Real Time Signal Processing

CPE 381 Foundations of Signals & Systems
for Computer Engineers

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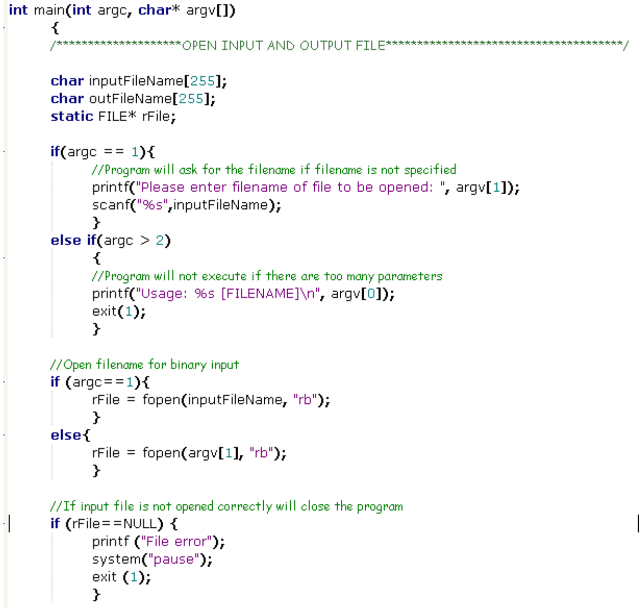
Project: File I/O

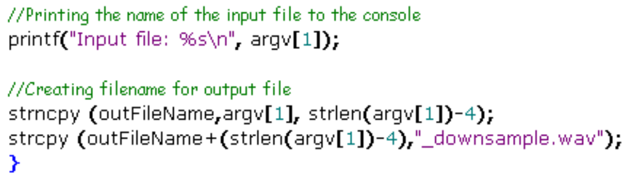
- Provide file name
- command line arg



```
int main(int argc, char* argv[])
	{
	/********************OPEN INPUT AND OUTPUT FILE****************************************/

	char inputFileName[255];
	char outFileName[255];
	static FILE* rFile;

	if(argc == 1){
		//Program will ask for the filename if filename is not specified
		printf("Please enter filename of file to be opened: ", argv[1]);
		scanf("%s",inputFileName);
		}
	else if(argc > 2)
		{
		//Program will not execute if there are too many parameters
		printf("Usage: %s [FILENAME]\n", argv[0]);
		exit(1);
		}

	//Open filename for binary input
	if (argc==1){
		rFile = fopen(inputFileName, "rb");
		}
	else{
		rFile = fopen(argv[1], "rb");
		}

	//If input file is not opened correctly will close the program
	if (rFile==NULL) {
		printf ("File error");
		system("pause");
		exit (1);
		}
```

Project: File I/O

- get output file ready

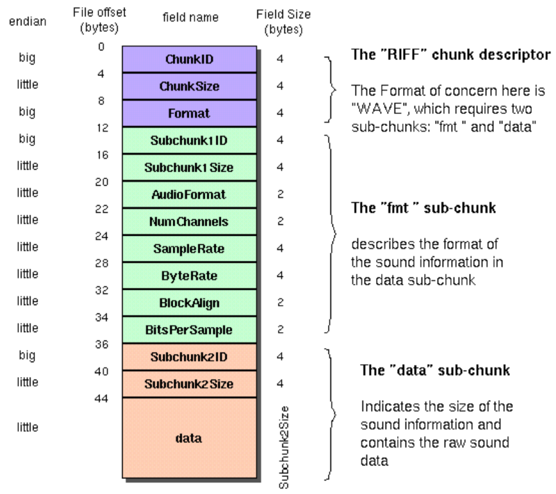


```
//Printing the name of the input file to the console
printf("Input file: %s\n", argv[1]);

//Creating filename for output file
strncpy (outFileName,argv[1], strlen(argv[1])-4);
strcpy (outFileName+(strlen(argv[1])-4),"_downsample.wav");
}
```

Reading formatted files #1

- WAV file header

The Canonical WAVE file format



endian	File offset (bytes)	field name	Field Size (bytes)
big	0	ChunkID	4
little	4	ChunkSize	4
big	8	Format	4
big	12	Subchunk1 ID	4
little	16	Subchunk1 Size	4
little	20	AudioFormat	2
little	22	NumChannels	2
little	24	SampleRate	4
little	28	ByteRate	4
little	32	BlockAlign	2
little	34	BitsPerSample	2
big	36	Subchunk2ID	4
little	40	Subchunk2Size	4
little	44	data	Subchunk2Size

The "RIFF" chunk descriptor

The Format of concern here is "WAVE", which requires two sub-chunks: "fmt " and "data"

The "fmt " sub-chunk

describes the format of the sound information in the data sub-chunk

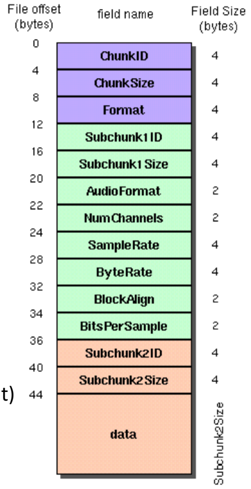
The "data" sub-chunk

Indicates the size of the sound information and contains the raw sound data

Reading formatted files #1

- reading file header

```
//WAV Header File Info
const int HEADER_SIZE = 44;
const int SAMP_RATE_OFFSET = 24;
const int BPS_OFFSET = 34;
const int NUM_CH_OFFSET = 22;
const int AUD_FORM_OFFSET = 20;
const int SUB_CHANK_SIZE_OFFSET = 40;
const int BYTE_RATE_OFFSET = 28;

//The header for PCM is always 44 bytes long
fread(rBuffer, sizeof(char), HEADER_SIZE*sizeof(char), rFile);

sampleSize = *(short*)(rBuffer+BPS_OFFSET);//size of each sample (8/16 bit)
subChunkSize = *(unsigned long*)(rBuffer+SUB_CHANK_SIZE_OFFSET);

//Holds size in bytes the samples take up in the file
sampleRate  = *(long*)(rBuffer+SAMP_RATE_OFFSET);
audioFormat = *(short*)(rBuffer+AUD_FORM_OFFSET);
numOfChannels = *(short*)(rBuffer+NUM_CH_OFFSET);
byteRate = *(long*)(rBuffer+BYTE_RATE_OFFSET);
```

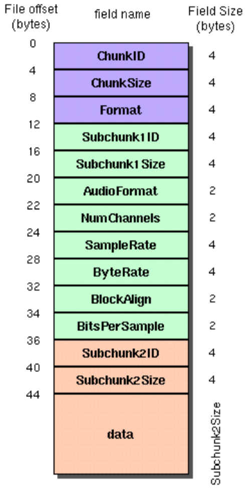


File offset (bytes)	field name	Field Size (bytes)
0	ChunkID	4
4	ChunkSize	4
8	Format	4
12	Subchunk1ID	4
16	Subchunk1Size	4
20	AudioFormat	2
22	NumChannels	2
24	SampleRate	4
28	ByteRate	4
32	BlockAlign	2
34	BitsPerSample	2
36	Subchunk2ID	4
40	Subchunk2Size	4
44	data	Subchunk2Size

Reading formatted files #2

- define structures

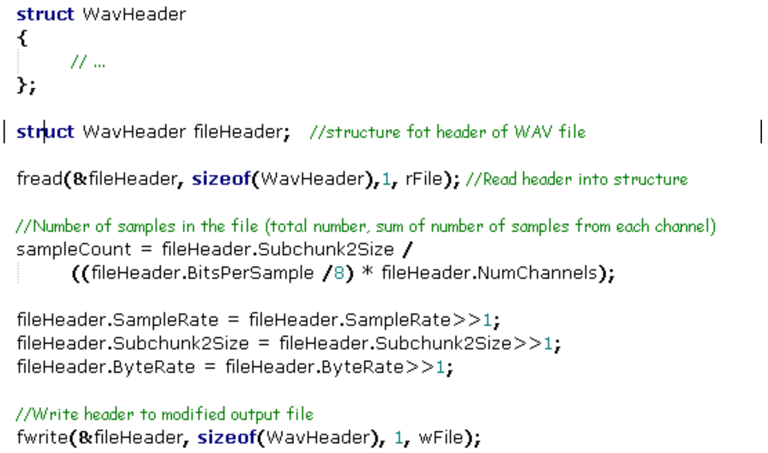
```
struct WavHeader
        {
        unsigned long  ChunkID ;     // the letters "RIFF" in ASCII form
        unsigned long ChunkSize;     // This is the size of the entire file in bytes
                // minus 8 bytes for the two fields not included in this count: ChunkID and ChunkSize.
        unsigned long Format;                //Contains the letters "WAVE"
        unsigned long Subchunk1ID;           //Contains the letters "fmt "
        unsigned long Subchunk1Size;         //16 for PCM
        unsigned short AudioFormat;          //PCM = 1 (i.e. Linear quantization)
                // Values other than 1 indicate some form of compression.
        unsigned short NumChannels;          //Mono = 1, Stereo = 2, etc.
        unsigned long SampleRate;            //8000, 44100, etc.
        unsigned long ByteRate;      //SampleRate * NumChannels * BitsPerSample/8
        unsigned short BlockAlign;   //NumChannels * BitsPerSample/8
        unsigned short BitsPerSample;        //    8 bits = 8, 16 bits = 16, etc.
        unsigned long Subchunk2ID;    //  Contains the letters "data"
        unsigned long Subchunk2Size;  //NumSamples * NumChannels * BitsPerSample/8
        };
```



File offset (bytes)	field name	Field Size (bytes)
0	ChunkID	4
4	ChunkSize	4
8	Format	4
12	Subchunk1ID	4
16	Subchunk1Size	4
20	AudioFormat	2
22	NumChannels	2
24	SampleRate	4
28	ByteRate	4
32	BlockAlign	2
34	BitsPerSample	2
36	Subchunk2ID	4
40	Subchunk2Size	4
44	data	Subchunk2Size

Reading formatted files #3

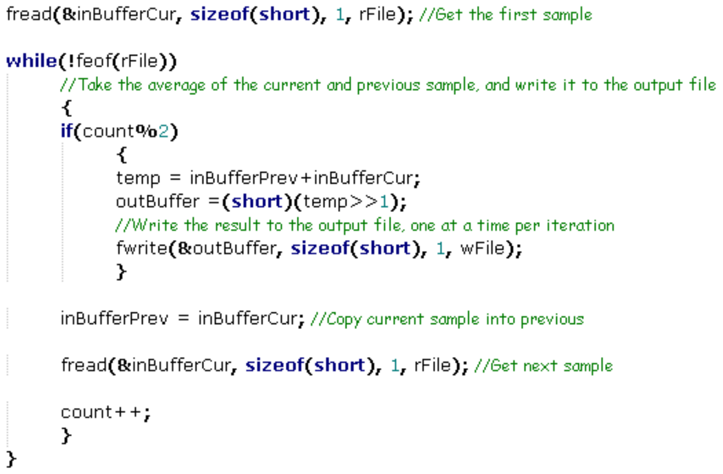
- … and use structures



```
struct WavHeader
{
        // ...
};

struct WavHeader fileHeader;   //structure fot header of WAV file

fread(&fileHeader, sizeof(WavHeader),1, rFile); //Read header into structure

//Number of samples in the file (total number, sum of number of samples from each channel)
sampleCount = fileHeader.Subchunk2Size /
        ((fileHeader.BitsPerSample /8) * fileHeader.NumChannels);

fileHeader.SampleRate = fileHeader.SampleRate>>1;
fileHeader.Subchunk2Size = fileHeader.Subchunk2Size>>1;
fileHeader.ByteRate = fileHeader.ByteRate>>1;

//Write header to modified output file
fwrite(&fileHeader, sizeof(WavHeader), 1, wFile);
```

Processing

- … sample by sample
  - example: mono/16 bit



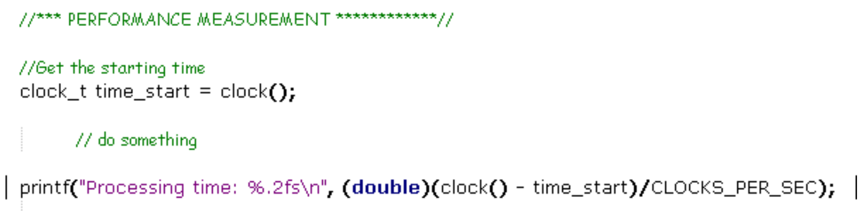
```
fread(&inBufferCur, sizeof(short), 1, rFile); //Get the first sample

while(!feof(rFile))
        //Take the average of the current and previous sample, and write it to the output file
        {
        if(count%2)
                {
                temp = inBufferPrev+inBufferCur;
                outBuffer =(short)(temp>>1);
                //Write the result to the output file, one at a time per iteration
                fwrite(&outBuffer, sizeof(short), 1, wFile);
                }

        inBufferPrev = inBufferCur; //Copy current sample into previous

        fread(&inBufferCur, sizeof(short), 1, rFile); //Get next sample

        count++;
        }
}
```

“Noise”

- ❑ in assignment
  - ❑ sine wave
  - ❑ random noise

Performance measurement

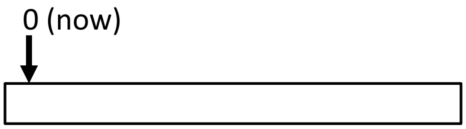
- ❑ profile critical sections of the code

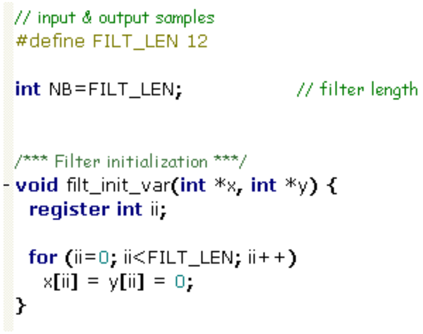


```
//*** PERFORMANCE MEASUREMENT ************//

//Get the starting time
clock_t time_start = clock();

        // do something

printf("Processing time: %.2fs\n", (double)(clock() - time_start)/CLOCKS_PER_SEC);
```

Filtering

- Init

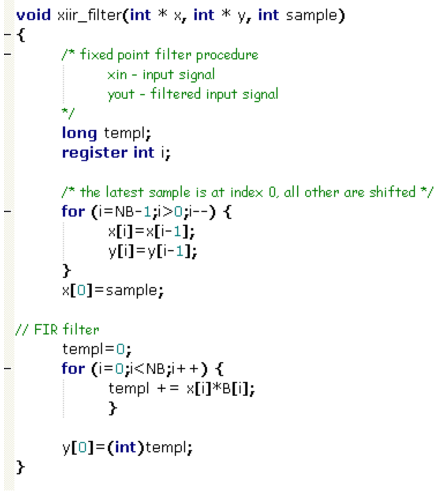




```
// input & output samples
#define FILT_LEN 12

int NB=FILT_LEN;                // filter length


/*** Filter initialization ***/
void filt_init_var(int *x, int *y) {
  register int ii;

  for (ii=0; ii<FILT_LEN; ii++)
    x[ii] = y[ii] = 0;
}
```

Filtering

- FIR filter (floating point)



```
void xiir_filter(int * x, int * y, int sample)
{
        /* fixed point filter procedure
                xin - input signal
                yout - filtered input signal
        */
        long templ;
        register int i;

        /* the latest sample is at index 0, all other are shifted */
        for (i=NB-1;i>0;i--) {
                x[i]=x[i-1];
                y[i]=y[i-1];
        }
        x[0]=sample;

// FIR filter
        templ=0;
        for (i=0;i<NB;i++) {
                templ += x[i]*B[i];
                }

        y[0]=(int)templ;
}
```