Chapter 11
Sampled Auditory Content

The Design and Implementation of Multimedia Software

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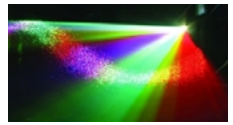
Jones and Bartlett Publishers

www.jbpub.com

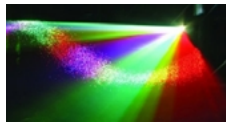


Temporal Sampling

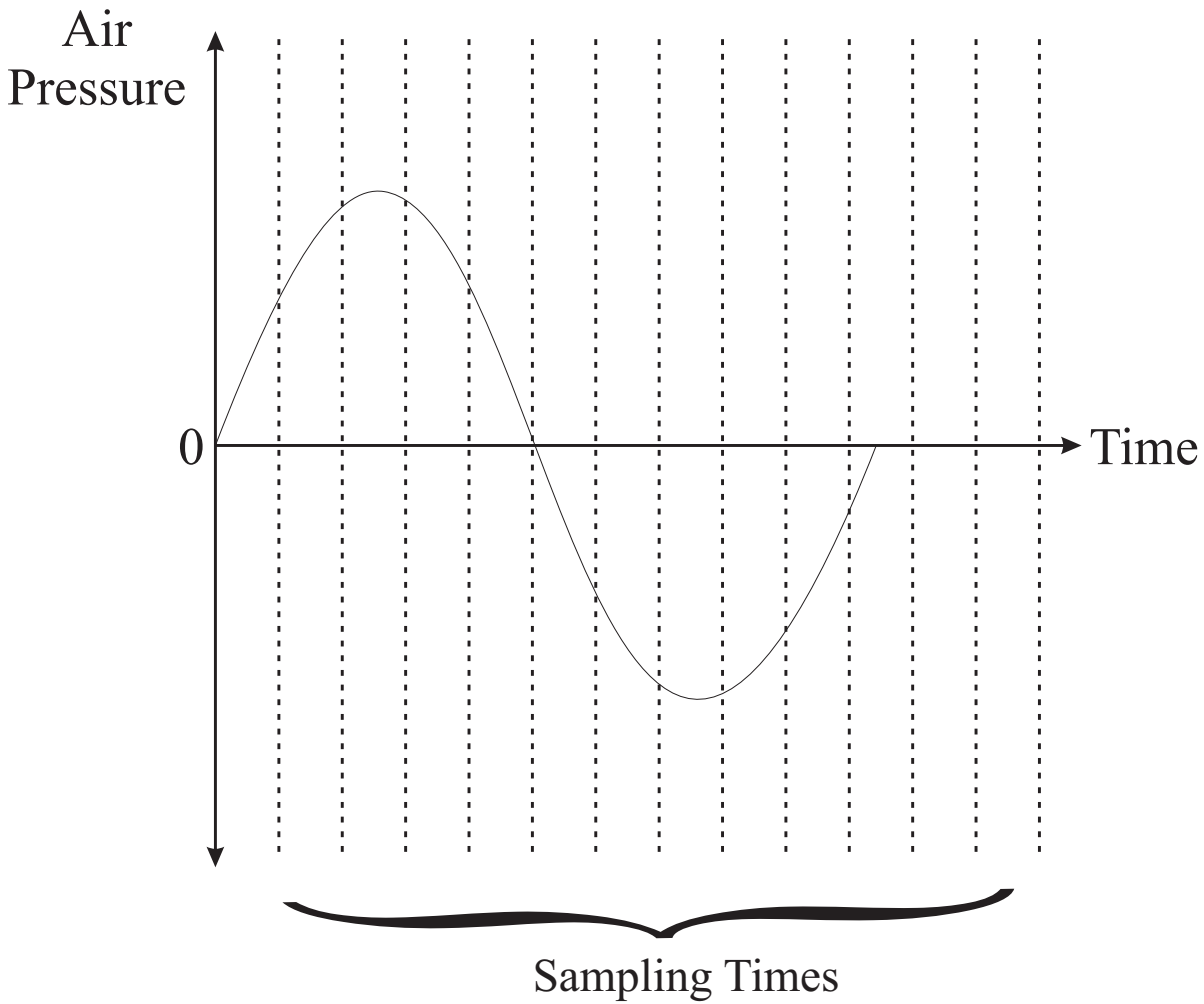
Definition

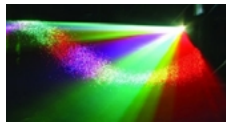
Temporal sampling involves measuring the wave at (usually regular) discrete points in time.

- CDs normally use a 44.1kHz sampling rate (i.e., contain 44,100 samples per second).
- DVD audio normally uses a 96kHz sampling rate (i.e., the the audio track contains 96,000 samples per second).



Temporal Sampling (cont.)



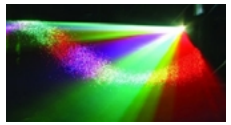


Quantization

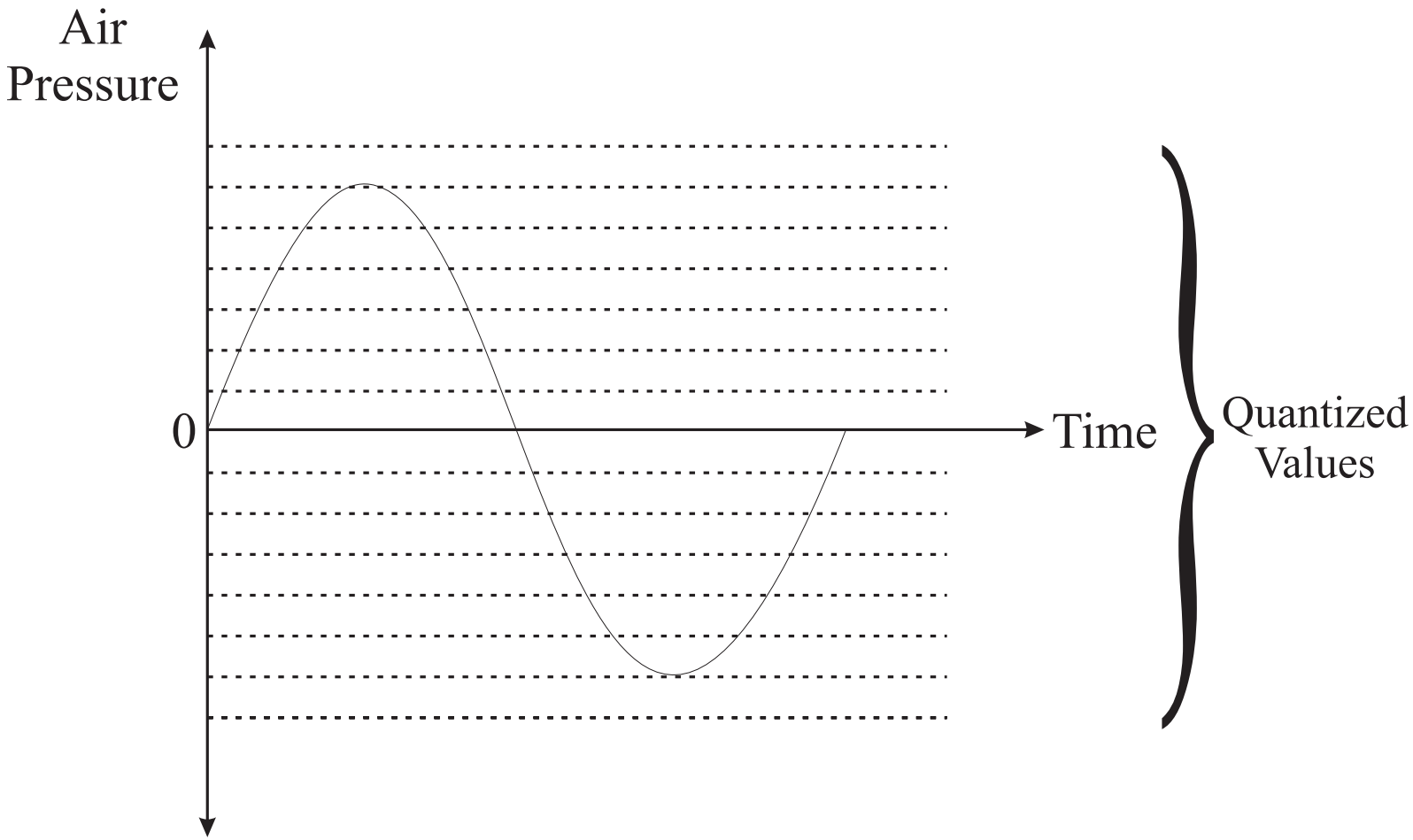
Definition

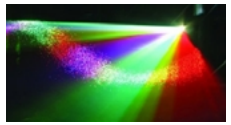
Quantization involves limiting the measured amplitudes to a discrete set of values.

For example, if 8 bits quantization is used there are 256 different amplitudes and the actual amplitude is rounded or truncated to one of these 256 values.



Quantization (cont.)

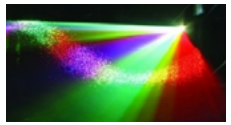




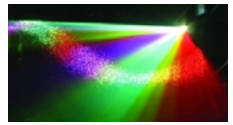
The `AudioFormat` Class

An `AudioFormat` object has, among others, the following attributes:

- The number of channels (e.g., mono, stereo).
- The sampling rate.
- The quantization (i.e., the number of bits per sample).
- The encoding technique (e.g., linear pulse code modulation, nonlinear mu-law).



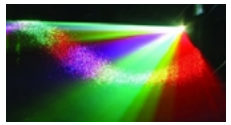
Presentation of Sampled Audio in Java



The `Clip` Class

A `Clip` object:

- Is a type of `Line` that contains data that can be loaded prior to presentation.
- Renders its sampled auditory content when its `start()` method is called.



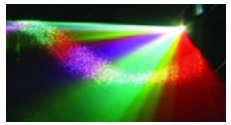
Creating a Clip Object

```
// Get the resource
finder = ResourceFinder.createInstance();
is     = finder.findInputStream("/"+args[0]);

// Create an AudioInputStream from the InputStream
stream = AudioSystem.getAudioInputStream(is);
```

```
// Create a Clip (i.e., a Line that can be pre-load
clip = AudioSystem.getClip();
```

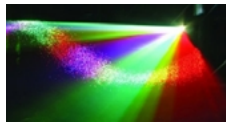
```
// Tell the Clip to acquire any required system
// resources and become operational
clip.open(stream);
```



Using a `Clip` Object

```
// Present the Clip (without blocking the
// thread of execution)
clip.start();
```

Note that `start()` method does not block the thread of execution.

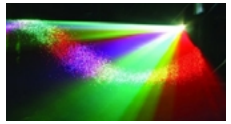


Requirements

F11.1 Encapsulate signals.

F11.2 Operate on signals.

F11.3 Present/render these signals.



An Overview

- Any Encapsulation Must Include:

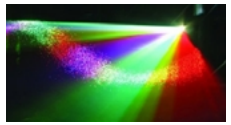
  The sample points for all of the signals (i.e., one signal for monophonic, two signals for stereophonic, etc).

  Information about the sampling process.

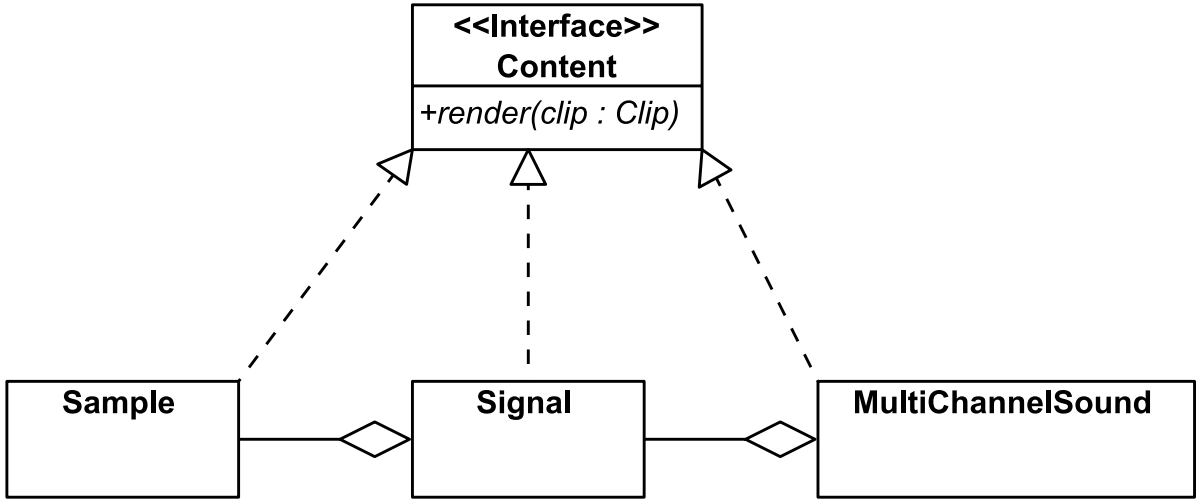
- Some Observations:

  Information about the sampling process can be stored in an `AudioFormat` object.

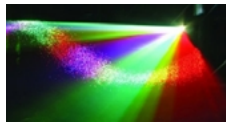
  All that remains is to consider ways to encapsulate samples and signals.



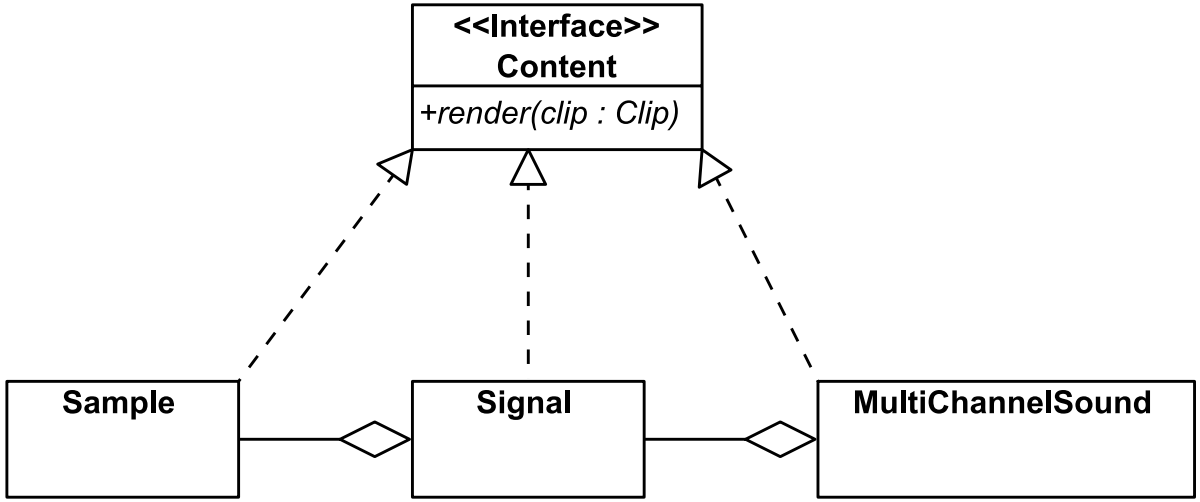
Alternative 1



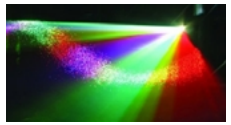
What are the shortcomings?



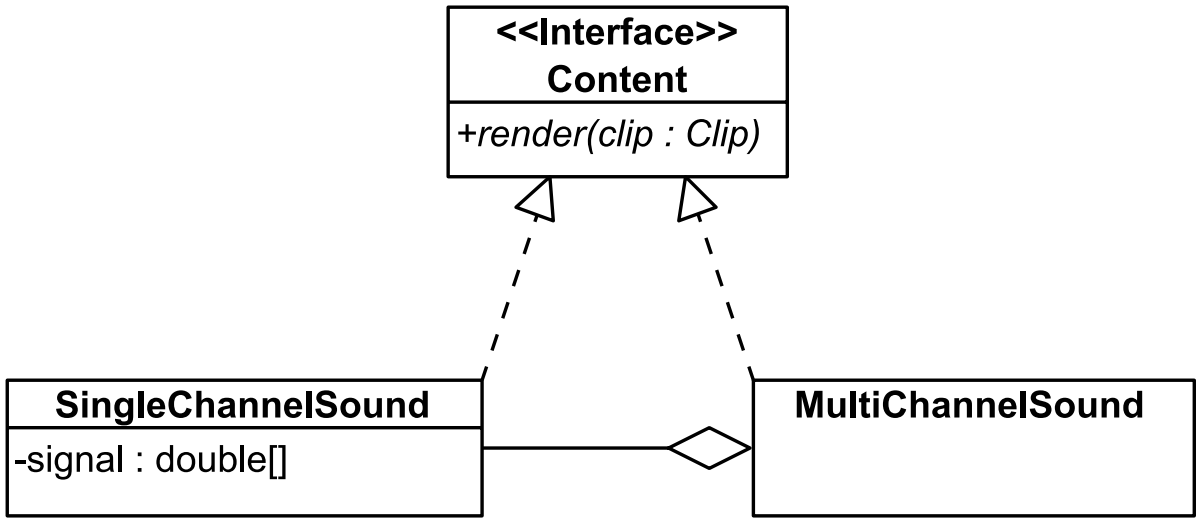
Alternative 1



Since a sample is nothing but a numeric value, there is no reason to have a `Sample` class.



Alternative 2



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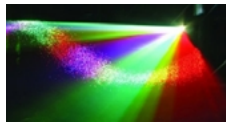
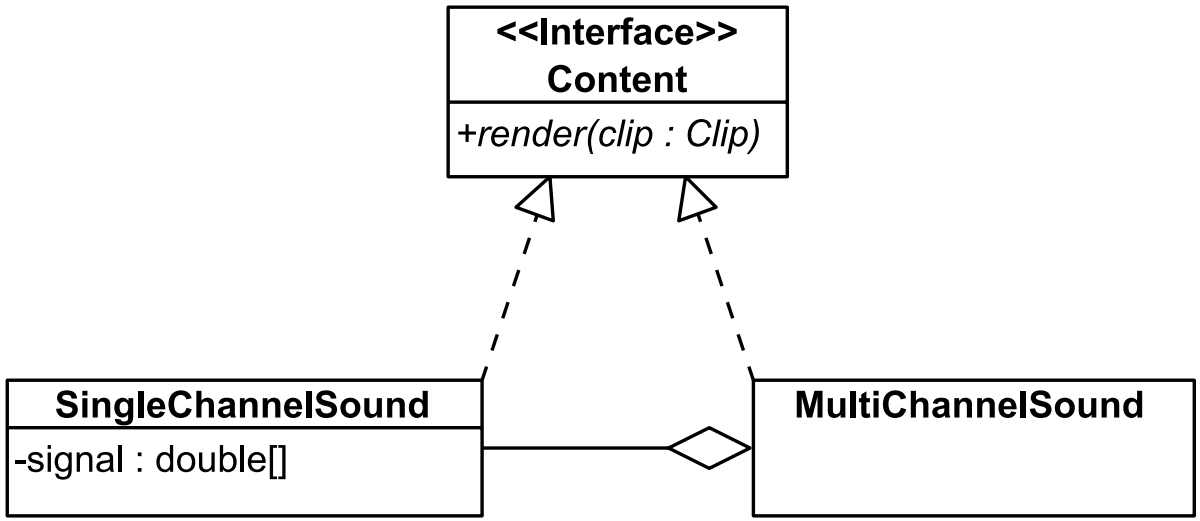
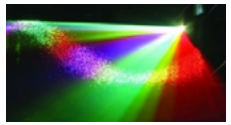


Alternative 2

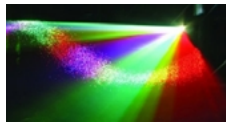


Many existing file formats make it difficult to independently/sequentially construct `SingleChannelSound` objects and then combine them into a `MultiChannelSound` object.



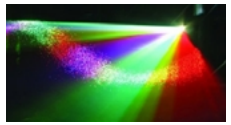
Alternative 3

BufferedSound
-signal : List<double[]>
+render(clip : Clip)



BufferedSound – Structure

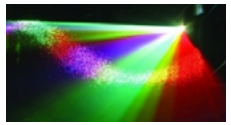
```
package auditory.sampled;

import java.util.*;
import javax.sound.sampled.*;

public class BufferedSound implements Content
{
    private ArrayList<double[]>  channels;
    private AudioFormat          format;
    private int                  numberOfSamples;

    private static final double MAX_AMPLITUDE       =  32767.0;
    private static final double MIN_AMPLITUDE       = -32767.0;
    private static final int    SAMPLE_SIZE_IN_BITS = 16;
    private static final int    BYTES_PER_CHANNEL   = SAMPLE_SIZE_IN_BITS/8;
}
```



BufferedSound – Constructor

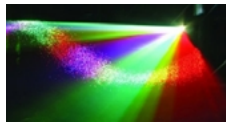
To simplify the discussion that follows, this class uses sampling processes that vary only in their sampling rates; all other aspects of the process are standardized. This is evident in the explicit value constructor of the class.

```
public BufferedSound(float sampleRate)
{
   format = new AudioFormat(
      AudioFormat.Encoding.PCM_SIGNED,
      sampleRate,          // Sample rate in Hz
      SAMPLE_SIZE_IN_BITS, // Sample size in bits
      0,                   // Number of channels
      0,                   // Frame size in bytes
      sampleRate,          // Frame rate in Hz
      true);               // Big-endian or not

   channels = new ArrayList<double[]>();
   numberOfSamples = 0;
}
```



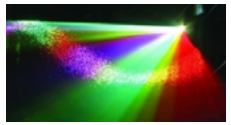
BufferedSound – addChannel()

```
public synchronized void addChannel(double[] signal)
{
   if (numberOfSamples == 0) numberOfSamples = signal.length;

   if (numberOfSamples == signal.length)
   {
      channels.add(signal);
      updateAudioFormat();
   }
}
```



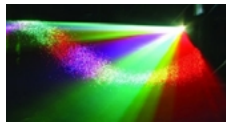
BufferedSound– updateAudioFormat()

```
private void updateAudioFormat()
{
   format = new AudioFormat(
      format.getEncoding(),                  // Encoding
      format.getSampleRate(),                // Sample rate in Hz
      format.getSampleSizeInBits(),          // Sample size in bits
      channels.size(),                       // Number of channels
      channels.size()*BYTES_PER_CHANNEL,     // Frame size in bytes
      format.getSampleRate(),                // Frame rate in Hz
      format.isBigEndian());                 // Big-endian or not
}
```



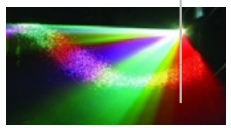
BufferedSound – matches()

```
public synchronized boolean matches(BufferedSound other)
{
   boolean      result;

   result = false;
   result = getAudioFormat().matches(other.getAudioFormat()) &&
            (getNumberOfSamples() == other.getNumberOfSamples());


   return result;
}
```



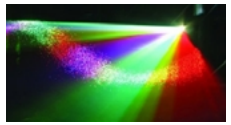
BufferedSound – append()

```
public synchronized void append(BufferedSound other)
{
   ArrayList<double[]>  temp;
   double[]             otherSignal, tempSignal, thisSignal;
   Iterator<double[]>   i, j;

   if (matches(other))
   {
      temp = new ArrayList<double[]>();

      i = channels.iterator();
      j = other.channels.iterator();
      while (i.hasNext())
      {
         thisSignal  = i.next();
         otherSignal = j.next();

         // Allocate space for the new signal
         tempSignal = new double[thisSignal.length +
                                 otherSignal.length];

         // Copy the current signal
         System.arraycopy(thisSignal, 0,
                          tempSignal, 0, thisSignal.length);

         // Append the other left signal
         System.arraycopy(otherSignal, 0,
                          tempSignal, thisSignal.length,
```



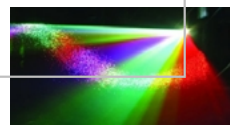
BufferedSound – append() (cont.)

```
                                otherSignal.length);

            // Save the longer signal
            temp.add(tempSignal);
        }
        channels = temp;
    }
}
```

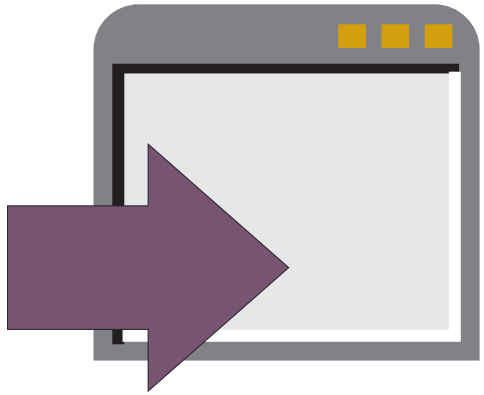


BufferedSoundFactory – Pure Tones

```
public BufferedSound createBufferedSound(double frequency,
                                         int    length,
                                         float  sampleRate,
                                         double amplitude)
{
   BufferedSound     sound;
   double            radians,radiansPerSample, rmsValue;
   double[]          signal;
   int n;

   //samples =      samples/sec * sec
   n         = (int)(sampleRate * (double)length/1000000.0);

   signal    = new double[n];
   //  rads/sample  = ( rads/cycle * cycles/sec)/ samples/sec
   radiansPerSample = (Math.PI*2.0 * frequency) / sampleRate;
   for (int i=0; i<signal.length; i++)
   {
      // rad  =   rad/sample     * sample
      radians = radiansPerSample * i;

      signal[i] = amplitude * Math.sin(radians);
   }
   sound = new BufferedSound(sampleRate);
   sound.addChannel(signal);
   return sound;
}
```



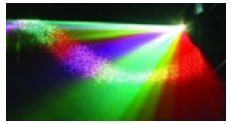
Pure Tones – Demonstration



In `extras`:

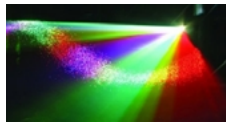
PureTone.html

```
java -cp BufferedSound.jar BufferedSoundApplication NONE 200
```



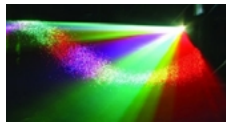
BufferedSoundFactory – Using an AudioInputStream

```
public BufferedSound createBufferedSound(AudioInputStream inStream)
                              throws IOException,
                                     UnsupportedAudioFileException
{
   AudioFormat         inFormat, pcmFormat;
   AudioInputStream    pcmStream;
   BufferedSound       sound;
   byte[]              rawBytes;
   double[]            leftSignal, monoSignal, rightSignal;
   int                 bufferSize, offset, n, sampleLength;
   int[]               signal;
}
```



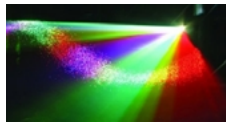
BufferedSoundFactory – Encoding

```
inFormat = inStream.getFormat();

 // Convert ULAW and ALAW to PCM
 if ((inFormat.getEncoding() == AudioFormat.Encoding.ULAW) ||
     (inFormat.getEncoding() == AudioFormat.Encoding.ALAW)   ) {

     pcmFormat = new AudioFormat(
                         AudioFormat.Encoding.PCM_SIGNED,
                         inFormat.getSampleRate(),
                         inFormat.getSampleSizeInBits()*2,
                         inFormat.getChannels(),
                         inFormat.getFrameSize()*2,
                         inFormat.getFrameRate(),
                         true);

     pcmStream = AudioSystem.getAudioInputStream(pcmFormat,
                                                 inStream);
 }
 else // It is PCM
 {
    pcmFormat = inFormat;
    pcmStream = inStream;
 }
```



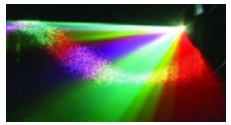
BufferedSoundFactory – Buffer

```
// Create a buffer and read the raw bytes
bufferSize = (int)(pcmStream.getFrameLength())
                   * pcmFormat.getFrameSize();

rawBytes = new byte[bufferSize];
offset   = 0;
n        = 0;
while (pcmStream.available() > 0)
{
   n  = pcmStream.read(rawBytes, offset, bufferSize);
   offset += n;
}
```



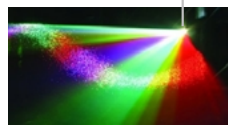
BufferedSoundFactory – Conversion

```
// Convert the raw bytes
if (pcmFormat.getSampleSizeInBits() == 8)
{
   signal = processEightBitQuantization(rawBytes, pcmFormat);
}
else
{
   signal = processSixteenBitQuantization(rawBytes, pcmFormat);
}
```



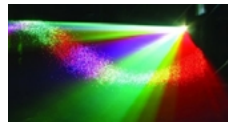
BufferedSoundFactory – Channels

```
sound = new BufferedSound(pcmFormat.getSampleRate());

// Process the individual channels
if (pcmFormat.getChannels() == 1)  // Mono
{
   sampleLength = signal.length;
   monoSignal   = new double[sampleLength];

   for (int i=0; i<sampleLength; i++)
   {
      monoSignal[i]  = signal[i]; // Convert to double
   }
   sound.addChannel(monoSignal);
}
else                                // Stereo
{
   sampleLength = signal.length/2;
   leftSignal   = new double[sampleLength];
   rightSignal  = new double[sampleLength];

   for (int i=0; i<sampleLength; i++)
   {
      leftSignal[i]  = signal[2*i];
      rightSignal[i] = signal[2*i+1];
   }
   sound.addChannel(leftSignal);
```



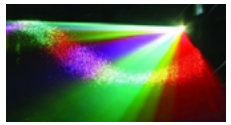
BufferedSoundFactory – Channels (cont.)

```
            sound.addChannel(rightSignal);
        }
```



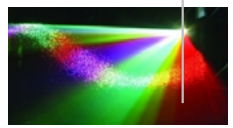
BufferedSoundFactory – 8-bit

```
private int[] processEightBitQuantization(
                                    byte[]      rawBytes,
                                    AudioFormat format)
{
   int          lsb, msb;
   int[]        signal;
   String       encoding;


   signal = new int[rawBytes.length];
   encoding = format.getEncoding().toString();

   if (encoding.startsWith("PCM_SIGN"))
   {
      for (int i=0; i<rawBytes.length; i++)
         signal[i] = rawBytes[i];
   }
   else
   {
      for (int i=0; i<rawBytes.length; i++)
         signal[i] = rawBytes[i]-128;
   }

   return signal;
}
```



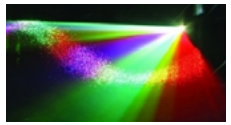
BufferedSoundFactory – 16-bit

```
private int[] processSixteenBitQuantization(
                                        byte[]      rawBytes,
                                        AudioFormat format)
{
   int         lsb, msb;
   int[]       signal;

   signal = new int[rawBytes.length / 2];
   if (format.isBigEndian())  // Big-endian
   {
      for (int i=0; i<signal.length; i++)
      {
         // First byte is high-order byte
         msb = (int) rawBytes[2*i];

         // Second byte is low-order byte
         lsb = (int) rawBytes[2*i+1];

         signal[i] = msb << 8 | (255 & lsb);
      }
   }
   else                          // Little-endian
   {
      for (int i=0; i<signal.length; i++)
      {
         // First byte is low-order byte
         lsb = (int) rawBytes[2*i];
```



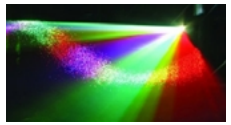
BufferedSoundFactory – 16-bit (cont.)

```
            // Second byte is high-order byte
            msb = (int) rawBytes[2*i+1];

            signal[i] = msb << 8 | (255 & lsb);
        }
    }
    return signal;
}
```

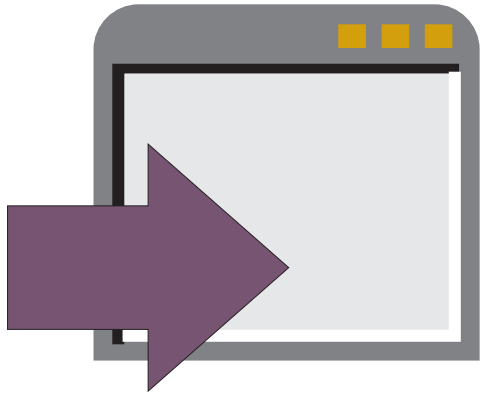


BufferedSoundFactory – Using a File

```
public BufferedSound createBufferedSound(String name)
                              throws IOException,
                                     UnsupportedAudioFileException
{
   AudioInputStream        stream;
   URL                     url;

   url    = finder.findURL(name);
   stream = AudioSystem.getAudioInputStream(url);

   return createBufferedSound(stream);
}
```



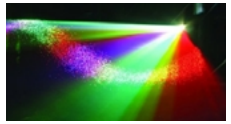
Using a File – Demonstration



In `extras`:

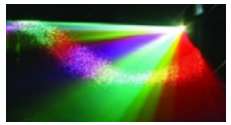
FilePlayer.html

```
java -cp FilePlayer.jar FilePlayerApplication preface.aif
```



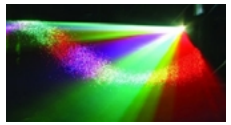
Unary Operations

```
package auditory.sampled;

public interface BufferedSoundUnaryOp
{
    public BufferedSound filter(BufferedSound src,
                                BufferedSound dest);
}
```

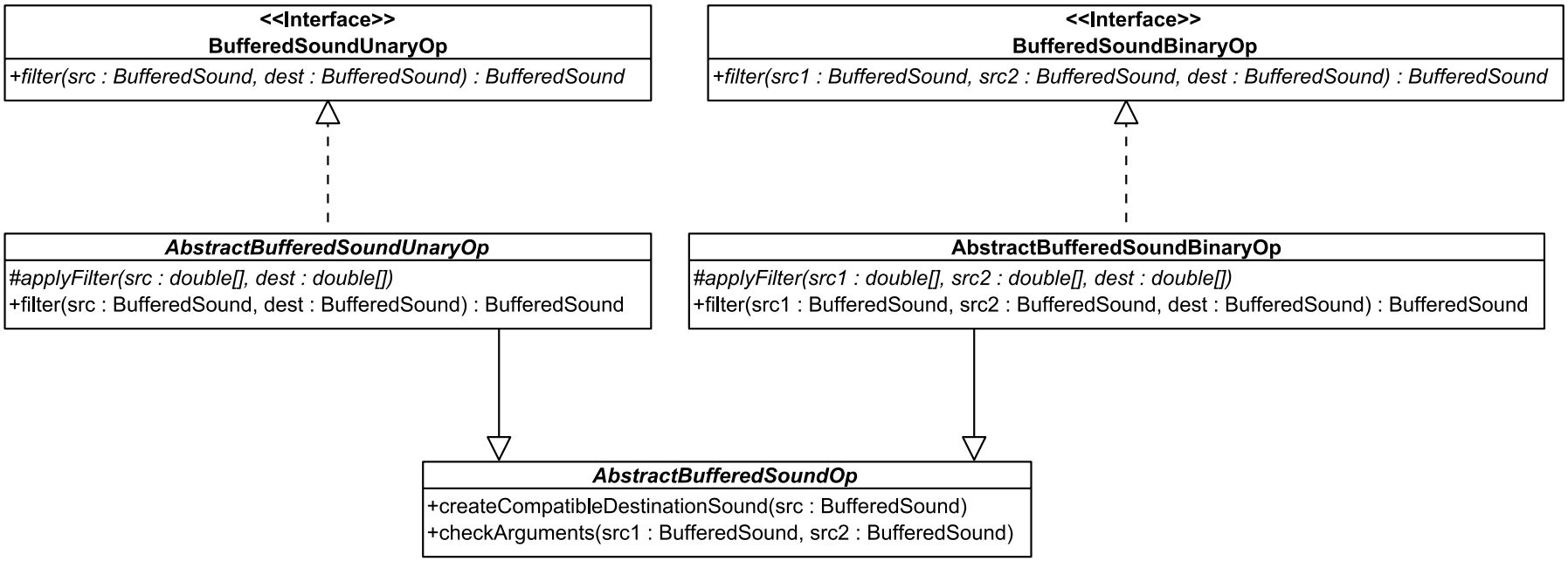


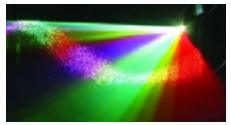
Binary Operations

```
package auditory.sampled;

public interface BufferedSoundBinaryOp
{
    public BufferedSound filter(BufferedSound src1, BufferedSound src2,
                                BufferedSound dest)
                         throws IllegalArgumentException;
}
```



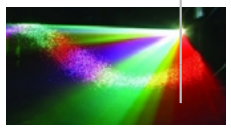
Operating on Sampled Auditory Content





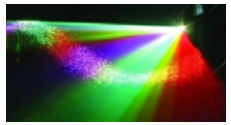
AbstractBufferedSoundOp

```
package auditory.sampled;

public abstract class AbstractBufferedSoundOp
{
    public BufferedSound createCompatibleDestinationSound(
                                          BufferedSound src)
    {
       BufferedSound        temp;
       float                sampleRate;
       int                  channels, length;

       channels   = src.getNumberOfChannels();
       length     = src.getNumberOfSamples();
       sampleRate = src.getSampleRate();

       temp = new BufferedSound(sampleRate);

       for (int i=0; i<channels; i++)
       {
          temp.addChannel(new double[length]);
       }

       return temp;
    }

    protected void checkArguments(BufferedSound a, BufferedSound b)
                                  throws IllegalArgumentException
    {
```



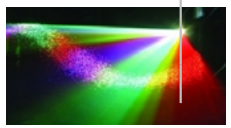
AbstractBufferedSoundOp (cont.)

```
        if (!a.matches(b))
            throw(new IllegalArgumentException("Argument Mismatch"));
    }
}
```



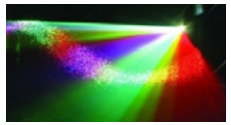
AbstractBufferedSoundUnaryOp

```
package auditory.sampled;

import java.util.*;

public abstract class      AbstractBufferedSoundUnaryOp
                extends    AbstractBufferedSoundOp
                implements BufferedSoundUnaryOp
{
    public abstract void applyFilter(double[] source,
                                     double[] destination);

    public void applyFilter(Iterator<double[]> source,
                            Iterator<double[]> destination)
    {
       while (source.hasNext())
       {
          applyFilter(source.next(), destination.next());
       }
    }

    public BufferedSound filter(BufferedSound src,
                                BufferedSound dest)
    {
       Iterator<double[]>    source, destination;

       // Construct the destination if necessary; otherwise check it
       if (dest == null)
          dest = createCompatibleDestinationSound(src);
```



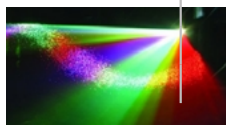
AbstractBufferedSoundUnaryOp (cont.)

```
        // Get the source channels
        source      = src.getSignals();

        // Get the destination channels
        destination = dest.getSignals();

        // Apply the filter
        applyFilter(source, destination);

        return dest;
    }
}
```



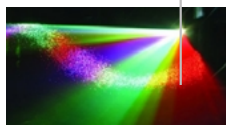
AbstractBufferedSoundBinaryOp

```
package auditory.sampled;

import java.util.*;

public abstract class      AbstractBufferedSoundBinaryOp
                extends    AbstractBufferedSoundOp
                implements BufferedSoundBinaryOp
{
    public abstract void applyFilter(double[] source1,
                                     double[] source2,
                                     double[] destination);

    public void applyFilter(Iterator<double[]> source1,
                            Iterator<double[]> source2,
                            Iterator<double[]> destination)
    {
       while (source1.hasNext())
       {
          applyFilter(source1.next(), source2.next(), destination.next());
       }
    }

    protected void checkArguments(BufferedSound a,
                                  BufferedSound b)
                                  throws IllegalArgumentException
    {
       if (!a.matches(b))
           throw(new IllegalArgumentException("Argument Mismatch"));
```



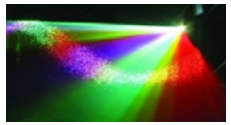
AbstractBufferedSoundBinaryOp (cont.)

```
}

public BufferedSound filter(BufferedSound src1,
                            BufferedSound src2,
                            BufferedSound dest)
                     throws IllegalArgumentException
{
   Iterator<double[]>   source1, source2, destination;


   // Check the properties of the two source sounds
   checkArguments(src1, src2);

   // Construct the destination if necessary; otherwise check it
   if (dest == null)
      dest = createCompatibleDestinationSound(src1);
   else
      checkArguments(src1, dest);


   // Get the source channels
   source1     = src1.getSignals();
   source2     = src2.getSignals();

   // Get the destination channels
   destination = dest.getSignals();
```



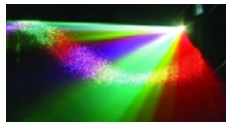
AbstractBufferedSoundBinaryOp (cont.)

```
        // Apply the filter
        applyFilter(source1, source2, destination);

        return dest;
    }
}
```

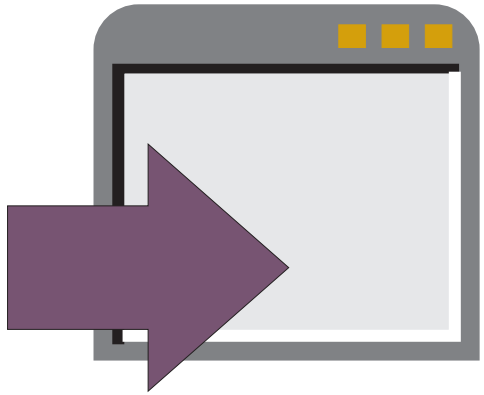


AddOp

```
package auditory.sampled;

public class AddOp extends    AbstractBufferedSoundBinaryOp
{
    public void applyFilter(double[] source1, double[] source2,
                            double[] destination)
    {
       for (int i=0; i<source1.length; i++)
       {
             destination[i] = source1[i] + source2[i];
       }
    }
}
```



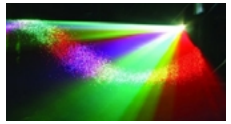
AddOp – Demonstration



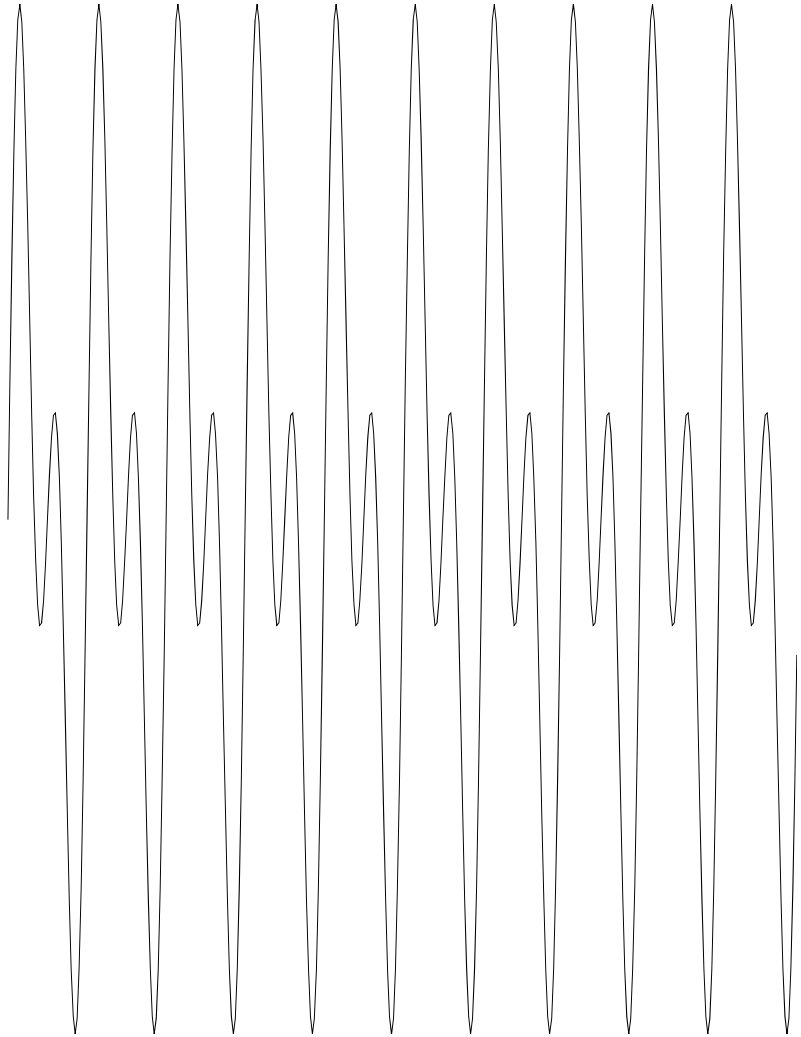
In extras:

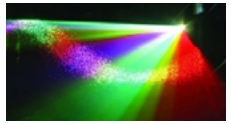
AddOp.html

```
java -cp BufferedSound.jar BufferedSoundApplication BINARY 100 ADD 200
```

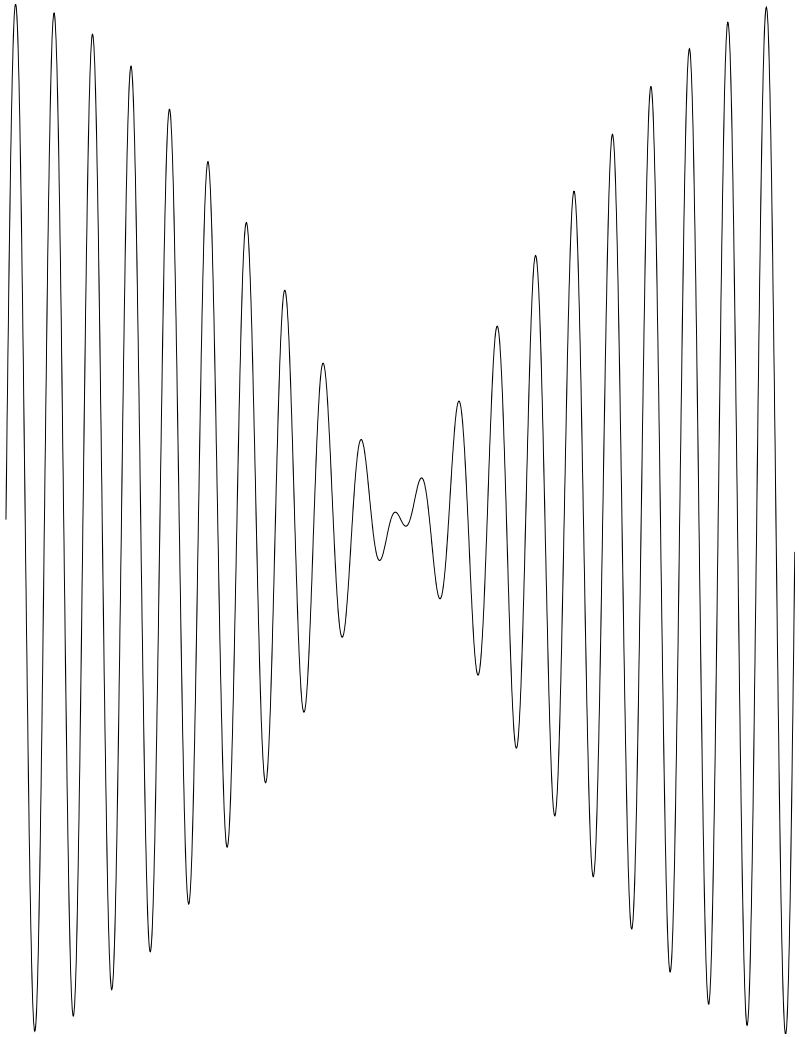


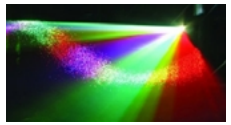
The Result of Adding 100Hz and 200Hz Sine Waves



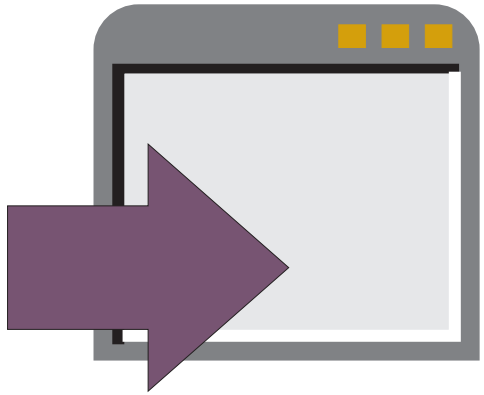


Adding 100Hz and 105Hz Sine Waves – Beating





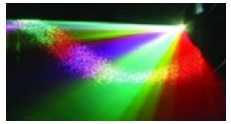
AddOp – Demonstration



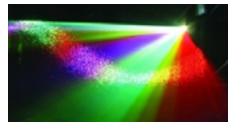
In extras:

Beating.html

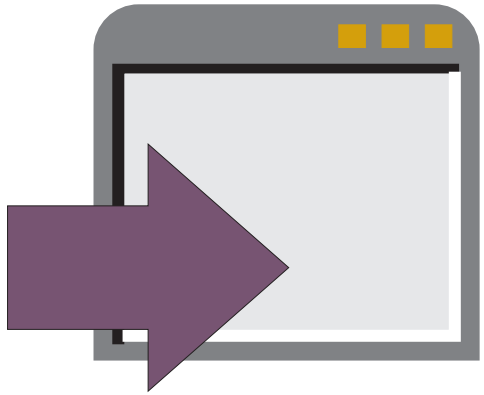
```
java -cp BufferedSound.jar BufferedSoundApplication BINARY 100 ADD 105
```



```
package auditory.sampled;

public class ReverseOp extends    AbstractBufferedSoundUnaryOp
{
    public void applyFilter(double[] source, double[] destination)
    {
       int       length;

       length    = source.length;

       for (int i=0; i<length; i++)
       {
          destination[i]  = source[length-1-i];
       }
    }
}
```



ReverseOp – Demonstration



In extras:

Original

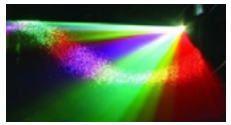
Number9.html

```
java -cp FilePlayer.jar FilePlayerApplication number9.aif
```

Reversed

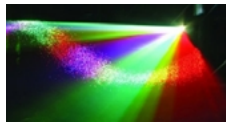
Reverse.html

```
java -cp BufferedSound.jar BufferedSoundApplication UNARY number9.aif REVERSE
```

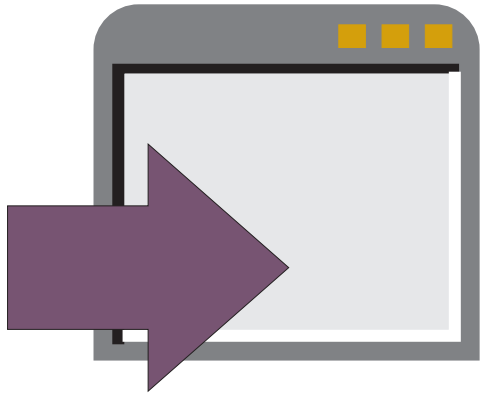


InvertOp

```
package auditory.sampled;

public class InvertOp extends    AbstractBufferedSoundUnaryOp
{
    public void applyFilter(double[] source, double[] destination)
    {
       int       length;

       length    = source.length;

       for (int i=0; i<length; i++)
       {
          destination[i]  = -source[i];
       }
    }
}
```



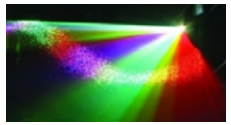
InvertOp – Demonstration



In extras:

InvertOp.html

```
java -cp BufferedSound.jar BufferedSoundApplication UNARY preface.aif INVERT
```



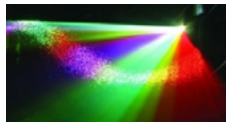
Categorizing Filters

- Causal:

  Use only sample points ‘before’ the current point.

- Non-Causal:

  Can use sample points ‘after’ the current point.



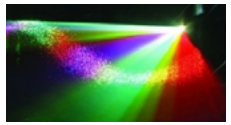
Categorizing Filters (cont.)

- Finite:

  Only use the source.

- Infinite:

  Use both the source and the destination.



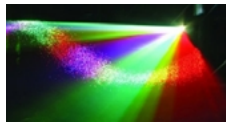
Categorizing Filters (cont.)

- Linear:

  Only combine the sample points using addition and multiplication by a constant.

- Non-Linear:

  Combine sample points in any fashion.



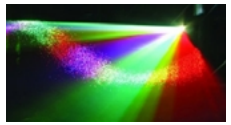
Categorizing Filters (cont.)

- Time-Invariant:

  Do not change over time.

- Adaptive:

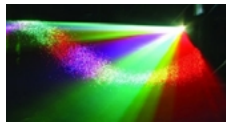
  Change over time.



Categorizing Filters (cont.)

Letting d denote the destination, s denote the source, and w and v denote weights, an *infinite, linear, causal filter* is a filter than can be expressed as follows:

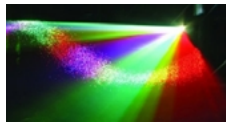
$$d_i = \sum_{k=0}^{n} s_{i-k} w_k + \sum_{j=0}^{m} d_{i-j} v_j \text{ for all } i$$



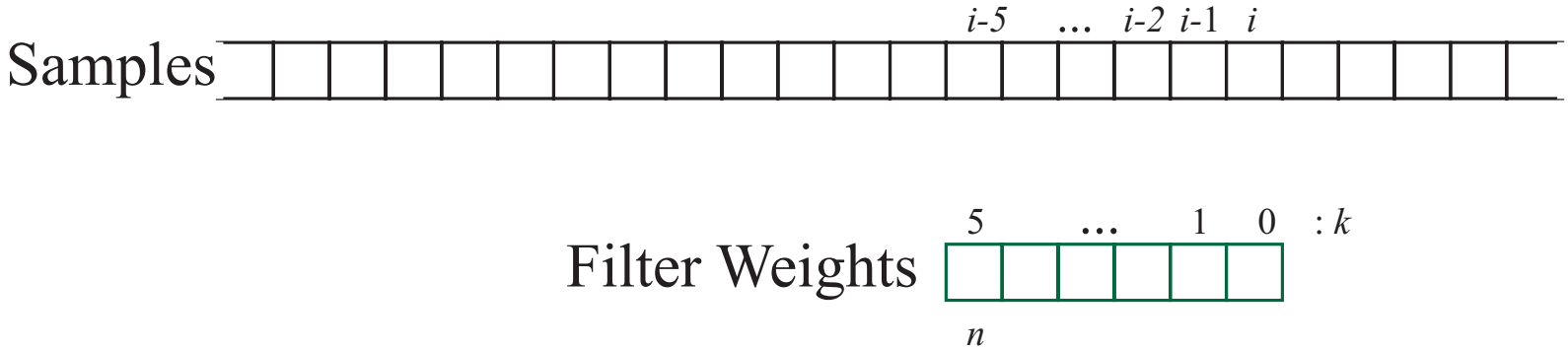
Categorizing Filters (cont.)

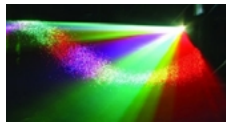
A *finite, linear causal filter* (which is also called a *finite impulse response* or FIR filter) is a filter that can be expressed as follows:

$$d_i = \sum_{k=0}^{n} s_{i-k} w_k \text{ for all } i$$



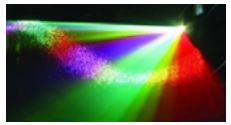
A FIR Filter





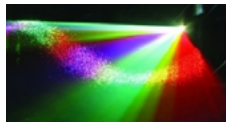
FIRFilter – Structure

```
package auditory.sampled;

public class FIRFilter
{
    private double[]     weights;

    public FIRFilter(double[] weights)
    {
       this.weights = new double[weights.length];
       System.arraycopy(weights, 0, this.weights, 0, weights.length);
    }
}
```



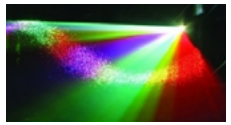
FIRFilter – Getters

```
public int getLength()
{
   int   length;

   length = 0;
   if (weights != null) length = weights.length;

   return length;
}
public double getWeight(int index)
{
   double   weight;

   weight = 0.0;
   if ((weights == null) && (index == weights.length-1))
   {
      weight = 1.0;
   }
   else if ((index >=0) && (index < weights.length-1))
   {
      weight = weights[index];
   }

   return weight;
}
```



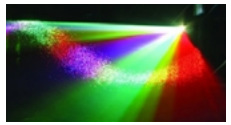
FIRFilterOp – Structure

```
package auditory.sampled;

public class FIRFilterOp extends AbstractBufferedSoundUnaryOp
{
    private FIRFilter          fir;

    public FIRFilterOp(FIRFilter fir)
    {
       this.fir = fir;
    }
}
```

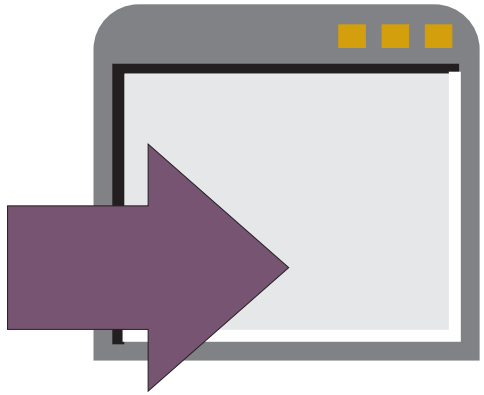


FIRFilterOp – applyFilter()

```
public void applyFilter(double[] source, double[] destination)
{
   double    weight;
   int       length, n;

   n         = fir.getLength();
   length    = source.length;

   // Copy the first n-2 samples
   for (int i=0; i<n-1; i++)
   {
      destination[i]  = source[i];
   }

   // Filter the remaining samples
   for (int i=n-1; i<length; i++)
   {
      for (int k=0; k<n; k++)
      {
         weight        = fir.getWeight(k);

         destination[i]  += source[i-k]  * weight;
      }
   }
}
```



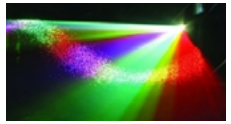
FIRFilterOp – Demonstration



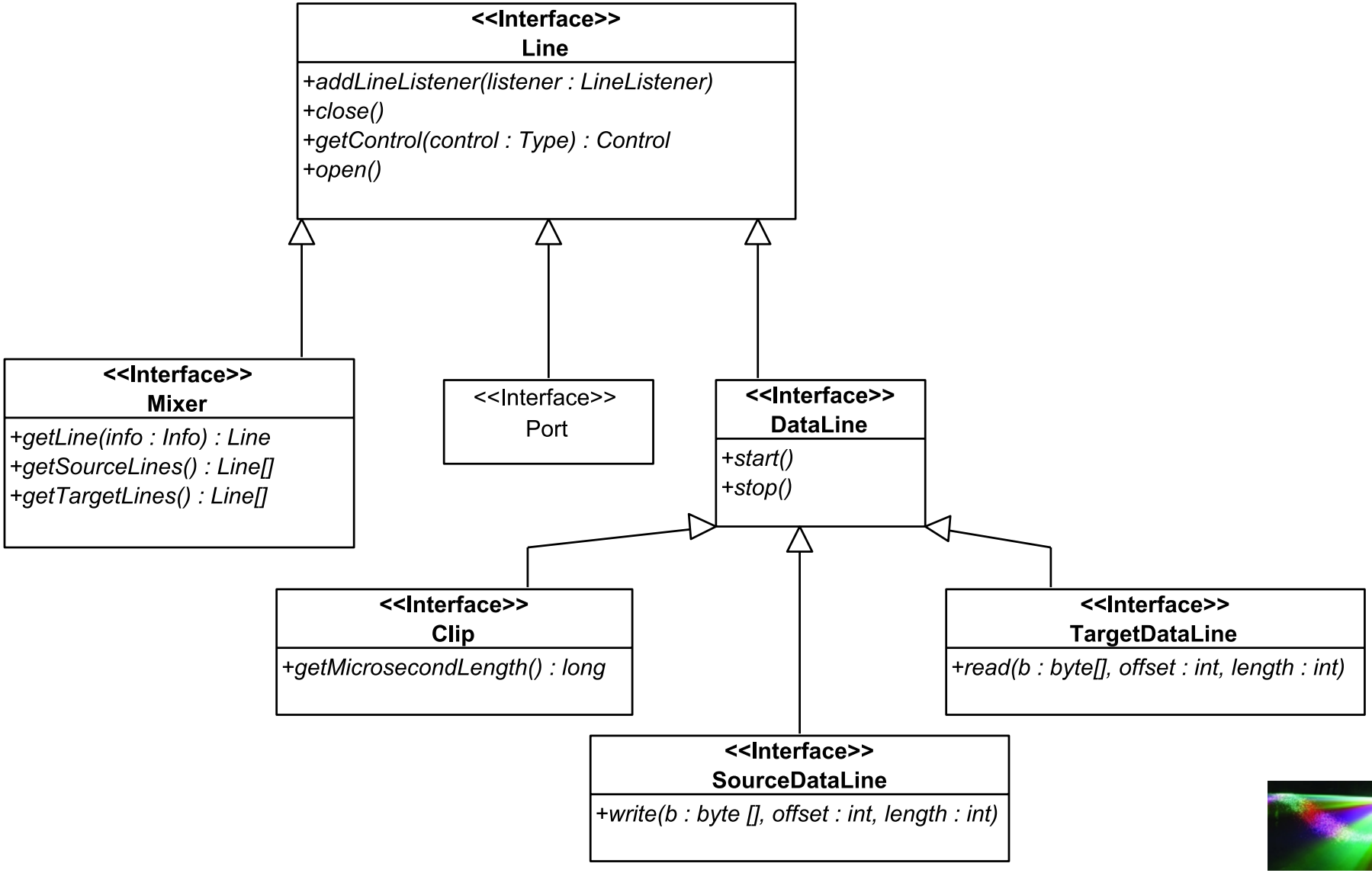
In extras:

FIRFilterOp.html

```
java -cp BufferedSound.jar BufferedSoundApplication UNARY preface.aif FIR
```

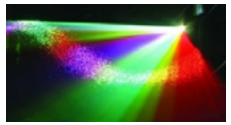


The Java Sound API



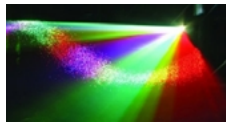
BoomBox – Structure

```
package auditory.sampled;

import java.util.*;
import javax.sound.sampled.*;

public class BoomBox implements LineListener
{
    private Content         content;
    private Clip            clip;
    private final Object    sync = new Object();

    public BoomBox(Content content)
    {
       this.content = content;
    }
}
```



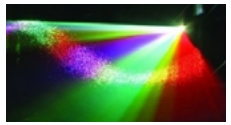
BoomBox – Listeners

```
private Vector<LineListener> listeners = new Vector<LineListener>();
```



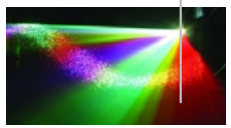
BoomBox – Managing Listeners

```
public void addLineListener(LineListener listener)
{
   listeners.add(listener);
}

public void removeLineListener(LineListener listener)
{
   listeners.remove(listener);
}
```



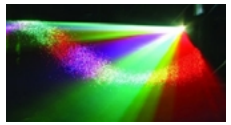
BoomBox – update()

```
public void update(LineEvent evt)
{
   Enumeration      e;
   LineEvent.Type   type;
   LineListener     listener;

   synchronized(sync)
   {
      // Forward the LineEvent to all LineListener objects
      e = listeners.elements();
      while (e.hasMoreElements())
      {
         listener = (LineListener)e.nextElement();
         listener.update(evt);
      }

      // Get the type of the event
      type = evt.getType();

      // Process STOP events
      if (type.equals(LineEvent.Type.STOP))
      {
         sync.notifyAll();
         clip.close();
         clip.removeLineListener(this);
         clip = null;
      }
   }
```



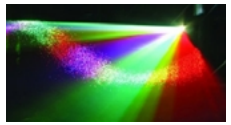
BoomBox – update() (cont.)

```
    }
}
```



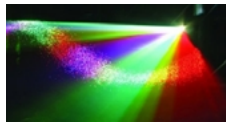
BoomBox – Rendering

```
public void start(boolean block)
            throws LineUnavailableException
{
   Clip                clip;

   clip = AudioSystem.getClip();
   clip.addLineListener(this); // So the calling thread can be informed
   content.render(clip);
   synchronized(sync)
   {
      // Wait until the Clip stops [and notifies us by
      // calling the update() method]
      if (block)
      {
         try
         {
            sync.wait();
         }
         catch (InterruptedException ie)
         {
            // Ignore
         }
      }
   }
}
```



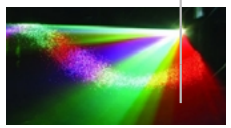
BufferedSound - Scaling

```
private short scaleSample(double sample)
{
   short     scaled;

   if      (sample > MAX_AMPLITUDE) scaled=(short)MAX_AMPLITUDE;
   else if (sample < MIN_AMPLITUDE) scaled=(short)MIN_AMPLITUDE;
   else                             scaled=(short)sample;

   return scaled;
}
```



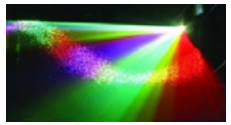
BufferedSound – Rendering

```
public synchronized void render(Clip clip)
                    throws LineUnavailableException
   size   = channels.size();
   length = getNumberOfSamples();
   frameSize  = format.getFrameSize();

   //  bytes           samples/channel *  bytes/channel     *  channels
   rawBytes = new byte[length          *  BYTES_PER_CHANNEL *     size];
   channel  = 0;
   iterator = channels.iterator();
   while (iterator.hasNext())
   {
      signal = iterator.next();
      offset = channel * BYTES_PER_CHANNEL;

      for (int i=0; i<length; i++)
      {
         scaled = scaleSample(signal[i]);

         // Big-endian
         rawBytes[frameSize*i+offset]   = (byte)(scaled >> 8);
         rawBytes[frameSize*i+offset+1] = (byte)(scaled & 0xff);

         // Little-endian
         // rawBytes[frameSize*i+offset+1] = (byte)(scaled >> 8);
         // rawBytes[frameSize*i+offset]   = (byte)(scaled & 0xff);
      }
      ++channel;
```



BufferedSound – Rendering (cont.)

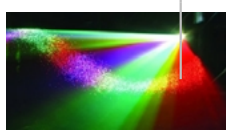
```
        }
        // Throws LineUnavailableException
        clip.open(format, rawBytes, 0, rawBytes.length);

        // Start the Clip
        clip.start();
```



NoiseOp

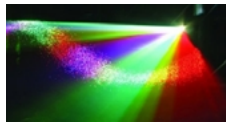
Recall that noise is a signal that is generated by a random process.

```
package auditory.sampled;

import java.util.Random;

public class NoiseOp extends    AbstractBufferedSoundUnaryOp
{
    private double        max;
    private Random        rng;

    public NoiseOp(double max)
    {
       this.max = max;
       rng = new Random(System.currentTimeMillis());

    }

    public void applyFilter(double[] source, double[] destination)
    {
       int       length;

       length    = source.length;

       for (int i=0; i<length; i++)
       {
          destination[i]  = source[i]  + (max - rng.nextDouble()*max*2.0);
```

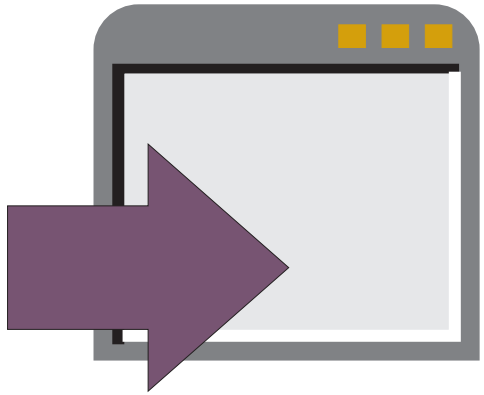


NoiseOp (cont.)

```
        }
    }
}
```



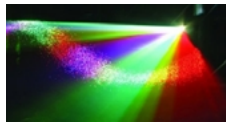
NoiseOp – Demonstration



In extras:

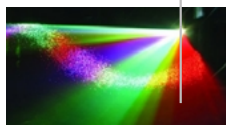
NoiseOp.html

```
java -cp BufferedSound.jar BufferedSoundApplication UNARY preface.aif NOISE
```



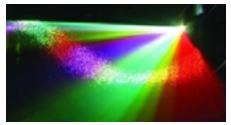
SpeedChangeOp

```
package auditory.sampled;

public class SpeedChangeOp extends AbstractBufferedSoundUnaryOp
{
    private double        multiplier;

    public SpeedChangeOp(double multiplier)
    {
       this.multiplier = multiplier;
    }

    public BufferedSound createCompatibleDestinationSound(BufferedSound src)
    {
       BufferedSound        temp;
       float                sampleRate;
       int                  channels, length;

       channels   = src.getNumberOfChannels();
       length     = src.getNumberOfSamples();
       sampleRate = src.getSampleRate() * (float)multiplier;

       temp = new BufferedSound(sampleRate);

       for (int i=0; i<channels; i++)
       {
          temp.addChannel(new double[length]);
       }
```

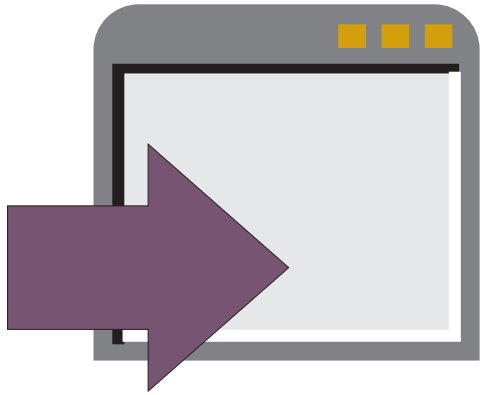


SpeedChangeOp (cont.)

```
        return temp;
    }

    public void applyFilter(double[] source, double[] destination)
    {
        int       length;

        length    = source.length;

        for (int i=0; i<length; i++)
        {
            destination[i]  = source[i];
        }
    }
}
```



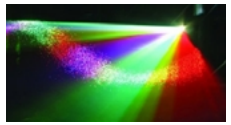
SpeedChangeOp – Demonstration



In extras:

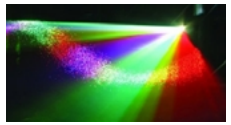
SpeedChangeOp.html

```
java -cp BufferedSound.jar BufferedSoundApplication UNARY preface.aif SPEEDCHANGE
```

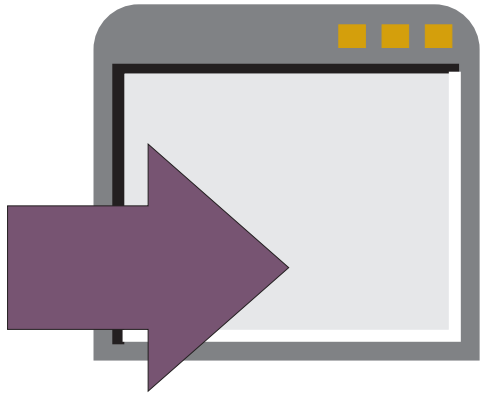


MultiplyOp

```
package auditory.sampled;

public class MultiplyOp extends AbstractBufferedSoundBinaryOp
{
    public void applyFilter(double[] source1, double[] source2,
                            double[] destination)
    {
       for (int i=0; i<source1.length; i++)
       {
             destination[i] = source1[i] * source2[i];
       }
    }
}
```



MultiplyOp – Demonstration



In extras:

MultiplyOp.html

```
java -cp BufferedSound.jar BufferedSoundApplication BINARY 100 MULTIPLY 200
```