

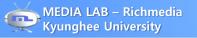
DSP Lab. Week 4 My Audio

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Media Lab. Rm567

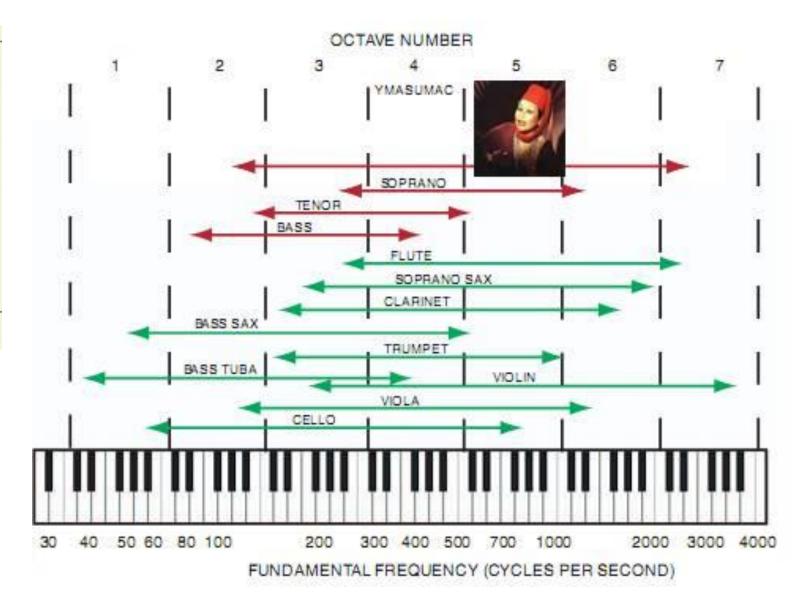
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Last update: September 2, 2019



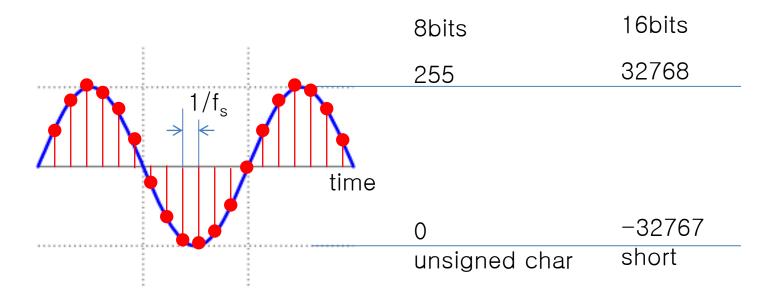
Frequency

```
1, A4(라)
                 = 440 Hz <-
2 A4#
         = 440*2^(1/12) = 466,1 Hz
3, B4(\lambda) = 440*2^{(2/12)} = 493,8 Hz
4, C5(\Xi) = 440*2^{(3/12)} = 523,25 Hz
         = 440*2^(4/12) = 554,36 Hz
6, D5(引) = 440*2^(5/12) = 587, 33 Hz
7, D5#
         = 440*2^(6/12) = 622,25 Hz
8, E5(0) = 440*2^(7/12) = 659,26 Hz
9, F5(II) = 440*2^(8/12) = 698, 46 Hz
10, F5#
        = 440*2^(9/12) = 739 99 Hz
11, G5(金) = 440*2^(10/12) = 784,00 Hz
12, G5# = 440*2^(11/12) = 830, 60 Hz
A5((라) = 440*2 = 880 Hz
```



ADC (Analog-to-Digital Conversion)

ADC = sampling + quantization Sampling = continuous to discrete, $2f_m < f_s$ Quantization = analog to digital, 2^B levels Range



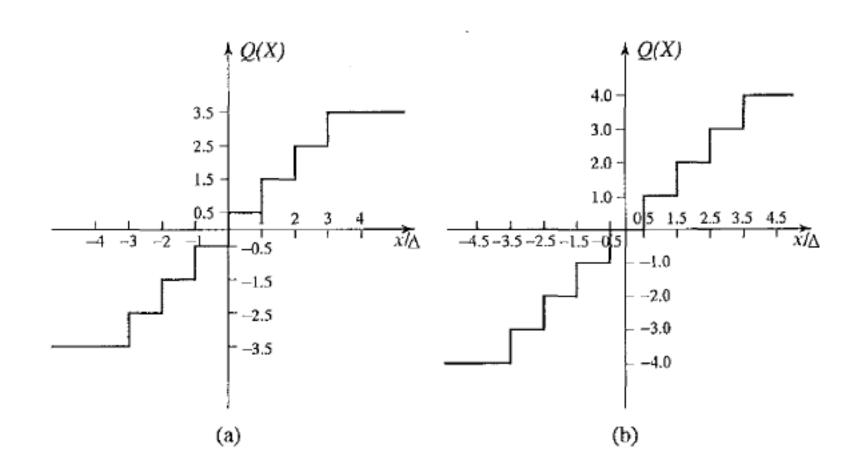
- ❖ To efficiently represent the source output, we have to reduce the number of distinct values to a much smaller set.
 - Uniform scalar quantization
 - Non-uniform scalar quantization
 - Vector quantizatoin

Uniform scalar quantization

- A uniform scalar quantizer partitions the domain of input values into equally spaced intervals, except possibly at the two outer intervals.
- The endpoints of partition intervals are called the quantizer's decision boundaries.
- The output or reconstruction value corresponding to each interval is taken to be the midpoint of the interval.
- The length of each interval is referred to as the step size, denoted by the symbol Δ .
- Uniform scalar quantizers are of two types: midrise and midtread
- In case of $\Delta=1$,

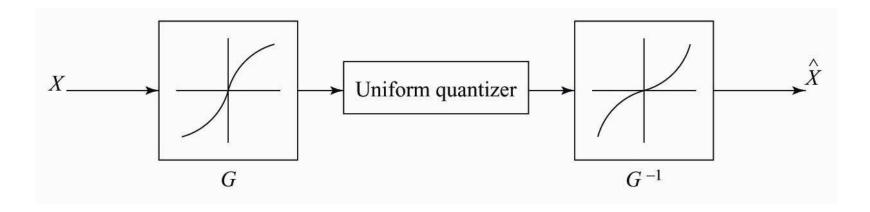
$$Q_{midrise}(x) = \lceil x \rceil - 0.5 \tag{8.4}$$

$$Q_{midtread}(x) = \lfloor x + 0.5 \rfloor \tag{8.5}$$



Companded quantizer

- the input is mapped by a compressor function G and then quantized using a uniform quantizer.
- After transmission, the quantized values are mapped back using an expanded function G⁻¹.

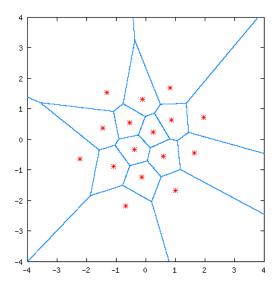


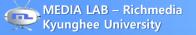
Vector quantization

■ Any compression system performs better if it operates on vectors or groups of samples rather than on individual symbols or samples.



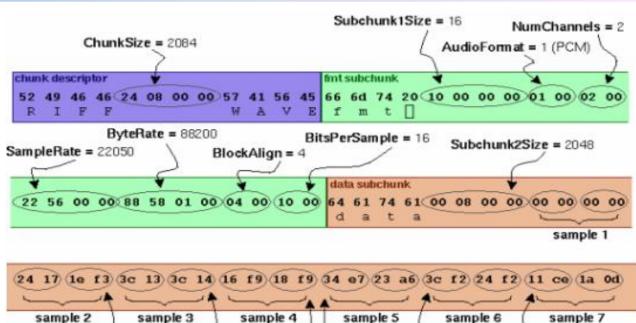
- n-component code vector represents vectors that lie within a region in n-dimensional space.
- A collection of these code vectors forms the codebook for the vector quantizer.





Wave File Header (44 bytes long)

```
#define WORD unsigned short
                                                                         ByteRate = 88200
#define DWORD unsigned int
                                                              SampleRate = 22050
void writeHeader(char *WaveFile, int nn)
          // RIFF chunk
          strcpv(WaveFile,"RIFF");
          *(DWORD *)(WaveFile+4) = nn-8;
          strcpy(WaveFile+8,"WAVE");
                                                                 sample 2
                                                                            sample 3
          // fmt chunk
          strcpy(WaveFile+12,"fmt"); WaveFile[15] = ' ';
          *(DWORD *)(WaveFile+16) = 16; // cksize
                                                                         right channel samples
          *(WORD *)(WaveFile+20) = 1; // wFormatTag PCM 1
          *(WORD *)(WaveFile+22) = 1; // nChannels
                                                                        mono 1 stereo 2
          *(DWORD *)(WaveFile+24) = 8000; // nSamplesPerSec
                                                                 8000Hz or 44100Hz
          *(DWORD *)(WaveFile+28) = 8000; // nAvgBytesPerSec
          *(WORD *)(WaveFile+32) = 1; // nBlockAlign = nChannels*wBitsPerSample/8
          *(WORD *)(WaveFile+34) = 8;
                                           // wBitsPerSample
                                                                       8bits or 16bits
          // data chunk
          strcpy(WaveFile+36,"data");
          *(DWORD *)(WaveFile+40) = nn;
                                           // cksize
          return;
```



left channel samples

Read header of a wave file

```
#include <stdio.h>
#include <string.h>
#include <stdlib.h>
#define WORD unsigned short
#define DWORD unsigned int
void main(){
// 1. Read the image file.
    char f[44];
   FILE *fff;
    if((fff = fopen("Beatles-LetItBe-wav.wav","rb")) == NULL){ printf("cant open Beatles-LetItBe-wav.wav.\n"); exit(123);}
// 2. Read the first 44 bytes
    if((fread(f,1,44,fff) != 44)){ printf("cant read Beatles-LetItBe-wav.wav..\n"); exit(444);
    fclose(fff);
// 3. Read all fields.
                  %c%c%c\n",f[0],f[1],f[2],f[3]);
    printf("RIFF
    printf("filesize %d\n",*(DWORD*)(f+4)); // sub-chunk 2 size
                   %c%c%c%c\n",f[8],f[9],f[10],f[11]);
    printf("WAVE
    printf("cksize
                   %d\n",*(DWORD*)(f+16));
    printf("channels %d\n",*(WORD*)(f+22));
                     %d\n",*(DWORD*)(f+24));
    printf("fs
    printf("bytes/s %d\n",*(DWORD*)(f+28));
    printf("bits/sam %d\n",*(WORD*)(f+34));
    printf("data
                     %c%c%c%n",f[36],f[37],f[38],f[39]);
    printf("cksize
                     %d\n",*(DWORD*)(f+40)); // sub-chunk 2 size
    delete f;
    getchar();
```

```
#include <stdio.h>
#include <string.h>
#include <stdlib.h>
#define WORD unsigned short
#define DWORD unsigned int
#define NN 4096
#define N 1024
∃void main()
                                                                                    // 4. modify audio signal to mono
// 1. Read the image file.
    char *f;
    f = new char[48];
                                                                                           if((fread(alldata,4,N,fff) != N)){
    FILE *fff, *fout, *foutr;
    if((fff = fopen("Beatles-LetItBe-way.way", "rb")) == NULL){
                                                                                               exit(244);
         printf("cant open Beatles-LetItBe-wav.wav.\m");
                                                                                           }
         exit(123);
                                                                                           // store left/right data
                                                                                           int ii.iii;
// 2. Read the first 48 bytes => write header
                                                                                           for(ii=iii=0;ii<N;ii++,iii+=2){
    if((fread(f,1,48,fff) != 48)){
                                                                                               leftdata[ii] = alldata[iii];
         printf("cant read Beatles-LetItBe-wav.wav..\"n");
                                                                                               rightdata[ii] = alldata[iii+1];
         exit(444);
                                                                                           // modify left/right data
    if((fout = fopen("Beatles-LetItBe-xxx.wav", "wb")) == NULL){
         printf("cant open Beatles-LetItBe-xxx.wav.\n");
         exit(123);
                                                                                           // .....
                                                                                           // restore left/right data
// 3. Read all fields.
                                                                                           for(ii=iii=0;ii<N;ii++,iii+=2){
    printf("RIFF
                      %c%c%c%c\mu".f[0].f[1].f[2].f[3]);
                                                                                               alldata[iii] = leftdata[ii];
    printf("filesize %d\n", *(D\WORD*)(f+4)); // sub-chunk 2 size
                                                                                               alldata[iii+1] = rightdata[ii];
    printf("WAVE
                      %c%c%c%c\\n",f[8],f[9],f[10],f[11]);
    printf("cksize %dMn",*(DWORD*)(f+16));
                                                                                    // 5. write on outut file
    printf("channels %d\n", *(WORD*)(f+22));
                                                                                           if((fwrite(alldata,4,N,fout) != N)){
    printf("fs
                      %d\n", * (D\WORD*) (f+24));
    printf("bytes/s %dMn", *(DWORD*)(f+28));
                                                                                               exit(246);
    printf("bits/sam %d\n", *(WORD*)(f+34));
                      %c%c%c%c\mu",f[36],f[37],f[38],f[39]);
    printf("data
                                                                                       } // all data
    printf("cksize %c\m", *(DWORD*)(f+40)); // sub-chunk 2 size
                                                                                       fclose(fff);
    DWORD FileSize = *(DWORD*)(f+40);
                                                                                       fclose(fout);
     if((fwrite(f,1,48,fout) != 48)){
                                                                                       delete f;
         printf("cant write header on Beatles-LetItBe-xxx.wav..\\n");
         exit(444);
    FileSize /= 4;
                        // 4bytes per sample
```

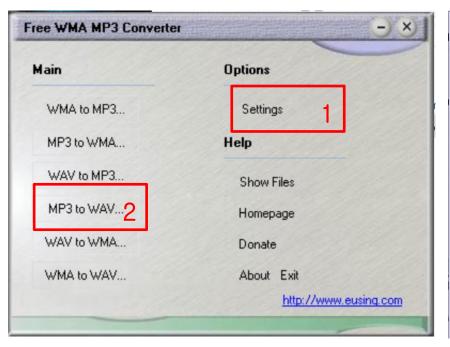
```
short leftdata[N],rightdata[N],alldata[N*2];
for(int n = 0; n < FileSize; n += N){ // all data
        printf("cant read %d-th data from Beatles-LetItBe-way.way..\m",n);
   // 이 부분이 여러분들이 프로그램할 자리.... Good luck!!
       printf("cant write %d-th data on Beatles-LetItBe-xxx.wav..\m",n);
```

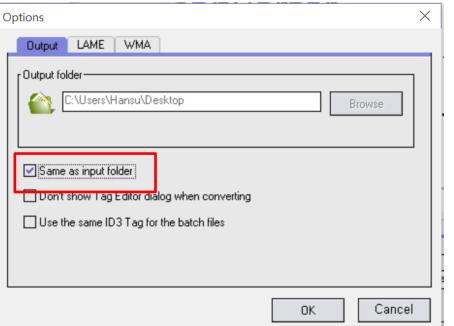
MP3 to WAV Converter

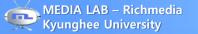
MP3 to wav converter "FWMCSetup.zip"

Free WMA MP3 Converter를 설치 후 실행한다.

- 그림1 과 같은 화면이 나오면 1. Settings을 누른다.
- 그림2 와 같은 화면이 나오면 Same as input folder를 체크!
- 그림1 에서 2. MP3 to WAV 클릭 후 convert할 음원 선택!







Week 4 assignment

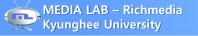
Let It Be 음악에, 중간에 산토끼를 넣어라.

```
(ex) 첫 '솔'은 0.5초, f<sub>s</sub>=44100이면, f<sub>max</sub>=22050
dt = 1/44100.0;
for(t=0; t<0.5; t+=dt){
 *(leftdata++) = *(rightdata++) = (short)(20000.0*sin(2*PI*784*t));
}
```

보통빠르게



```
1, A4(라)
                             = 440 Hz <-
2 A4#
           = 440 \times 2^{(1/12)} = 466, 1 \text{ Hz}
3.84(\text{A}) = 440 \times 2^{(2/12)} = 493.8 \text{ Hz}
4, C5(\Xi) = 440*2^{(3/12)} = 523,25 Hz
5, C5# = 440*2^(4/12) = 554.36 Hz
6, D5(1) = 440 \times 2^{5/12} = 587,33 Hz
7, D5# = 440 \times 2^{6} / 12 = 622, 25 \text{ Hz}
8, E5(\Box) = 440*2^(7/12) = 659,26 Hz
9, F5(\overline{H}) = 440 \times 2^{\circ}(8/12) = 698, 46 Hz
10, F5# = 440 \times 2^{\circ}(9/12) = 739.99 Hz
11, G5(金) = 440*2^(10/12) = 784,00 Hz
12, G5# = 440*2^(11/12) = 830, 60 Hz
A5((라) = 440*2
                             = 880 \, \text{Hz}
```



Week 4 assignment

"KLAS에 제출할 때 다음 사항을 꼭 지켜주세요"

- 1. 파일명: "Lab00_요일_대표자이름.zip"
- Ex) Lab01_목_홍길동.zip (압축 툴은 자유롭게 사용)
- 2. 제출 파일 (보고서와 프로그램을 압축해서 제출)
 - 보고서 파일 (hwp, word): 이름, 학번, 목적, 변수, 알고리즘(순서), 결과 분석, 느낀 점
 - 프로그램

DSP 실험 보고서

과제 번호	Lab01	제출일	2019.09.02
학번/이름	200000000 홍길동		
		200000000 푸리에	

1. 목적	
2. 변수	
3. 알고리즘	
4. 결과분석	
5. 느낀 점	

