ * \ numChannels;
47
                      overlapIndAmount\ =\ OVERLAP\ *\ numChannels;
48
49
                      offset = FRAMES / OVERLAP;
50
                      outputBufferIndAmount = bufferIndAmount + overlapIndAmount;
                      wholeIndAmount = wavInput.getNumFrames() * numChannels;
51
52
              }
53
54
                * Create empty output file like input file but stretched
55
56
57
                * @param outputPath path where to create output file
58
59
              private void createStretchedOutputFile(String outputPath, int stretch) {
60
                      try {
61
62
                               wavOutput = WavFile.newWavFile(new File(outputPath),
                                                wavInput.getNumChannels()\ ,\ wavInput.getNumFrames()\ *\ stretch\ ,
63
64
                                                wavInput.getValidBits(), wavInput.getSampleRate());
65
                      } catch (Exception e) {
66
67
                               System.err.println(e.toString());
```

```
68
            }
 69
70
        private void run() {
 71
            \tt double[] \quad buffer = new \ double[bufferIndAmount];
 72
 73
            double [] output Buffer = new double [output BufferInd Amount];
            double [] outputWindowFunction;
 74
 75
            int lightLevel, frames read;
 76
            DFTStraight transformable;
 77
            ToneModulation toneModulation;
 78
 79
            transformable = new DFTStraight();
80
            toneModulation = new ToneModulation(bufferIndAmount);
 81
 82
            int counter = 0;
83
            try
                do {
 84
 85
                     //read next 'FRAMES' into buffer — amplitudes(t)
                    frames_read = wavInput.readFramesWithOverlap(buffer, FRAMES, OVERLAP);
86
 87
                     //get current level of light
 88
                     lightLevel = LightLevel.getAverageLightLevel(buffer);
 89
 90
                     91
 92
                         if (!(wavInput.getFrameCounter() == offset) && !(wavInput.getFrameCounter
93
        \hookrightarrow () == (wholeIndAmount - offset))) {
                             buffer = Filter.apply(buffer, Filter.BlackmanNuttall(bufferIndAmount))
94
        \hookrightarrow :
 95
                         }
 96
97
                         //run Fourier transform
98
                         transformable.run(buffer);
99
                         //stretch in 'light level' times
100
                         toneModulation.setCurrentData(transformable);
101
                         transformable.setData(toneModulation.stretch(lightLevel));
102
103
                         //run inverse Fourier transform
104
105
                         buffer = DFTInverse.run(transformable.getData());
106
107
                         //apply output window function
                         outputWindowFunction = Filter.getOutputWindowFunc(Filter.BlackmanNuttall(Filter.BlackmanNuttall(Filter))) \\
108
        → bufferIndAmount));
109
                         buffer = Filter.apply(buffer, outputWindowFunction);
110
                         for (int j = 0; j < bufferIndAmount; <math>j++)
111
                             outputBuffer[j + i] += buffer[j];
112
113
114
                         //read next 'FRAMES' into buffer — amplitudes(t)
                         frames_read = wavInput.readFramesWithOverlap(buffer, FRAMES, OVERLAP);
115
116
117
118
                     //write data to new .waw file
                    wavOutput .writeFrames (outputBuffer, FRAMES);
119
120
121
                     //move data for overlap
122
                     outputBuffer = move(outputBuffer, overlapIndAmount);
123
124
                    toneModulation.setPreviousData(transformable);
125
                     counter++:
                } while (frames read != 0);
126
127
                //todo: adjust volume
128
129
                //todo: fasten velocity of playback
130
                //play(outputPath);
131
132
            } catch (Exception e) {
133
134
                System.err.println(e.toString());
135
136
137
138
        private void closeFiles() {
139
            try {
140
```

```
141
                wavInput.close();
142
                wavOutput.close();
143
144
            } catch (Exception e) {
145
                System.err.println(e.toString());
146
147
        }
148
149
        /**
150
            Move data to the left with filling with 0
         * @param data data to move
151
152
         * @param offset amount of steps to move
         st @return array with nulls in the end and 'data' values moved on offset
153
154
        public static double[] move(double[] data, int offset) {
155
156
            double [] result = new double [data.length];
157
158
            System.arraycopy (data, offset, result, 0, data.length - offset);
159
160
            return result;
161
        }
162
        /*private static void play(String pathname) {
163
164
165
                 Clip c = AudioSystem.getClip();
166
                 AudioInputStream ais = AudioSystem.getAudioInputStream(new File(pathname));
167
168
                 c.open(ais);
169
                c.loop(0);
170
171
                Thread.sleep(1000);
172
            } catch (Exception e) {
173
                System.err.println(e.toString());
174
        }*/
175
176
        public static void main(String[] args) {
177
178
           if (args.length < 2) throw new IllegalArgumentException("As_the_arguments_of_the_
179
        → program_"
                     "input_path_and_output_path_are_needed.");
180
181
            SunRadio radio = new SunRadio();
182
183
            radio.openWavFile(args[0]);
184
185
186
            radio.run();
187
188
            radio.closeFiles();
189
        }
190
```

Листинг 6: AM. java

```
1
  package com.sunradio.math;
2
3
 4
   * Amplitude modulation
   * @author V. Kremneva
5
 7
   public class AM {
8
9
10
        st Modulate values according to conditions with the coefficient of modulation -0.65.
         Assume 'values' as amplitude values therefore perform an Amplitude Modulation.
11
12
        * @param values an array with amplitude values to modulate
13
        * @param conditions an array of modulation conditions for each amplitude value
14
        * @return a double array of modulated values of amplitudes
15
16
        * @throws IllegalArgumentException if amount of conditions is less than amount of values
17
18
       public \ static \ double [] \ modulate (double [] \ values , \ double [] \ conditions) \ \{
19
           return modulate (values, conditions, -0.65);
20
21
22
```

```
^{23}
       * Modulate values according to conditions.
24
       * Assume 'values' as amplitude values therefore perform an Amplitude Modulation.
25
^{26}
       * @param values an array with amplitude values to modulate
27
       * @param conditions an array of modulation conditions for each amplitude value
28
       * @param modulationCoeff modulation coefficient which picks up by trial and error
29
       * @return a double array of modulated values of amplitudes
30
       * @throws IllegalArgumentException if amount of conditions is less than amount of values
31
32
      private static double [] modulate (double [] values, double [] conditions, double

→ modulationCoeff)
33
              throws IllegalArgumentException {
34
          35
36
37
                       "should_be_equal_or_more_than_size_of_an_array_of_values.\n" + \,
                      "Conditions size = " + conditions length + " . Values size = " + values.
38
      \hookrightarrow length);
39
          }
40
41
          double [] modulated;
          modulated = new double [values.length];
42
43
44
          double maxCond:
45
          \max Cond = conditions[0];
46
          for (int i = 1; i < values.length; i++)
47
               if (conditions[i] > maxCond) maxCond = conditions[i];
48
49
          for (int i = 0; i < values.length; i++)
              modulated[i] = values[i] * (1 + modulationCoeff * conditions[i] / Math.abs(maxCond
50
      → ));
51
52
          return modulated;
53
      }
54
```

Листинг 7: DFTStraight.java

```
package com.sunradio.math;
 1
  import com. external. Complex;
  import java.util.Arrays;
 4
5
  import static java.lang.Math.*;
 7
 8
    * Straight discrete Fourier transform.
10
    * @author V. Kremneva
11
  public class DFTStraight {
12
13
14
       private Complex[] data; //Complex data
       private double maxAmplitude; //maximum value of real amplitude in 'data' array
15
       private double minAmplitude; //minimum value of real amplitude in 'data' array private int size; //size of 'data' array
16
17
       private boolean is Transformed; //flag whether was 'data' transformed by DFT or not
18
19
       public DFTStraight() {
20
21
            \max Amplitude = 0.0;
22
            minAmplitude = 0.0;
23
            size = 0:
24
            isTransformed = false;
^{25}
       }
26
27
       public double getMaxAmplitude() {
28
           return max Amplitude;
29
30
31
       public double getMinAmplitude() {
32
            return minAmplitude;
33
34
35
       public boolean isTransformed() {
           return isTransformed;
36
37
38
```

```
39
            public Complex[] getData() {
 40
                 return data;
 41
 42
 43
            public int getSize() {
 44
                 return size;
 45
 46
 47
            /** Get a double phases from 'data' array.
 48
             * @return a double array which contains phase values
 49
 50
            public double[] getPhases() {
 51
 52
                  double [] phases = new double [size];
 53
                  double allowance;
                  for (int i = 0; i < size; i++) {
 54
 55
                        \begin{array}{lll} \textbf{if} & (\,\mathrm{data}\,[\,\mathrm{i}\,]\,.\,\mathrm{re}\,(\,) \, > \, 0\,) & \mathrm{allow}\,\mathrm{ance} \, = \, 0\,; \\ \textbf{else} & \textbf{if} & (\,\mathrm{data}\,[\,\mathrm{i}\,]\,.\,\mathrm{im}\,(\,) \, > \, 0\,) & \mathrm{allow}\,\mathrm{ance} \, = \, \mathrm{PI}\,; \end{array}
 56
 57
 58
                        else allowance = -PI;
 59
                        phases[i] = allowance + atan(data[i].im() / data[i].re());
 60
 61
                 }
 62
 63
                 return phases;
 64
           }
 65
 66
            /** Get a phase value on specific harmonic
 67
 68
             * @param n frequency value of harmonic
 69
             * @return double value of phase of harmonic
 70
 71
            public double getPhase(int n) {
                 double allowance;
 72
                  \mbox{\bf if} \ (\, d\, at\, a\, [\, n\, ]\, .\,\, r\, e\, (\,) \ > \ 0\, ) \ \ allow\, a\, n\, c\, e \ = \ 0\, ; 
 73
 74
                  else if (data[n].im() > 0) allowance = PI;
                 \mathbf{else} \ \ \mathbf{allowance} \ = \ -\mathrm{PI} \, ;
 75
 76
 77
                 return allowance + atan(data[n].im() / data[n].re());
 78
           }
 79
 80
            public static double getPhase(Complex[] data, int n) {
 81
                  double allowance;
 82
                  if (data[n].re() > 0) allowance = 0;
                  else if (data[n].im() > 0) allowance = PI;
 83
 84
                  else allowance = -PI;
 85
 86
                 \textbf{return} \hspace{0.2cm} \textbf{allowance} \hspace{0.1cm} + \hspace{0.1cm} \textbf{atan} \hspace{0.1cm} (\hspace{0.1cm} \textbf{data} \hspace{0.1cm} [\hspace{0.1cm} n \hspace{0.1cm}] \hspace{0.1cm} . \hspace{0.1cm} \textbf{im} \hspace{0.1cm} (\hspace{0.1cm}) \hspace{0.1cm} / \hspace{0.1cm} \hspace{0.1cm} \textbf{data} \hspace{0.1cm} [\hspace{0.1cm} n \hspace{0.1cm}] \hspace{0.1cm} . \hspace{0.1cm} \textbf{re} \hspace{0.1cm} (\hspace{0.1cm}) \hspace{0.1cm} ) \hspace{0.1cm} ;
 87
 88
            /** Get a double amplitudes from 'data' array.
 89
 90
 91
             * @return a double array which contains amplitude values
 92
 93
            public double[] getAmplitudes() {
 94
                 double[] amplitudes = new double[size];
 95
 96
                  for (int i = 0; i < size; i++)
                        amplitudes[i] = data[i].abs() / ((size - 1) * 2); // '-1)*2' due to cutting in
 97
           \hookrightarrow half
 98
                 return amplitudes;
 99
100\,
           }
101
102
            static double [] get Amplitudes (Complex [] data) {
103
                 double [] amplitudes = new double [data.length];
104
105
                  for (int i = 0; i < data.length; i++)
106
                       amplitudes[i] = data[i].abs() / ((data.length - 1) * 2); // '-1)*2' due to cutting
                 in half
107
                 return amplitudes;
108
109
110
            static double [ ] getPhases (Complex [ ] data) {
111
112
                  double [] phases = new double [data.length];
```

```
double allowance;
113
             for (int i = 0; i < phases.length; <math>i++) {
114
115
116
                 if (data[i].re() > 0) allowance = 0;
117
                 else if (data[i].im() > 0) allowance = PI;
118
                 else allowance = -PI;
119
120
                 phases[i] = allowance + atan(data[i].im() / data[i].re());
121
             }
122
123
             return phases;
124
        }
125
126
        /** Get an amplitude value on specific harmonic
127
         * @param n frequency value of harmonic
128
129
         * @return double value of amplitude of harmonic
130
        public double getAmplitude(int n) {
131
132
             return data [n].abs() / ((size - 1) * 2);
             // '-1) *2' due to cutting in half
133
134
135
136
        public void setData(Complex[] newData) {
137
             size = newData.length;
138
             data = Arrays.copyOf(newData, size);
139
140
             double max, min;
             max = Double.MIN VALUE; min = Double.MAX VALUE;
141
142
             for (int i = 0; i < size; i++) {
143
                 if (this.getAmplitude(i) > max) max = this.getAmplitude(i);
                 if (this.getAmplitude(i) < min) min = this.getAmplitude(i);</pre>
144
145
146
             max Amplitude = max; min Amplitude = min;
147
        }
148
149
        /**
150
         * Cut Complex array in two pieces.
151
         * We need this because the periods of the input data become split into "positive"
         * and "negative" frequency complex components. As a result, only half of array * contains data we are interested in and the rest of array is just a reflection with
152
153
         * opposite sign.
154
155
156
         * @param dataToCut Complex data we want to be cut
157
        private void cutDataInHalf(Complex[] dataToCut) {
158
159
             size = size / 2 + 1; //'+1' to include center value
160
161\,
             data = new Complex[size];
162
             data = Arrays.copyOf(dataToCut, size);
163
164
        /** Run transform with search of max and min value of amplitude.
165
166
167
          * @param buffer an array of the magnitudes
168
         * @return a Complex array which contains amplitude and phase values
169
         * @throws IllegalArgumentException if buffer is empty
170
        public Complex[] run(double[] buffer) throws IllegalArgumentException {
171
172
173
             size = buffer.length:
174
             Double realAmplitude;
175
             Complex cBuffer, expDegree, tempData[];
176
             tempData = new Complex[size];
177
178
             if (size == 0) throw new IllegalArgumentException("Size_of_a_buffer_cannot_be_<_1.\n_
        \hookrightarrow \ \operatorname{size} \ \_= \ \_" \ + \ \operatorname{size} \ ) \ ;
179
180
             max Amplitude = Double.MIN VALUE:
             minAmplitude = Double.MAX VALUE;
181
182
             for (int i = 0; i < size; i++) {
183
                 tempData[i] = new Complex(0, 0);
184
185
                 for (int j = 0; j < size; j++) {
                      cBuffer = new Complex (buffer [j]);
186
                      expDegree = new Complex(0, -2 * PI * j * i / size);
187
```

```
188
                          tempData[i] = tempData[i].add(cBuffer.mult(expDegree.exp()));
189
                     }
190
191
                     realAmplitude = tempData[i].abs() / size;
192
                     \textbf{if} \hspace{0.2cm} (\hspace{0.1cm} real Amplitude \hspace{0.1cm} > \hspace{0.1cm} max Amplitude \hspace{0.1cm} | \hspace{0.1cm} max Amplitude \hspace{0.1cm} = \hspace{0.1cm} real Amplitude \hspace{0.1cm} ;
193
                     if (realAmplitude < minAmplitude) minAmplitude = realAmplitude;</pre>
194
195
196
                isTransformed = true;
197
                cutDataInHalf(tempData);
198
199
200
               return data;
201
          }
202
203
204
           * Change phase values in 'data' without changing amplitudes
205
           * @param newPhases an array of phase values we want to apply
206
207
208
          void apply New Phases (double [] new Phases) {
209
                double a, b, allowance;
                for (int i = 0; i < size; i++) {
210
211
                     if (data[i].re() > 0) allowance = 0;
212
                     else if (data[i].im() > 0) allowance = -PI;
213
                     else allowance = PI;
214
                     a = data[i].abs() / sqrt(1 + pow(tan(newPhases[i] + allowance), 2.0));
215
216
                     b = a * tan(newPhases[i] + allowance);
217
218
                     //we get 'b' from equation for phase and 'a' from my condition:
219
                     //i want the real amplitudes be the same
220
221
                     data[i] = new Complex(a, b);
222
               }
223
          }
224
          static\ Complex []\ apply New Phases (double []\ new Phases\ ,\ Complex []\ old Data)\ \{
225
226
               double a, b, allowance;
               Complex\,[\,]\quad result\ =\ new\ Complex\,[\,new\,P\,hases\,.\,lengt\,h\,]\,;
227
228
                \label{eq:formula} \textbf{for} \hspace{0.2cm} (\hspace{0.1cm} \texttt{int} \hspace{0.2cm} \hspace{0.1cm} \texttt{i} \hspace{0.1cm} = \hspace{0.1cm} \texttt{0}\hspace{0.1cm} ; \hspace{0.2cm} \texttt{i} \hspace{0.1cm} < \hspace{0.1cm} \texttt{newPhases.length}\hspace{0.1cm} ; \hspace{0.2cm} \texttt{i} \hspace{0.1cm} + \hspace{0.1cm} ) \hspace{0.1cm} \hspace{0.1cm} \{
                     if (oldData[i].re() > 0) allowance = 0;
229
230
                     else if (oldData[i].im() > 0) allowance = -PI;
231
                     else allowance = PI;
232
233
                     a = oldData[i].abs() / sqrt(1 + pow(tan(newPhases[i] + allowance), 2.0));
                     b = a * tan(newPhases[i] + allowance);
234
235
                     //we get 'b' from equation for phase and 'a' from my condition:
236
237
                     //i want the real amplitudes be the same
238
239
                     result[i] = new Complex(a, b);
240
241
                return result;
242
          }
243
244
           * Change amplitude values in 'data' without changing phases
245
246
247
           * @param newAmplitudes an array of amplitude values we want to apply
248
          void apply New Amplitudes (double [] new Amplitudes) {
249
250
                 int sign;
251
                 double a, b;
252
                 for (int i = 0; i < size; i++) {
253
                      if (this.getPhase(i) < 0) sign = -1;
254
                      else sign = 1;
255
                      b = pow(data[i].im(), 2.0) * pow(newAmplitudes[i], 2.0) * pow(size, 2.0);
                      b \, = \, b \, / \, \left( \, pow(\, data \, [\,\, i \,\,] \, . \, re \, (\,) \,\, , \,\, \, 2.0 \,) \,\, + \,\, pow(\, data \, [\,\, i \,\,] \, . \, im \, (\,) \,\, , \,\, \, 2.0 \,) \,\, \right);
256
257
                      b = sqrt(b);
258
                      a = sign * sqrt(pow(newAmplitudes[i], 2.0) * pow(size, 2.0) - pow(b, 2.0));
259
260
261
                     //we get 'a' from equation for real amplitude and 'b' from my condition:
262
                     //i want the real phase be the same
263
```

```
264
                 data[i] = new Complex(a, b);
265
              }
        }
266
^{267}
        static Complex [] apply New Amplitudes (double [] new Amplitudes , Complex [] old Data) {
268
269
             int sign;
             double a, b;
270
             Complex [] \quad result = new \quad Complex [new Amplitudes.length];
271
272
             for (int i = 0; i < newAmplitudes.length; <math>i++) {
273
                 if (getPhase(oldData, i) < 0) sign = -1;
274
                 else sign = 1;
275
                 b = pow(oldData[i].im(), 2.0) * pow(newAmplitudes[i], 2.0) * pow(newAmplitudes.
        \hookrightarrow length, 2.0);
                 b = b / (pow(oldData[i].re(), 2.0) + pow(oldData[i].im(), 2.0));
276
277
                 b = sqrt(b);
278
279
                 a = sign * sqrt (pow(newAmplitudes[i], 2.0) * pow(newAmplitudes.length, 2.0) - pow(
        \hookrightarrow b, 2.0));
280
281
                 //we get 'a' from equation for real amplitude and 'b' from my condition:
282
                 //i want the real phase be the same
283
                 result[i] = new Complex(a, b);
284
285
             }
286
287
             return result;
288
289
```

Листинг 8: DFTInverse.java

```
1
  package com.sunradio.math;
 2
 3
   import com. external. Complex:
 4
 5
   import java.util.Arrays;
 6
 7
   import static java.lang.Math.PI;
 8
9
10
   * Inverse discrete Fourier transform.
   * @author V. Kremneva
11
12
   public class DFTInverse {
13
14
15
16
        * Restore frequency complex components from just one half.
17
18
        * @param dataToRestore array of data to restore
19
        * @return full spectrum Complex array
20
21
       private static Complex[] restoreData(Complex[] dataToRestore) {
22
^{23}
           int oldSize = dataToRestore.length;
^{24}
           int newSize = oldSize * 2 - 2;
25
           Complex[] result = new Complex[newSize];
^{26}
27
           System.arraycopy (dataToRestore, 0, result, 0, oldSize);
28
29
           for (int i = oldSize; i < newSize; i++)
                result [i] = dataToRestore[newSize - i].conj();
30
31
^{32}
           return result;
33
       }
34
       /** Run transform.
35
36
37
        * @param transformed a Complex array which contains amplitude and phase values
38
        * @return an array of amplitudes
39
40
       public static double[] run(Complex[] transformed){
41
42
           Complex [] data;
43
           data = restoreData(transformed);
44
45
           int size = data.length;
```

```
46
         Complex exp degree, magnitude;
47
         double [] result = new double [size];
48
49
         for (int i = 0; i < size; i++) {
50
             magnitude = new Complex(0, 0);
51
             52
53
54
                 magnitude = magnitude.add(data[j].mult(exp degree.exp()));
55
             }
56
57
             result[i] = magnitude.re() / size;
58
         }
59
60
         return result;
61
62
```

Листинг 9: Filter.java

```
package com.sunradio.math;
 3
   import static java.lang.Math.*;
 4
 5
 6
    * Filter functions
 7
    * @author V. Kremneva
9
   public class Filter {
10
11
          * Window function of Blackman-Nuttall
12
13
14
          * @param size is the length of array to be windowed
15
          * @return an array of values of this window
16
         public static double[] BlackmanNuttall(int size) {
17
18
              double [] result = new double [size];
19
              //constants from formula of -BlackmanNuttall window
20
21
              final\ double\ A0 = 0.3635819;
22
              final double A1 = 0.4891775;
              final double A2 = 0.1365995;
23
^{24}
              final double A3 = 0.0106411;
25
^{26}
              double firstCos, secondCos, thirdCos;
              {f for} (int i = 0; i < size; i++) {
27
                   28
^{29}
30
31
32
                    result [i] = A0 - firstCos + secondCos - thirdCos;
33
              }
34
35
              return result;
36
37
38
          * Window function of Blackman-Nuttall
39
40
41
          * @param n point where value of this function is needed
          st @param size is the length of array to be windowed
42
43
          * @return value of this window function in 'n'
44
45
         public static double BlackmanNuttall(int n, int size) {
46
              //constants from formula of -BlackmanNuttall window
              final\ double\ A0 = \ 0.3635819;
47
48
              final double A1 = 0.4891775;
              final double A2 = 0.1365995;
49
50
              final double A3 = 0.0106411;
51
52
              double firstCos, secondCos, thirdCos;
              \begin{array}{l} {\rm first\,Cos} \,=\, A1 \,*\, \cos \left( \left( \, 2 \,*\, PI \,*\, n \right) \,/\, \left( \, {\rm size} \,-\, 1 \right) \right); \\ {\rm second\,Cos} \,=\, A2 \,*\, \cos \left( \left( \, 4 \,*\, PI \,*\, n \right) \,/\, \left( \, {\rm size} \,-\, 1 \right) \right); \\ {\rm t\,hird\,Cos} \,=\, A3 \,*\, \cos \left( \left( \, 6 \,*\, PI \,*\, n \right) \,/\, \left( \, {\rm size} \,-\, 1 \right) \right); \end{array}
53
54
55
```

```
return A0 - firstCos + secondCos - thirdCos;
 57
 58
        }
59
 60
 61
         * Apply window function to an array of values
 62
 63
         * @param amplitudes an array to be windowed
64
         \ast @param windowFunc an array with values of window {\bf function}
         * @return an array with transformed 'amplitudes' according to 'windowFunc'  
* @throws IllegalArgumentException \mathbf{if} length of 'windowFunc' is less than length of '
 65
 66

→ amplitudes '

 67
        public static double[] apply(double[] amplitudes, double[] windowFunc)
68
 69
                throws IllegalArgumentException {
 70
 71
            int size = amplitudes.length;
 72
 73
            if (size > windowFunc.length) throw new IllegalArgumentException("Length_of_window_

→ function jis too small");
 74
            75
 76
 77
                 result[i] = amplitudes[i] * windowFunc[i];
 78
 79
            return result;
 80
        }
81
 82
 83
         * Remove noise from amplitudes array
 84
 85
         * @param toDenoise array to remove noise from
         * @return sort of clean array
86
 87
        88
89
 90
            double maxAmplitude, eps;
91
            max Amplitude = toDenoise [0];
92
            for (double val: toDenoise)
 93
94
                 if(val > maxAmplitude) maxAmplitude = val;
 95
            eps = maxAmplitude * DENOISE COEFF;
96
97
            for (int i = 0; i < toDenoise.length; i++)
98
                 if (abs(toDenoise[i]) < eps) toDenoise[i] = 0.0;</pre>
99
100
            return toDenoise;
101
        }
102
103
104
         * Get function to filter output data before writing to file
105
106
           @param inputWindowFunc function used to filter input data
107
         * @return function to filter output data
108
109
        public static double[] getOutputWindowFunc(double[] inputWindowFunc) {
110
            double [] outputWindowFunc = new double [inputWindowFunc.length];
111
            double sum = 0.0;
112
            for (double val: inputWindowFunc)
113
114
                sum += val * val;
115
            for (int i = 0; i < outputWindowFunc.length; i++)</pre>
116
                outputWindowFunc[i] = inputWindowFunc[i] / sum;
117
118
119
            return outputWindowFunc;
120
121
```

Листинг 10: Interpolation.java

```
package com.sunradio.math;

/**

* Class to interpolate.

* * @author V.Kremneva
```

```
class Interpolation {
9
10
        * Perform linear interpolation by two points
11
        * @param x0 x from the first point
12
        * @param y0 f(x) from the first point
13
        * @param x1 x from the second point
        * @param y1 f(x) from the second point
14
15
        * @param x x from the point we are interested in
16
        * @ return f(x) from the point we are interested in
17
18
       static double linear By X (double x0, double y0, double x1, double y1, double x) {
           \textbf{return} \ \ y0 \ + \ (x \ - \ x0\,) \ * \ (y1 \ - \ y0\,) \ / \ (x1 \ - \ x0\,) \,;
19
20
21
22
23
        * Perform linear interpolation by two points
24
        * @param x0 x from the first point
25
        * @param y0 f(x) from the first point
26
        * @param x1 x from the second point
27
        * @param y1 f(x) from the second point
28
        * @param y f(x) from the point we are interested in
29
        * @return x from the point we are interested in
30
31
       static double linearByY(double x0, double y0, double x1, double y1, double y) {
32
           return x0 + (y - y0) * (x1 - x0) / (y1 - y0);
33
34
```

Листинг 11: Scale.java

```
1
  package com.sunradio.math;
2
3
 4
   * Helps to scale data
5
   * @author V. Kremneva
6
 7
   public class Scale {
 8
9
10
        * Scale numeric data
11
        * @param arr is an array of data to scale
12
        * @param from is the lower bound of scaling
13
        * @param to is the upper bound of scaling
14
15
        * @return scaled array of data
16
        * @throws IllegalArgumentException if the data array is empty
17
18
       public static <T extends java.lang.Number> double [] run(T[] arr, double from, double to)
19
               throws IllegalArgumentException {
20
21
           int size = arr.length;
           if (size < 1) throw new IllegalArgumentException("Array_of_the_values_cannot_be_empty"
22
23
           if (from > to) {
24
                {\tt double\ temp\ =\ from\ ;}
^{25}
               from = to; to = temp;
26
           }
27
28
           double maxLevel = arr [0].doubleValue();
29
           double minLevel = arr [0].doubleValue();
30
           double current;
31
           for (int i = 1; i < size; i++) {
32
                current = arr[i].doubleValue();
33
34
                if (current > maxLevel) maxLevel = current;
35
                else if (current < minLevel) minLevel = current;</pre>
36
           }
37
           double step = (to - from) / (maxLevel - minLevel);
38
39
40
           double \ [] \ scaledLightLevel = new \ double \ [size];
41
           for (int i = 0; i < size; i++)
                scaledLightLevel[i] = (arr[i].doubleValue() - minLevel) * step + from;
42
43
44
           return scaledLightLevel;
```

```
egin{array}{c} 45 \ 46 \ \end{array} \}
```

Листинг 12: ToneModulation.java

```
package com.sunradio.math;
 3
   import com. external. Complex;
 5
 6
    * Tone modulation.
 7
 8
    * @author V. Kremneva
9
   public class ToneModulation {
10
        private \ double \cite{Lagrangian} \ previous \ Phases; \ //values \ of \ previous \ phases
11
       private double [] previous Amplitudes; //values of previous amplitudes private double [] current Phases; //values of current phases
12
13
14
        private double[] currentAmplitudes; //values of current amplitudes
15
        private int length; //length of all of the arrays
16
17
18
        private \ boolean \ previous Is Set; \ //indicates \ whether \ previous \ data \ was \ \textbf{set}
        private boolean currentIsSet; //indicates whether current data was set
19
^{20}
21
        public ToneModulation() {
22
            previousIsSet = false;
23
            currentIsSet = false;
24
^{25}
^{26}
        public ToneModulation(int size) {
27
            l\,en\,g\,t\,h\ =\ s\,i\,z\,e\;;
28
29
            previousPhases = new double[size];
30
            previousAmplitudes = new double[size];
31
32
            previousIsSet = true:
33
            currentIsSet = false;
34
       }
35
36
        public void setPreviousData(DFTStraight prevData) {
37
            previous Amplitudes = prevData.get Amplitudes();
38
            previousPhases = prevData.getPhases();
39
40
            previousIsSet = true:
41
42
        public void setCurrentData(DFTStraight currentData) {
43
44
            currentAmplitudes = currentData.getAmplitudes();
            currentPhases = currentData.getPhases();
45
46
47
            currentIsSet = true;
48
       }
49
50
51
        * Stretch data in N times
52
        * @param coefficient in how many times to stretch
53
54
        * @return stretched data
55
56
        public Complex[] stretch(int coefficient) throws ToneModulationException {
57
            if (!previousIsSet || !currentIsSet)
                 throw new ToneModulation Exception ("Previous_and_current_data_must_be_set");
58
59
60
            int newSize = length * coefficient;
61
            Complex [] result = new Complex [newSize];
62
            double [] newPhases = new double [newSize];
            double[] newAmplitudes = new double[newSize];
double[] velocity = new double[length];
63
64
65
66
            for (int i = 0; i < length; i++)
67
                 velocity[i] = currentPhases[i] - previousPhases[i];
68
69
            for (int i = 0; i < length; i++)
                 \hat{for} (int j = 0; j < coefficient; j++)
70
71
                     newPhases[i + j] = currentPhases[i] + velocity[i];
```

```
72
  73
                                                                             result = DFTStraight.applyNewPhases(newPhases, result);
 74
  75
                                                                             \label{eq:formula} \mbox{for } (\mbox{int} \ i \ = \ 0\,; \ i \ < \mbox{length}\,; \ i++)
                                                                                                         \dot{for} (int j = 0; j < coefficient; j++)
  76
                                                                                                                                     [i] = [i] 
 77

→ coefficient , currentAmplitudes[i], j);
  78
  79
                                                                             result = DFTStraight.applyNewAmplitudes(newAmplitudes, result);
  80
  81
                                                                            return result;
  82
                                                }
  83
```

Листинг 13: ToneModulationException.java

```
package com.sunradio.math;
1
2
3
4
   * Exception for ToneModulation class
5
6
     @author V. Kremneva
7
8
  public class ToneModulationException extends Exception {
9
10
       public ToneModulationException(String message) {
11
           super (message);
12
13
```

Листинг 14: SunRadioTest.groovy

```
package com.sunradio.core
 3
  import com.external.WavFile
 4
   import com. sunradio. math. AM
  import com. sunradio. math. DFTInverse
 6
  import com. sunradio. math. DFTStraight
 7
   import com. sunradio. math. Filter
q
   class SunRadioTest extends GroovyTestCase {
10
       final FRAMES = 100
       final EPS = 0.00001
11
12
       DFTStraight transformable = new DFTStraight()
13
       int numChannels, indAmount, framesRead
14
15
       double [] buffer, lightLevel, modulated, amplitudes
16
17
18
       void testRun() {
19
           try {
                 inputPath = "C:\\ Users\\LEV\\ IdeaProjects\\SunRadio\\launch.wav";
^{20}
21
               // outputPath = "C:\\Users\\LEV\\IdeaProjects\\SunRadio\\new1.wav";
22
23
           } catch (Exception e) {
24
               System.err.println(e.toString())
25
26
       }
27
^{28}
       void testMove() {
^{29}
           int offset = 3
30
           double[] before = [1, 2, 3, 4, 5, 6, 7]
31
           double [] afterExpected = [4, 5, 6, 7, 0, 0, 0]
32
           double[] after Actual
33
34
           afterActual = SunRadio.move(before, offset)
35
36
           for (int i = 0; i < before.length; i++)
37
                assert Equals (after Expected [i], after Actual [i])
38
       }
39
```

Листинг 15: DFTStraightTest.groovy

```
package com.sunradio.math
 2
 3
    import static java.lang.Math.*
 4
    class DFTStraightTest extends GroovyTestCase {
 5
 6
         \mathrm{final}\ \mathrm{EPS}\ =\ 0.00001
 7
 8
         final\ ITERATIONS = 100
 9
         final SPLIT = 100
10
         final NUMBER = 80.0
11
         \begin{array}{lll} \mbox{double} \ [] & \mbox{buffer} = \mbox{new double} \ [\mbox{ITERATIONS}] \\ \mbox{double} \ [] & \mbox{result} = \mbox{new double} \ [\mbox{ITERATIONS}] \end{array}
12
13
         DFTStraight dftStraight = new DFTStraight()
14
15
16
         //f(t) = sin(t)
17
         void testSin() {
18
19
               //Split the sinus period in SPLIT pieces and take the sinus value in each of them
^{20}
               for (int i = 0; i < ITERATIONS; i++)
21
                    buffer[i] = sin(2 * PI * i / SPLIT)
22
23
               dftStraight.run(buffer)
24
               result = dftStraight.getAmplitudes()
^{25}
               int \ amount \ = \ 0 \ , \ size \ = \ dft \, Straight \ . \, get \, Size \, (\,)
26
27
               for (int i = 0; i < size; i++)
28
                    if \hspace{0.1cm} (\hspace{0.1cm} \mathtt{result}\hspace{0.1cm} [\hspace{0.1cm} \mathtt{i}\hspace{0.1cm}] \hspace{0.1cm} > \hspace{0.1cm} \mathtt{EPS}\hspace{0.1cm})
29
                         amount++
30
31
               assert Equals (1, amount)
32
33
         //f(t) = NUMBER*sin(t)
34
35
         //Test passes with any number
36
         void testConstMultSin() {
37
               //Split the sinus period in SPLIT pieces and take the sinus value in each of them
38
39
               for (int i = 0; i < ITERATIONS; i++)
                    buffer[i] = NUMBER * sin(2 * PI * i / SPLIT)
40
41
               dftStraight.run(buffer)
42
43
               result = dftStraight.getAmplitudes()
44
               int amount = 0, size = dftStraight.getSize()
45
               for (int i = 0; i < size; i++)
46
                    if \hspace{0.1cm} (\hspace{0.1cm} \texttt{result}\hspace{0.1cm} [\hspace{0.1cm} \texttt{i}\hspace{0.1cm}] \hspace{0.1cm} > \hspace{0.1cm} \texttt{EPS}\hspace{0.1cm})
47
48
                         amount++
49
50
               assert Equals (1, amount)
51
52
         //f(t) = sin(NUMBER*t)
53
54
         void testSinMultConst() throws IllegalArgumentException {
55
56
               i\,f\ ((\text{NUMBER} > \text{SPLIT})\ |\ |\ (\text{NUMBER} < \text{SPLIT}\ /\ 2))
                    throw new Illegal Argument Exception ("Number_should_be_less_than_SPLIT_and_more_than
57

→ 
¬SPLIT/2¬due¬to¬sinus¬period")
58
               //Split the sinus period in SPLIT pieces and take the sinus value in each of them
59
               for (int i = 0; i < ITERATIONS; i++)
60
                    buffer[i] = sin(NUMBER * 2 * PI * i / SPLIT)
61
62
               dftStraight.run(buffer)
63
               result = dftStraight.getAmplitudes()
64
65
66
               int amount = 0, size = dftStraight.getSize()
              for (int i = 0; i < size; i++)
67
68
                    if (result[i] > EPS)
69
                         amount++
70
71
               assert Equals (1, amount)
72
         }
73
74
         //f(t) = NUMBER*sin(t) + sin(Number*t)
75
         void testTwoSinuses() {
```

```
76
 77
             if ((NUMBER > SPLIT) || (NUMBER < SPLIT / 2))
                  throw new Illegal Argument Exception ("Number_should_be_less_than_SPLIT_and_more_than
 78

→ 
¬SPLIT/2¬due¬to¬sinus¬period")
 79
              ^{\prime}/\operatorname{Split} the sinus period {f in} SPLIT pieces and take the sinus value {f in} each of them
 80
             for (int i = 0; i < ITERATIONS; i++)
 81
                  buffer[i] = NUMBER*sin(2 * PI * i / SPLIT) + sin(NUMBER * 2 * PI * i / SPLIT)
 82
 83
 84
             dftStraight.run(buffer)
             result = dftStraight.getAmplitudes()
 85
 86
 87
             int amount = 0, size = dftStraight.getSize()
 88
             \mathbf{for} \quad (int \quad i = 0; \quad i < size; \quad i++)
                  if (result[i] > EPS)
 89
 90
                      amount++
 91
 92
             assert Equals (2, amount)
        }
 93
 94
        void test Apply New Amplitudes () {
 95
 96
             double [] modulation
 97
             DFTStraight dftStraightApplied = new DFTStraight()
 98
 99
                 (int i = 0; i < ITERATIONS; i++)
100
                  buffer[i] = sin(2 * PI * i / SPLIT)
101
             dftStraight.run(buffer)
102
103
104
             modulation = new double [dftStraight.size]
105
             Random random = new Random()
             \label{eq:formula} \mbox{for (int $i=0$; $i<$dftStraight.size; $i++$)}
106
107
                  modulation [i] = abs (random.nextDouble())
108
109
             dftStraightApplied.setData(dftStraight.getData())
110
             dftStraightApplied.applyNewAmplitudes(modulation)
111
112
             \verb"double" old Phase", new Phase"
             for (int i = 0; i < dftStraight.size; i++) {
113
114
                 oldPhase = dftStraight.getPhase(i)
115
                 newPhase = dftStraightApplied.getPhase(i)
116
117
                 assert abs(oldPhase - newPhase) < EPS
118
             }
119
        }
120
        void testApplyNewPhases() {
121
             double[] modulation
122
123
             DFTStraight \ dftStraightApplied = new \ DFTStraight()
124
             for (int i = 0; i < ITERATIONS; i++)
125
126
                  buffer[i] = sin(2 * PI * i / SPLIT)
127
             dftStraight.run(buffer)
128
129
             modulation = new double [dftStraight.size]
130
13\,1
             Random random = new Random()
132
             for (int i = 0; i < dftStraight.size; i++)
133
                  modulation[i] = abs(random.nextDouble())
134
             dftStraightApplied.setData(dftStraight.getData())
135
136
             dftStraightApplied.applyNewPhases(modulation)
137
138
             \ double\ old Amplitude\ ,\ new Amplitude
139
             for (int i = 0; i < dftStraight.size; i++) {
140
                  old Amplitude = dft Straight.get Amplitude(i)
                 new\,Amplitude \,=\, dft\,St\,raig\,ht\,A\,p\,p\,lied\,.\,get\,A\,mp\,litud\,e\,(\,i\,)
141
142
143
                  assert abs(oldAmplitude - newAmplitude) < EPS
144
            }
145
146
```

Листинг 16: DFTInverseTest.groovy

```
3
       import static java.lang.Math.PI
  4
       import static java.lang.Math.sin
  6
        class DFTInverseTest extends GroovyTestCase {
                   \mathtt{final} \ \mathrm{EPS} \ = \ 0.00001
  7
                   final\ ITERATIONS = 100
  8
                   final\ SPLIT\ =\ 100
  g
10
                   final NUMBER = 80.0
11
                   double[] income = new double[ITERATIONS]
12
13
                   DFTStraight outcome straight = new DFTStraight()
                   \tt double\,[\,] \quad outcome\_inverse \, = \, new \ double\,[\,ITERATIONS\,]
14
15
16
                   // f(t) = \sin(t)
                   public void testSin() {
17
                              //Split the sinus period in SPLIT pieces and take the sinus value in each of them
18
19
                              for (int i = 0; i < ITERATIONS; i++)
                                         income[i] = sin(2 * PI * i / SPLIT)
20
21
22
                             \verb"outcome_straight.run" (\verb"income")"
                              outcome_inverse = DFTInverse.run(outcome_straight.getData())
23
24
25
                              double difference;
26
                              int amount = 0;
                              for (int i = 0; i < ITERATIONS; i++) {
27
28
                                         difference = outcome_inverse[i] - income[i]
29
                                         if (difference > EPS) amount++
30
31
32
                              assert Equals (0, amount)
33
34
                  \label{eq:local_local_local} //\,f(\,t\,) \, = \, NUMBER*\,sin\,(\,t\,) \, + \, sin\,(\,Number*t\,) \\ public \ void \ test\,TwoSin\,(\,) \ throws \ IllegalArgumentException \ \{
35
36
37
                              if ((NUMBER > SPLIT) |  (NUMBER < SPLIT / 2))
38
                                         throw\ new\ Illegal Argument Exception ("Number\_should\_be\_less\_than\_SPLIT\_and\_more\_than Linear Control of the control of the

→ 
¬SPLIT/2 ¬due ¬to ¬sinus ¬period")
39
                              //\mathrm{Split} the sinus period in SPLIT pieces and take the sinus value in each of them
40
41
                              \label{eq:formula} \textbf{for} \quad (\ \text{int} \quad i \ = \ 0 \, ; \quad i \ < \ \text{ITERATIONS} \, ; \quad i + +)
42
                                         income[i] = NUMBER*sin(2 * PI * i / SPLIT) + sin(NUMBER * 2 * PI * i / SPLIT)
43
44
                              outcome straight.run(income)
                              outcome_inverse = DFTInverse.run(outcome_straight.getData())
45
46
47
                              double difference;
48
                             int amount = 0;
                              \label{eq:for_state} \textbf{for} \hspace{0.2cm} (\hspace{0.1cm} \text{int} \hspace{0.2cm} i \hspace{0.1cm} = \hspace{0.1cm} 0 \hspace{0.1cm} ; \hspace{0.2cm} i \hspace{0.1cm} < \hspace{0.1cm} \text{ITERATIONS} \hspace{0.1cm} ; \hspace{0.2cm} i \hspace{0.1cm} + \hspace{0.1cm} ) \hspace{0.1cm} \hspace{0.1cm} \{
^{49}
                                         difference = outcome_inverse[i] - income[i]
if (difference > EPS) amount++
50
51
52
53
                              assert Equals (0, amount)
54
55
                  }
56
```

Листинг 17: FilterTest.groovy

```
package com. sun radio. math
 2
 3
   import static java.lang.Math.PI
 4
   import static java.lang.Math.sin
 5
 6
   class FilterTest extends GroovyTestCase {
        final\ ITERATIONS = 100
 7
 8
        final SPLIT = 80
 9
        final EPS = 0.00001
10
1\,1
        double [] buffer = new double [ITERATIONS]
        double [] winFunc = new double [ITERATIONS] double [] applied = new double [ITERATIONS]
12
13
14
15
        void testApply() {
16
17
             //Split the sinus period in SPLIT pieces and take the sinus value in each of them
```

```
18
             for (int i = 0; i < ITERATIONS; i++)
19
                  buffer[i] = sin(2 * PI * i / SPLIT)
20
21
             winFunc = Filter.BlackmanNuttall(ITERATIONS)
^{22}
             applied = Filter.apply(buffer, winFunc)
^{23}
24
             for (int i = 0; i < ITERATIONS; i++)
25
                  assert \ applied \left[ \ i \ \right] \ - \ \left( \ buffer \left[ \ i \ \right] \ * \ winFunc \left[ \ i \ \right] \right) \ < \ EPS
26
27
28
        void testGetOutputFilter() {
^{29}
             double [] inputWindowFunc = Filter.BlackmanNuttall(ITERATIONS)
             double [] outputWindowFunc = Filter.getOutputWindowFunc(inputWindowFunc)
30
31
             double sum = 0.0
32
33
             for (int i = 0; i < ITERATIONS; i++)
34
                 sum += inputWindowFunc[i]*outputWindowFunc[i]
35
36
             assertEquals(1.0, sum)
37
        }
38
```

Листинг 18: ScaleTest.groovy

```
package com.sunradio.math
 1
2
 3
    class ScaleTest extends GroovyTestCase {
          void testRun() {
 4
 5
                final EPS = 0.00000001
                def arr = [5, 8, 7, 2] as Integer [] def from = 0.0
 6
 7
 8
                d\,ef\ t\,o\ =\ 1.0
9
10
                def \ outputExpected = [0.5, 1.0, 5 / 6, 0.0] \ as \ double[]
11
                \begin{array}{lll} double\,[\,] & output\,Value\,=\,S\,cale\,.\,run\,(\,a\,r\,\,,\,\,from\,\,,\,\,to\,) \\ \mbox{for} & (\,int\ i\,=\,0\,;\,\,i\,<\,4\,;\,\,i++) \end{array} \label{eq:control_eq}
12
13
                       assertTrue((outputExpected[i] - outputValue[i]) < EPS)
14
15
          }
16
```