

**Goal:****Manipulation of WAV audio files:**

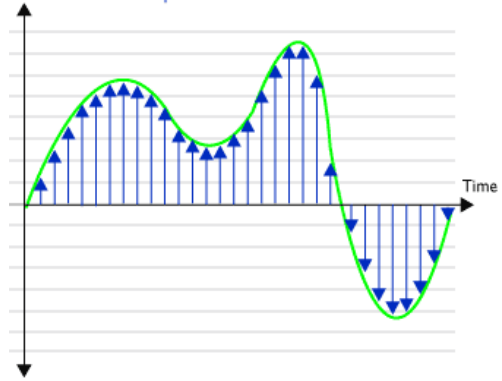
(1) Implement a function to adjust the tone (bass vs. treble) of an audio recording.

**Objective:**

Familiarity with the WAV audio library (libwav.a).

**Background:**

Waveform Audio File Format (more commonly known as WAV due to its filename extension) is a Microsoft file format standard for storing an audio recording. WAV files store a digital representation of sound by measuring the signal amplitude at regular intervals and recording the sampled value as an integer. Two basic properties determine quality: the sampling rate, which is the number of times per second that samples are taken; and the bit depth, which determines the number of possible digital values that can be used to represent each sample. This assignment uses single-channel (monophonic) WAV files that store sampled audio as a one-dimensional array of 16-bit signed integers.

**Download:**

Download and unpack file lab8.zip from Camino. It contains a partially completed program (main8.c), a pre-compiled library file (libwav.a) for manipulating WAV audio files, and an associated include file (wav.h).

**WAV Library:** Audio recordings are stored in memory using the following data structure:

```
typedef int16_t SAMPLE ;

typedef struct
{
    unsigned    sample_rate ;
    unsigned    num_samples ;
    SAMPLE      samples[] ;
} AUDIO ;
```

The library file (libwav.a) provides several functions needed for the manipulation of WAV files:

```
AUDIO *NewAudio(unsigned samples, unsigned rate) ;
```

Returns a pointer to memory allocated to hold an audio recording in which the number of samples is specified by parameter *samples* and sampled a number of times per second specified by parameter *rate*. The actual sample values, however, are not initialized and must be replaced by the user.

```
AUDIO *ReadWAV16(char *filespec) ;
```

Allocates memory to hold an image and fills it from a 16-bit monophonic WAV file. Parameter *filespec* is the name of the file given as a character string. The return value is a pointer to the audio in memory and must be used as an argument to all audio manipulation functions.

```
void WriteWAV16(char *filespec, AUDIO *audio) ;
```

Stores an audio recording from memory into a 16-bit monophonic WAV file. Parameter *filespec* is the filename specified as a character string.

```
void FreeAudio(AUDIO *audio) ;
```

Releases the memory used by an audio recording.

```
AUDIO *CopySegment(AUDIO *source, unsigned frstindex, unsigned lastindex) ;
```

Copies an audio segment from *source* starting at index position *frstindex*, through and including index position *lastindex*. Returns a pointer to a new memory representation of the copy.

```
AUDIO *InsertSegment(AUDIO *target, AUDIO *segment, unsigned at) ;
```

Inserts an audio segment specified by *segment* into the memory representation of *target* beginning at the index position specified by parameter *at*. Returns a pointer to the modified (lengthened) memory representation of *target*.

```
AUDIO *DeleteSegment(AUDIO *source, unsigned frstindex, unsigned lastindex) ;
```

Deletes an audio segment from *source* between index positions *frstindex* through *lastindex*. Returns a pointer to the modified (shortened) memory representation of *source*.

**Assignment:** You are to complete the source code for the following function that is located within the provided main program (main8.c):

```
AUDIO *AdjustTone(AUDIO *audio, unsigned percent_bass, unsigned percent_treble) ;
```

Adjusts the tone of an audio recording by adjusting the relative amount of bass and treble. Parameters *percent\_bass* and *percent\_treble* are unsigned integers in the range 0 to 100.

The apparent amount of *bass* (low frequencies) in a recording can be increased by using a running weighted average to smooth out the audio waveform and the apparent amount of *treble* (high frequencies) can be increased using the difference between successive samples. The pseudo-code shown below computes values for the bass and treble components and then mixes them with the original sample to produce the new sample value.

```

sample_avg ← 0.9 × sample_avg + 0.1 × orig_sample
sample_diff ← (orig_sample - prev_sample)

bass_part ← sample_avg × (percent_bass / 100)
orig_part ← orig_sample × (100 - percent_bass - percent_treb) / 100
treb_part ← sample_diff × (percent_treb / 100)

prev_sample ← orig_sample

curr_sample ← 2 × bass_part + orig_part + 2 × treb_part

```

Note: The values of *sample\_avg* and *prev\_sample* should be initialized to 0.

**Compilation:** Compile and link your program using the following command line:

**gcc -o lab8 main8.c -L. -lwav**

**Execution:** Execute your program using the following command syntax:

**./lab8 src-file dst-file tone-knob-degrees**

where *tone-knob-degrees* is an integer between -90 (full bass) and +90 (full treble); the program uses this value to compute values for *percent\_bass* and *percent\_treb*.

**When Done:** Demonstrate proper operation of your program to the teaching assistant and upload the completed source code for file *main8.c* to the lab drop box on Camino. Do not upload any other files.