

CPSC 501 - Assignment 2

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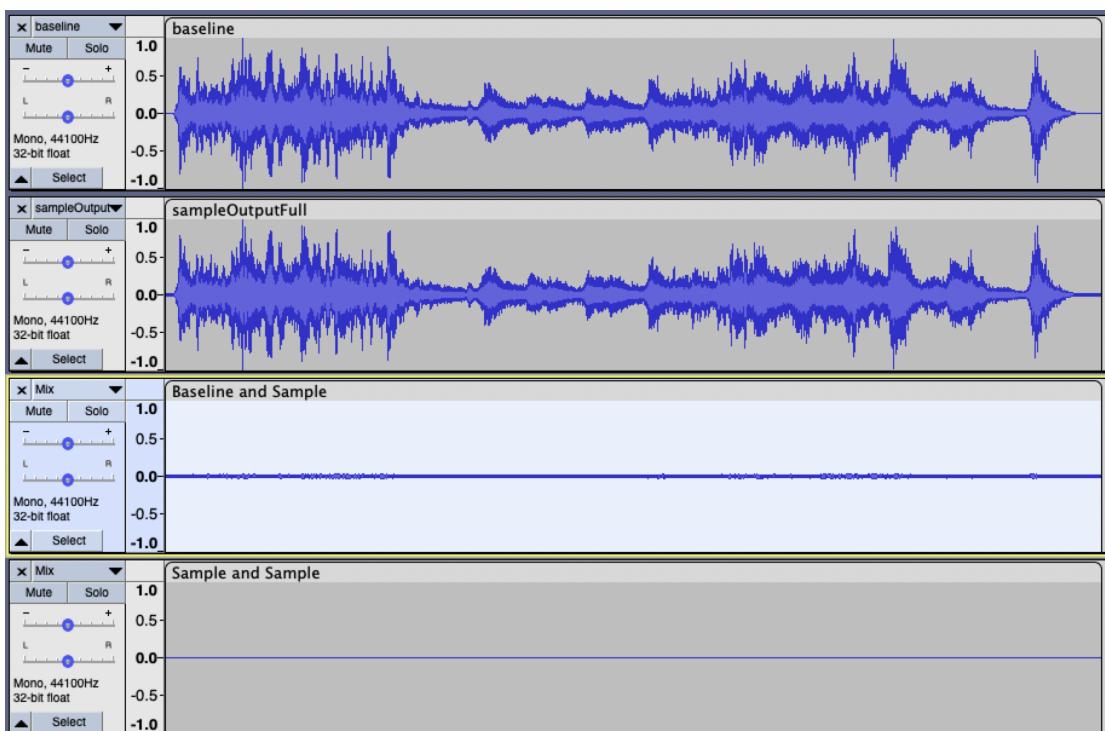
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TA Notes: Access to my gitlab repository can be found by clicking this link: <https://gitlab.cpsc.ucalgary.ca/alex.tanasescu/assignment-2-alex-tanasescu>. You should've also received an invitation to it in your email when I first invited you to my repo.

Regression Testing: For regression testing I will perform 3 different tests. The first test I will perform is the **audacity test**: importing the outputs from the two different programs in audacity, inverting one of the outputs and mixing them together in one track. If the two files are identical, then the frequencies will cancel each other out and the mix output will be silent. Below is an example comparing my baseline convolution program to the prof's sample output as well as identical sample output files for reference. We can see that the identical files make no sound and the graph is flat while there is a slight difference in my baseline and sample output. This can be confirmed with the **cmp test** where we use the linux command cmp to see if two files are the same, if they are there will be nothing returned however if they aren't, it will return the first byte at which they differ also demonstrated below with my baseline output and the prof's sample output. Finally the last test I will do is the **hearing test** where I listen if the outputs are similar. Even if the bits of the baseline and the sample are different, the outputs are similar which will be good enough for the purposes of this assignment.

Timings: To measure the timings, I ran each program 5 times (except the baseline since it takes a really long time) to have a fair sample size to understand how the code behaves. I then got rid of the lowest and high test time. I then averaged the 3 remaining times for the final comparison.

Instructions to Run: to run the program you can use the **g++ -O2 convolve.cpp -o convolve** to compile and **./convolve guitar_dry.wav big_hall_IR_mono.wav output.wav** to run. output.wav will be the output of the program which you can listen to



```
alex.tanasescu@linux02-wa:~/CPSC-501/A2$ cmp baseline.wav sampleOutputFull.wav
baseline.wav sampleOutputFull.wav differ: byte 45, line 1
alex.tanasescu@linux02-wa:~/CPSC-501/A2$
```

Baseline: The timings for my baseline took around just under 8 minutes (463 second/60 seconds = 7.7 min) as pictured below. I already demonstrated above how similar it is to the sample output even if bitwise it's different so it will make a good standard. This program is what we will use to compare all our optimizations.

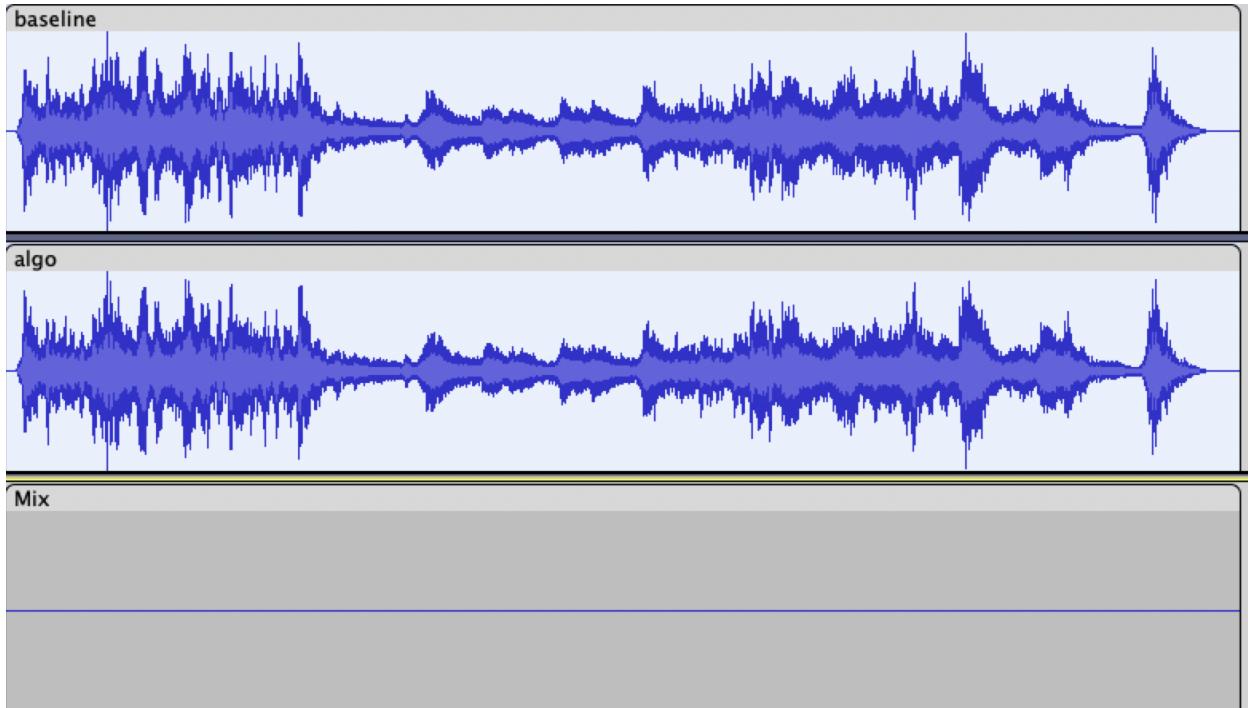
Each sample counts as 0.01 seconds.							
	% cumulative	self	self	total			
time	seconds	seconds	calls	ms/call	ms/call	name	
100.11	463.59	463.59				main	
0.01	463.63	0.04	2	20.03	20.03	readWavFile(int*, int*, char*)	
0.00	463.65	0.02	1	20.03	20.03	writeWavFile(double*, int, int, char*)	
0.00	463.65	0.00	1876187	0.00	0.00	std::abs(double)	
0.00	463.65	0.00	10	0.00	0.00	freadIntLSB(_IO_FILE*)	
0.00	463.65	0.00	8	0.00	0.00	freadShortLSB(_IO_FILE*)	
0.00	463.65	0.00	5	0.00	0.00	fwriteIntLSB(int, _IO_FILE*)	
0.00	463.65	0.00	4	0.00	0.00	fwriteShortLSB(short, _IO_FILE*)	
0.00	463.65	0.00	2	0.00	0.00	readWavFileHeader(int*, int*, _IO_FILE*)	
0.00	463.65	0.00	1	0.00	0.00	_GLOBAL__sub_I_main	
0.00	463.65	0.00	1	0.00	0.00	writeWavFileHeader(int, int, double, _IO_FILE*)	
0.00	463.65	0.00	1	0.00	0.00	__static_INITIALIZATION_and_destruction_0(int, int)	

Algorithmic Optimization: The first optimization I did is performing the convolution using the Fast Fourier Transform. This increased the speed of execution dramatically from 7.7 minutes to 3.03 seconds using the timings below.

Each sample counts as 0.01 seconds.							
	% cumulative	self	self	total			
time	seconds	seconds	calls	s/call	s/call	name	
15.13	0.47	0.47	3	0.16	0.94		
11.43	0.83	0.36	66060288	0.00	0.00		
9.18	1.11	0.29	33846896	0.00	0.00		
8.69	1.38	0.27	12582906	0.00	0.00		
6.76	1.59	0.21	12582908	0.00	0.00		
6.76	1.80	0.21	202375168	0.00	0.00		
4.35	1.94	0.14	66060288	0.00	0.00		
4.19	2.07	0.13	68157440	0.00	0.00		
3.70	2.18	0.12	66060288	0.00	0.00		
3.38	2.29	0.11	68157440	0.00	0.00		
2.74	2.37	0.09	12582906	0.00	0.00		
2.58	2.45	0.08	70254593	0.00	0.00		
2.58	2.53	0.08	66060288	0.00	0.00		
2.41	2.61	0.08	12582906	0.00	0.00		
2.25	2.68	0.07	136314880	0.00	0.00		
1.61	2.73	0.05	1	0.05	3.06		
1.29	2.77	0.04	4194304	0.00	0.00		
1.13	2.80	0.04	4194304	0.00	0.00		
0.97	2.83	0.03	66060288	0.00	0.00		
0.97	2.86	0.03	2	0.02	0.02		
0.80	2.89	0.03	12582906	0.00	0.00		
0.64	2.91	0.02	37748721	0.00	0.00		
0.64	2.93	0.02	12582908	0.00	0.00		
0.64	2.95	0.02	12582906	0.00	0.00		
0.64	2.97	0.02	1	0.02	0.03		
0.48	2.98	0.02	12582908	0.00	0.00		
0.48	3.00	0.02	12582906	0.00	0.00		
0.32	3.01	0.01	12582908	0.00	0.00		
0.32	3.02	0.01	12582906	0.00	0.00		
0.32	3.03	0.01	4194304	0.00	0.00		
0.32	3.04	0.01	4194304	0.00	0.00		
0.32	3.05	0.01	1876187	0.00	0.00		
0.32	3.06	0.01	2	0.01	0.01		
0.32	3.07	0.01	2	0.01	0.02		
0.32	3.08	0.01	2	0.01	0.05		
0.32	3.09	0.01	1	0.01	0.01		
0.32	3.10	0.01					
0.16	3.10	0.01	12582906	0.00	0.00		
0.16	3.11	0.00	14680065	0.00	0.00		
0.00	3.11	0.00	14680065	0.00	0.00		
0.00	3.11	0.00	12582906	0.00	0.00		

Each sample counts as 0.01 seconds.							
	% cumulative	self	self	total			
time	seconds	seconds	calls	s/call	s/call	name	
12.06	0.36	0.36	3	0.12	0.91		
9.04	0.63	0.27	12582906	0.00	0.00		
8.20	0.88	0.25	66060288	0.00	0.00		
7.37	1.10	0.22	202375168	0.00	0.00		
6.87	1.30	0.21	68157440	0.00	0.00		
6.87	1.51	0.21	338468966	0.00	0.00		
6.70	1.71	0.20	12582908	0.00	0.00		
5.36	1.87	0.16	66060288	0.00	0.00		
4.69	2.01	0.14	70254593	0.00	0.00		
4.02	2.13	0.12	68157440	0.00	0.00		
3.18	2.22	0.10	66060288	0.00	0.00		
2.85	2.31	0.09	66060288	0.00	0.00		
2.34	2.38	0.07	66060288	0.00	0.00		
2.34	2.45	0.07	4194304	0.00	0.00		
2.01	2.51	0.06	136314880	0.00	0.00		
1.67	2.56	0.05	12582906	0.00	0.00		
1.67	2.61	0.05	12582906	0.00	0.00		
1.67	2.66	0.05	12582906	0.00	0.00		
1.34	2.70	0.04	1	0.04	2.93		
1.00	2.73	0.03	37748721	0.00	0.00		
1.00	2.76	0.03	4194304	0.00	0.00		
0.84	2.78	0.03	12582906	0.00	0.00		
0.67	2.80	0.02	12582908	0.00	0.00		
0.67	2.82	0.02	12582908	0.00	0.00		
0.67	2.84	0.02	12582906	0.00	0.00		
0.67	2.86	0.02	1	0.02	0.02		
0.67	2.88	0.02					
0.33	2.89	0.01	14680065	0.00	0.00		
0.33	2.90	0.01	12582908	0.00	0.00		
0.33	2.91	0.01	12582906	0.00	0.00		
0.33	2.92	0.01	12582906	0.00	0.00		
0.33	2.93	0.01	12582906	0.00	0.00		
0.33	2.94	0.01	4194304	0.00	0.00		
0.33	2.95	0.01	4194304	0.00	0.00		
0.33	2.96	0.01	2	0.01	0.01		
0.33	2.97	0.01	2	0.01	0.01		
0.17	2.98	0.01	12582906	0.00	0.00		
0.00	2.98	0.00	6070374	0.00	0.00		
0.00	2.98	0.00	4194304	0.00	0.00		
0.00	2.99	0.00	202375168	0.00	0.00		

We can also see from the audacity test and the cmp test that the baseline and algorithm optimization wav files are identical. I also hear no difference between the two sound files



```
alex.tanasescu@linux02-wa:~/CPSC-501/A2$ cmp baseline.wav algo.wav
alex.tanasescu@linux02-wa:~/CPSC-501/A2$ █
```

To perform the optimization I changed the code from the convolution given to us in class

```
89     double *outputArray = new double[outputArraySize];
90
91
92     for (int i = 0; i < IRArraySize; i++)
93     {
94         for (int j = 0; j < inputArraySize; j++)
95         {
96             outputArray[i + j] += IRArray[i] * inputArray[j];
97         }
98     }
99 }
```

To the following convolution using the Fast Fourier Transform algorithm:

```
149 double* convolution(double* inputArray, int inputArraySize, double* IRArray, int IRArraySize, int* outputArraySize)
150 {
151     double *outputArray = new double[*outputArraySize];
152     int complexArraySize = *outputArraySize;
153
154     if (!isPowerOfTwo(complexArraySize))
155     {
156         complexArraySize = nextPowTwo(complexArraySize);
157     }
158
159     double* inputPadded = new double[complexArraySize];
160     double* IRPadded = new double[complexArraySize];
161
162     for (int i = 0; i < complexArraySize; i++)
163     {
164         if (i < inputArraySize)
165         {
166             inputPadded[i] = inputArray[i];
167         }
168         else
169         {
170             inputPadded[i] = 0;
171         }
172     }
173
174     for (int i = 0; i < complexArraySize; i++)
175     {
176         if (i < IRArraySize)
177         {
178             IRPadded[i] = IRArray[i];
179         }
180         else
181         {
182             IRPadded[i] = 0;
183         }
184     }
185
186     double* inputComplex = new double[complexArraySize*2];
187     double* IRComplex= new double[complexArraySize*2];
188
189     for (int i = 0; i < complexArraySize; i++)
190     {
191         inputComplex[i*2] = inputPadded[i];
192     }
193
194     for (int i = 0; i < complexArraySize; i++)
195     {
196         inputComplex[i*2+1] = 0;
197     }
198
199     for (int i = 0; i < complexArraySize; i++) {
200         IRComplex[i*2] = IRPadded[i];
201     }
202
203     for (int i = 0; i < complexArraySize; i++) {
204         IRComplex[i*2+1] = 0;
205     }
206
207     CArray inputData((Complex*)inputComplex, complexArraySize);
208     CArray IRData((Complex*)IRComplex, complexArraySize);
209
210     fft(inputData);
211     fft(IRData);
212
213     double* outputData = new double[*outputArraySize];
214     for (int i = 0; i < inputData.size(); i++)
215     {
216         inputData[i] = inputData[i]+IRData[i];
217     }
218
219     ifft(inputData);
220
221     for (int i = 0; i < *outputArraySize; i++)
222     {
223
224         outputData[i] = inputData[i].real();
225     }
226
227     return outputData;
228
229
230 }
```

Compiler Optimization: The second optimization I performed is the compiler optimization. I used the -O2 flag to optimize my code. We can see another dramatic speed up from the 3.03 seconds we had with the basic Fast Fourier Transformation to an average of 0.58 seconds using the Fast Fourier Transformation with the optimization flag. We can also see that both wav files are exactly the same with our audacity and cmp test which is expected since we didn't change any code. As well as sounding the same to my ears. We will use this flag for all of our following optimizations.

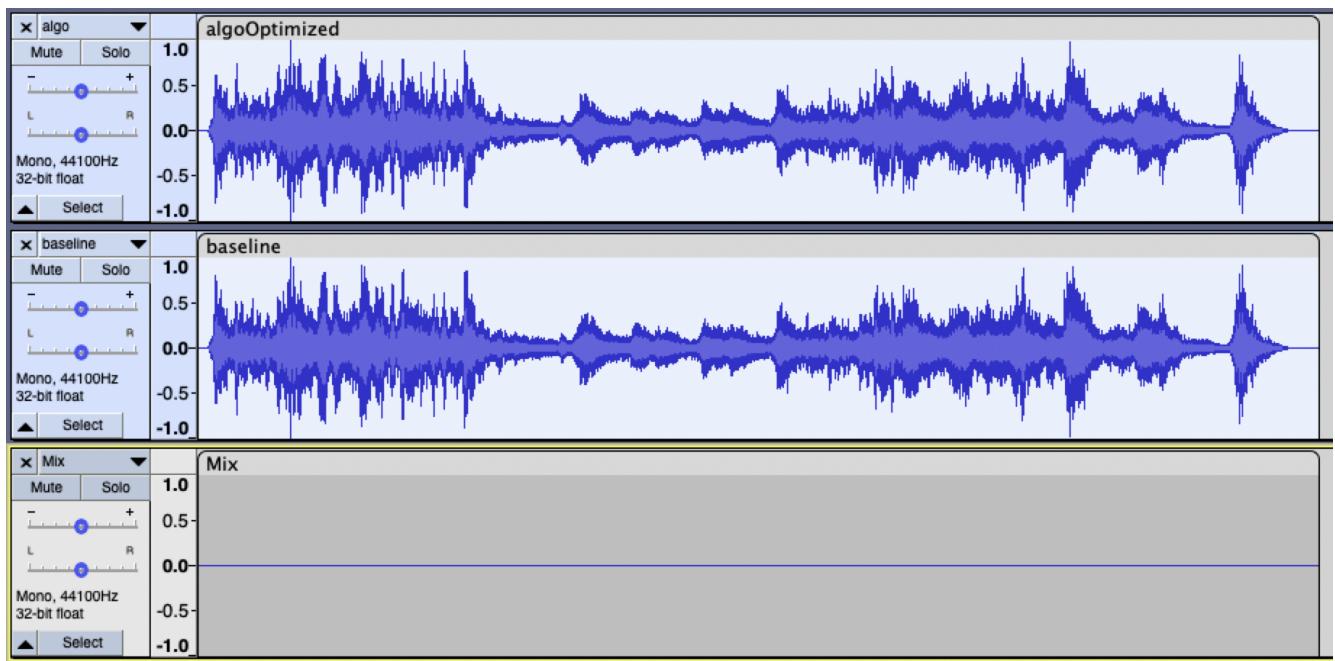
```
alex.tanasescu@linux02-wa:~/CPSC-501/A2$ cmp algoOptimized.wav baseline.wav
alex.tanasescu@linux02-wa:~/CPSC-501/A2$
```

Each sample counts as 0.01 seconds.						
%	cumulative	self		self	total	
time	seconds	seconds	calls	ms/call	ms/call	
86.89	0.52	0.52	3	173.78	173.78	
10.03	0.58	0.06	1	60.15	591.50	
1.67	0.59	0.01	1	10.03	10.03	
1.67	0.60	0.01	1	10.03	183.80	
0.00	0.60	0.00	4194304	0.00	0.00	
0.00	0.60	0.00	10	0.00	0.00	
0.00	0.60	0.00	8	0.00	0.00	

Each sample counts as 0.01 seconds.						
%	cumulative	self		self	total	
time	seconds	seconds	calls	ms/call	ms/call	
87.72	0.53	0.53	3	175.45	175.45	
10.03	0.59	0.06	1	60.15	591.50	
0.84	0.59	0.01	4194304	0.00	0.00	
0.84	0.60	0.01	2	2.51	2.51	
0.84	0.60	0.01	1	5.01	5.01	
0.00	0.60	0.00	10	0.00	0.00	
0.00	0.60	0.00	8	0.00	0.00	

Each sample counts as 0.01 seconds.

%	cumulative	self		self	total	
time	seconds	seconds	calls	ms/call	ms/call	
87.50	0.48	0.48	3	160.41	160.41	
12.76	0.55	0.07	1	70.18	551.40	
0.00	0.55	0.00	4194304	0.00	0.00	
0.00	0.55	0.00	10	0.00	0.00	
0.00	0.55	0.00	8	0.00	0.00	



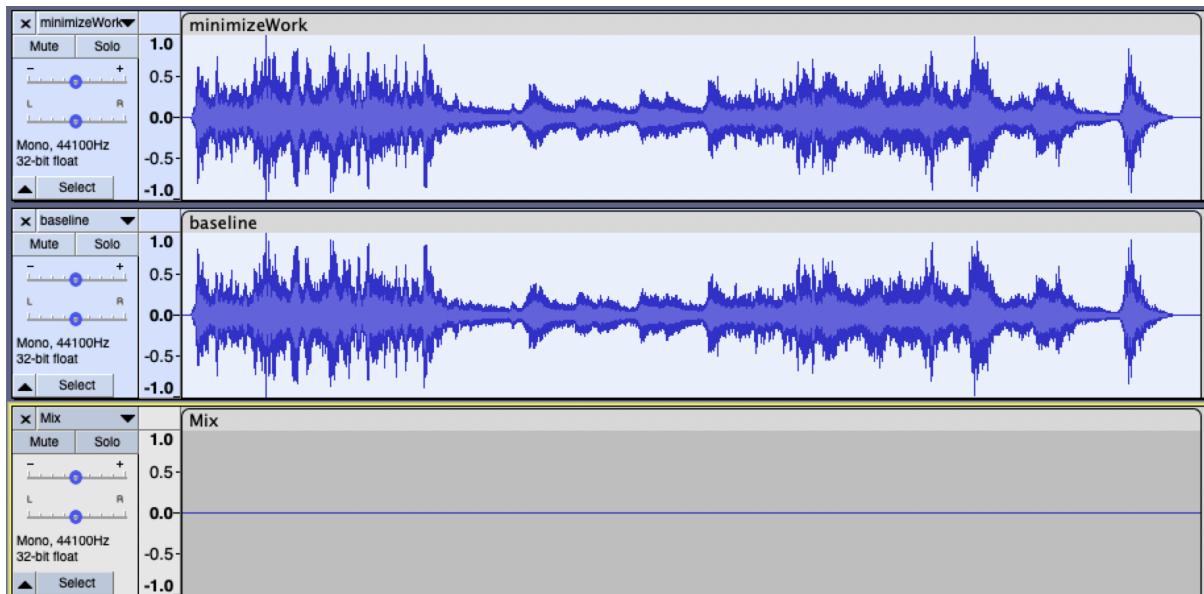
1st Code Tuning - Minimize Work: The first code tuning I performed is the minimize work. Inside my Fast Fourier method, that contains a loop and inside that it calculates the polar form of a complex number. This calculation is performed every time the loop is run. To optimize this, I simply did a calculation outside the loop and multiplied that with the loop counter. Shown below is snippets of my code before and after optimization for reference.

```
111     // combine
112     for (size_t k = 0; k < N/2; ++k)
113     {
114         Complex t = polar(1.0, -2 * PI * k / N) * odd[k];
115         x[k] = even[k] + t;
116         x[k+N/2] = even[k] - t;
117     }
```



```
111     // combine
112     double toMultiply = -2 * PI * 1/N;
113     for (size_t k = 0; k < N/2; ++k)
114     {
115         Complex t = polar(1.0, toMultiply*k) * odd[k];
116         x[k] = even[k] + t;
117         x[k+N/2] = even[k] - t;
118     }
119 }
120 }
```

We can see in the audacity and cmp test that the baseline and the output of the code tuning have the same contents so nothing has been changed as expected, as well as sounding the same. We can also see in the timings that there is a small but noticeable speedup in the code when we apply the code tuning technique in our original code along with the compiler optimization flag from 0.58 seconds to an average of 0.54. This is all shown on the next page



```
alex.tanasescu@linux02-wa:~/CPSC-501/A2$ cmp minimizeWork.wav baseline.wav
alex.tanasescu@linux02-wa:~/CPSC-501/A2$
```

Each sample counts as 0.01 seconds.

%	cumulative	self		self	total
time	seconds	seconds	calls	ms/call	ms/call
88.91	0.47	0.47	3	157.07	157.07
9.46	0.52	0.05	1	50.13	521.33
1.89	0.53	0.01	1	10.03	10.03
0.00	0.53	0.00	4194304	0.00	0.00
total	0.53	0.53			

time	seconds	seconds	calls	ms/call	ms/call
88.16	0.51	0.51	3	170.43	170.43
8.64	0.56	0.05	1	50.13	561.43
1.73	0.57	0.01	2	5.01	5.01
1.73	0.58	0.01	1	10.03	10.03
0.00	0.58	0.00	4194304	0.00	0.00

Each sample counts as 0.01 seconds.

%	cumulative	self		self	total
time	seconds	seconds	calls	ms/call	ms/call
86.76	0.45	0.45	3	150.38	150.38
7.71	0.49	0.04	1	40.10	501.28
1.93	0.50	0.01	2	5.01	5.01
1.93	0.51	0.01	2	5.01	5.01
unClos<std::ValArray, std::complex<double> >, std::complex<double>	0.51	0.01			
1.93	0.52	0.01	1	10.03	10.03
0.00	0.52	0.00	4194304	0.00	0.00
0.00	0.52	0.00	10	0.00	0.00

2nd Code Tuning - Loop Jamming: The second code tuning technique I performed is loop jamming. This is when two or more loops are merged in a single loop. This helps speed up execution time since it reduces the time to compile each separate loop. I had two different sets of two loops. I ended up merging the two loops in each set into one loop since they had the same loop counter and exit condition as shown in the code below.

```
177     for (int i = 0; i < complexArraySize; i++)
178     {
179         inputComplex[i*2] = inputPadded[i];
180     }
181
182     for (int i = 0; i < complexArraySize; i++)
183     {
184         inputComplex[i*2+1] = 0;
185     }
186
187     for (int i = 0; i < complexArraySize; i++) {
188         IRComplex[i*2] = IRPadded[i];
189     }
190
191     for (int i = 0; i < complexArraySize; i++) {
192         IRComplex[i*2+1] = 0;
193     }
194
```

```
176
177     for (int i = 0; i < complexArraySize; i++)
178     {
179         inputComplex[i*2] = inputPadded[i];
180         inputComplex[i*2+1] = 0;
181     }
182
183     for (int i = 0; i < complexArraySize; i++) {
184         IRComplex[i*2] = IRPadded[i];
185         IRComplex[i*2+1] = 0;
186     }
187
```

We can see on the average of the timings below that there is another small but noticeable speed up from the timing. We went from our time of 0.54 from the first optimization to 0.51 now.

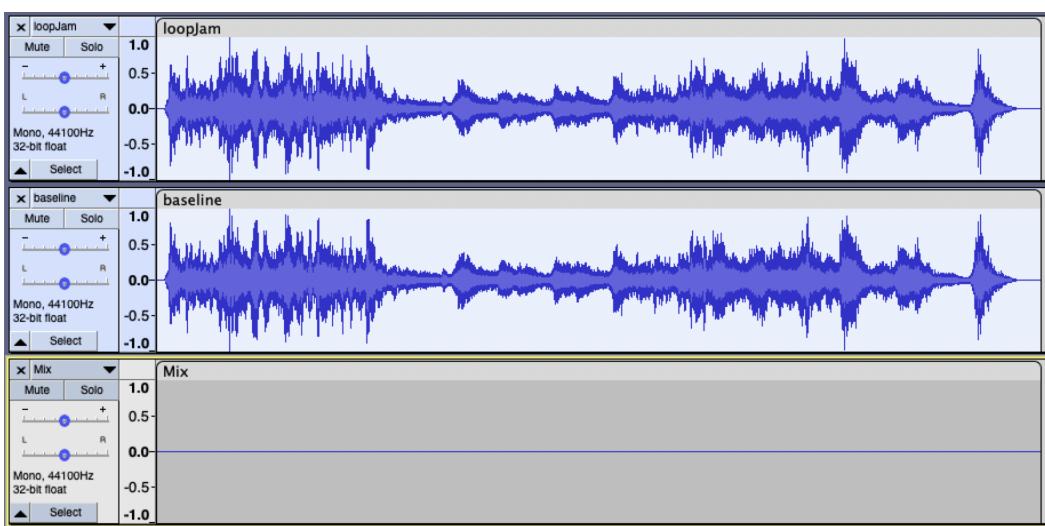
Each sample counts as 0.01 seconds.						
%	cumulative	self		self		total
time	seconds	seconds	calls	ms/call	ms/call	
85.64	0.41	0.41	3	137.02	137.02	
6.27	0.44	0.03	1	30.08	177.12	
6.27	0.47	0.03				
2.09	0.48	0.01	4194304	0.00	0.00	
0.00	0.48	0.00	2	0.00	0.00	

Each sample counts as 0.01 seconds.						
%	cumulative	self		self		total
time	seconds	seconds	calls	ms/call	ms/call	
91.73	0.43	0.43	3	143.70	143.70	
8.53	0.47	0.04				
0.00	0.47	0.00	4194304	0.00	0.00	
0.00	0.47	0.00	2	0.00	0.00	

time	seconds	seconds	calls	ms/call	ms/call
88.16	0.51	0.51	3	170.43	170.43
8.64	0.56	0.05	1	50.13	561.43
1.73	0.57	0.01	2	5.01	5.01
1.73	0.58	0.01	1	10.03	10.03
0.00	0.58	0.00	4194304	0.00	0.00

We can also see from the audacity and cmp test that the two wav files are the same in content. The hearing test confirms this as I hear no difference when playing the two sounds

```
alex.tanasescu@ms160-1ee:~/CPSC-501/A2$ cmp loopJam.wav baseline.wav
alex.tanasescu@ms160-1ee:~/CPSC-501/A2$
```



3rd Code Tuning - Unswitching: My 3rd code tuning is a form of unswitching. My initial code has loops that contain an if-else statement. This increases execution time since the compiler has to evaluate the if-else statement each time the loop is run. To fix this, since I know the if statement is going to be true for the first half of the loop and the else statement is going to be true for the second half, I split the loop into two sub loops and put the respective statements from each condition in each respective loop. Below is my code for reference

```
149     for (int i = 0; i < complexArraySize; i++)
150     {
151         if (i < inputArraySize)
152         {
153             inputPadded[i] = inputArray[i];
154         }
155         else
156         {
157             inputPadded[i] = 0;
158         }
159     }
160
161     for (int i = 0; i < complexArraySize; i++)
162     {
163         if (i < IRArraySize)
164         {
165             IRPadded[i] = IRArray[i];
166         }
167         else
168         {
169             IRPadded[i] = 0;
170         }
171     }
172 }
```

```
149     for (int i = 0; i < inputArraySize; i++)
150     {
151         inputPadded[i] = inputArray[i];
152     }
153
154
155     for (int i = inputArraySize; i < complexArraySize; i++ )
156     {
157         inputPadded[i] = 0;
158     }
159
160     for (int i = 0; i < IRArraySize; i++)
161     {
162         IRPadded[i] = IRArray[i];
163     }
164
165     for (int i = IRArraySize; i < complexArraySize; i++ )
166     [
167         IRPadded[i] = 0;
168     ]
169 }
```

We can see from the timings below that there is another small speedup but it is still there the speed up is another 0.03 seconds from our timings of before of 0.51 to 0.48

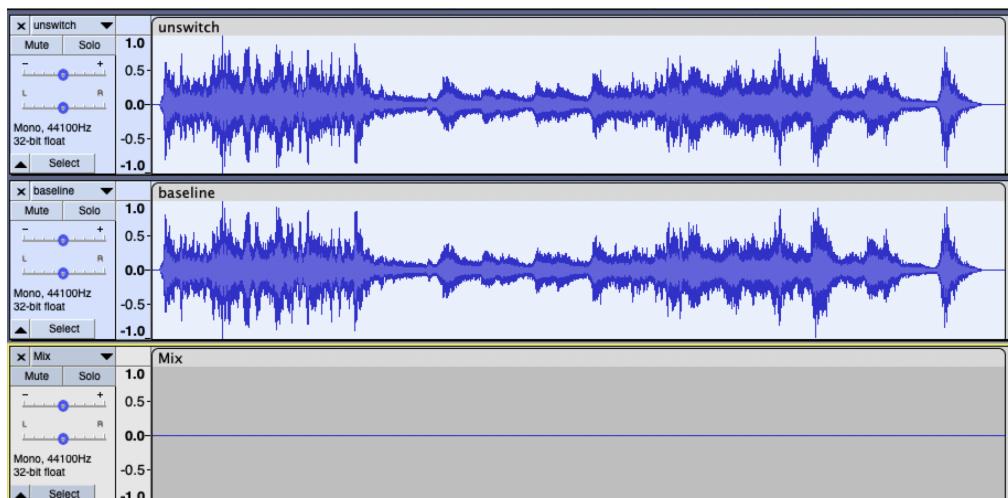
Each sample counts as 0.01 seconds.					
% time	cumulative seconds	self seconds	calls	self ms/call	total ms/call
80.83	0.40	0.40	3	132.02	132.02
6.14	0.43	0.03			
6.14	0.46	0.03			
4.09	0.48	0.02	1	20.05	162.10
2.05	0.49	0.01	4194304	0.00	0.00
1.02	0.49	0.01			
0.00	0.49	0.00	2	0.00	0.00
0.00	0.49	0.00	1	0.00	0.00

Each sample counts as 0.01 seconds.					
% time	cumulative seconds	self seconds	calls	self ms/call	total ms/call
90.24	0.45	0.45	3	150.40	150.40
8.02	0.49	0.04			
2.01	0.50	0.01			
0.00	0.50	0.00	4194304	0.00	0.00
0.00	0.50	0.00	2	0.00	0.00

Each sample counts as 0.01 seconds.					
% time	cumulative seconds	self seconds	calls	self ms/call	total ms/call
89.37	0.41	0.41	3	137.03	137.03
6.54	0.44	0.03			
2.18	0.45	0.01	4194304	0.00	0.00
2.18	0.46	0.01			
0.00	0.46	0.00	2	0.00	0.00

We can see in the audacity and cmp test that the baseline and the output of the code tuning have the same contents so nothing has been changed as expected. As well as sounding the same to my ears.

```
alex.tanasescu@ms160-1ee:~/CPSC-501/A2$ cmp unswitch.wav baseline.wav
alex.tanasescu@ms160-1ee:~/CPSC-501/A2$
```



4th Code Tuning - Partial Unrolling: My last and final code tuning I performed on my code is Partial Unrolling. Partial unrolling is an optimization technique used to decrease execution time. We do this by reducing the number of iterations the loop runs for and adjusts for these cases in the loop body and if statements after the loop body. Pictures are shown below for reference to explain how I applied this technique in my code.

```
double* outputData = new double[*outputArraySize];
for (int i = 0; i < inputData.size(); i++)
{
    inputData[i] = inputData[i]*IRData[i];
}

ifft(inputData);

for (int i = 0; i < *outputArraySize; i++)
{
    outputData[i] = inputData[i].real();
}
```

```
int i;
for ( i = 0; i < inputData.size() - 1; i+=2)
{
    inputData[i] = inputData[i]*IRData[i];
    inputData[i+1] = inputData[i+1]*IRData[i+1];
}
if (i == (inputData.size() - 1))
{
    inputData[i] = inputData[i]*IRData[i];
}

ifft(inputData);
int j;
for (j = 0; j < *outputArraySize - 1; j+=2)
{
    outputData[j] = inputData[j].real();
    outputData[j+1] = inputData[j+1].real();
}

if (j == ( *outputArraySize - 1))
{
    outputData[j] = inputData[j].real();
}
```

We can see from the timings below we have as big of a speedup as the other optimizations (0.03 seconds) as the average speed of our code improved from 0.48 seconds to 0.45 seconds. We can also see from the audacity and cmp test that the two wav files are the same in content. The hearing test confirms this as I hear no difference when playing the two sounds

Each sample counts as 0.01 seconds.

% time	cumulative seconds	self seconds	calls	self ms/call	total ms/call
89.11	0.40	0.40	3	133.66	133.66
8.91	0.44	0.04			
2.23	0.45	0.01	1	10.02	143.69
0.00	0.45	0.00	4194304	0.00	0.00

Each sample counts as 0.01 seconds.

% time	cumulative seconds	self seconds	calls	self ms/call	total ms/call
90.92	0.39	0.39	3	130.32	130.32
9.33	0.43	0.04			
0.00	0.43	0.00	4194304	0.00	0.00
0.00	0.43	0.00	2	0.00	0.00

Each sample counts as 0.01 seconds.

% time	cumulative seconds	self seconds	calls	self ms/call	total ms/call
89.58	0.42	0.42	3	140.34	140.34
6.40	0.45	0.03			
2.13	0.46	0.01	4194304	0.00	0.00
2.13	0.47	0.01	1	10.02	160.39
0.00	0.47	0.00	2	0.00	0.00

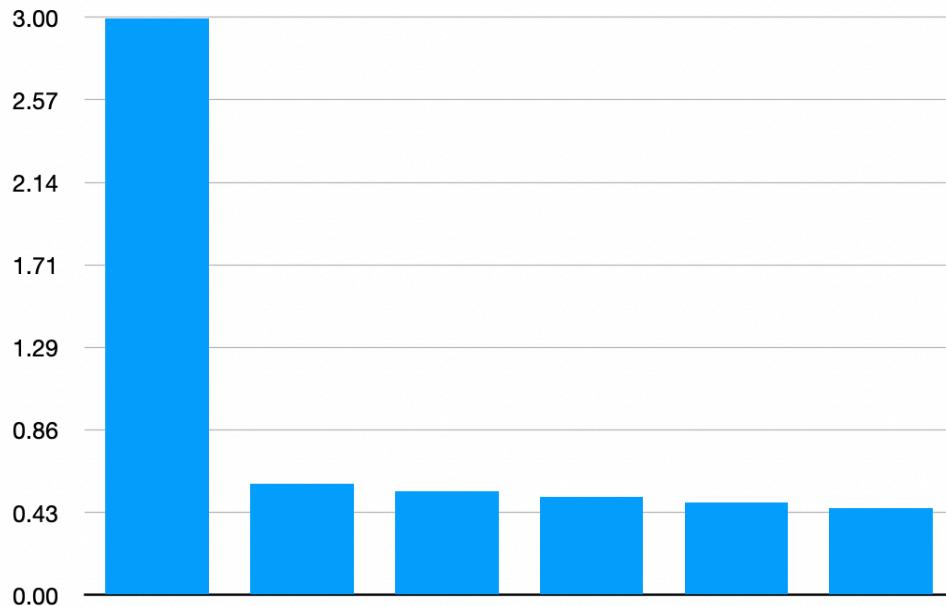


```
alex.tanasescu@linux01-ec:~/CPSC-501/A2$ cmp partialUnroll.wav baseline.wav
alex.tanasescu@linux01-ec:~/CPSC-501/A2$
```

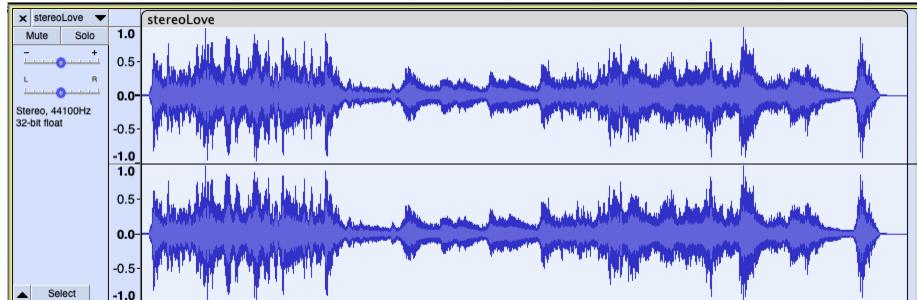
Conclusion: As we can see from the chart and table below(I didn't include baseline in the chart as it took too long and we wouldn't be able to see how the other optimizations performed) the code tuning optimizations I applied gave us a small boost in speed performance but not as much as the compiler optimization or the algorithmic substitution but it's still something that you can do if you want to speed up your code

Execution Speed for Different Optimizations

Optimization	Execution Speed
Baseline	463.59
FFT Algorithm w/o O2 Flag	3.03
FFT Algorithm w/ O2 Flag	0.58
Minimize Work w/ O2 Flag	0.54
Loop Jamming w/ O2 Flag	0.51
Unswitching w/ O2 Flag	0.48
Partial Unrolling w/O2 Flag	0.45



Bonus: I did attempt and finished the bonus portion of the assignment. I convolved the dry recording of the guitar with the stereo big hall impulse recording file. Not quite sure on how to test if it worked properly but I managed to download a VS Code extension that shows you the properties of a wav file and it shows that there are two channels. I also checked Audacity and that shows that there are two channels now so I'm assuming it worked. Below is also the code I added to my program to detect stereo files basically I check if the IR recording has two channels and if it does I convert the dry recording to stereo as well and tell the write wav file header method to write to a stereo file.



Key	Value
format	1 (uncompressed PCM)
number of channel	2 (stereo)
sampleRate	44100
byteRate	176400
blockAlign	4
bitsPerSample (bit depth)	16
fileSize	7430730 byte
duration	42.1240589569161s

```

if (IRChannels == 2)
{
    inputChannels = 2;
    outputChannels = 2;
    inputArray = convertToStereo(inputArray, &inputArraySize, inputArraySize);
}

/* Number of channels */

if (channels == 1)
{
    fwriteShortLSB((short) MONOPHONIC, outputFile);
}
else
{
    fwriteShortLSB((short) STEREOPHONIC, outputFile);
}

```