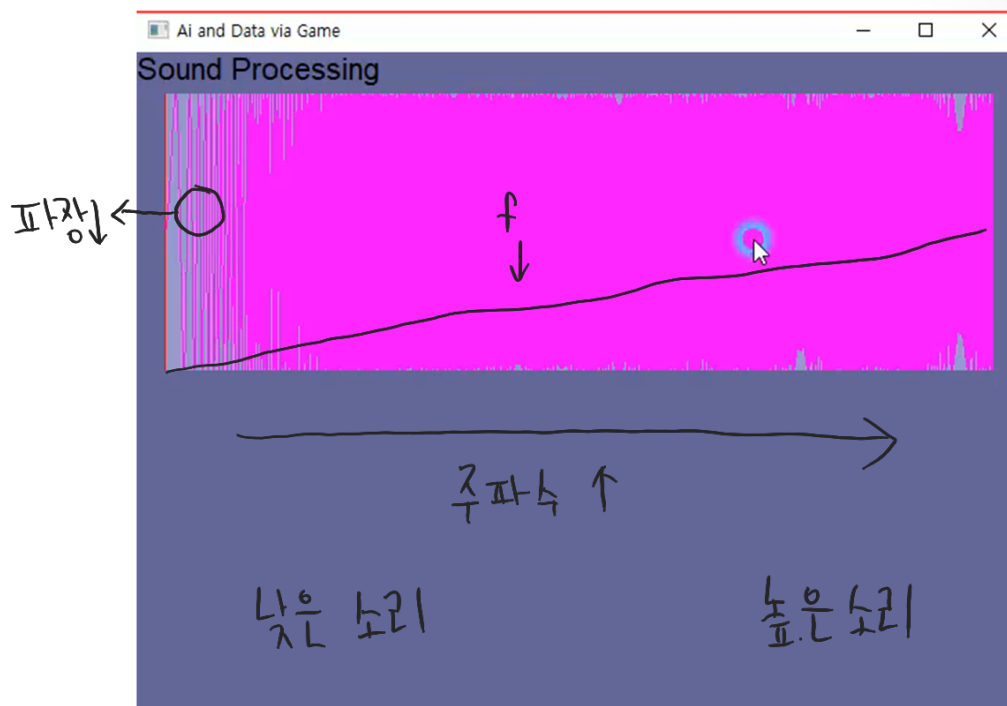


## 5. Sound Processing



# Time and frequency domain

- 정현파(sinusoids), 주기(period), 주파수(frequency)

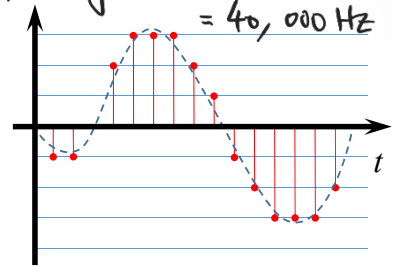
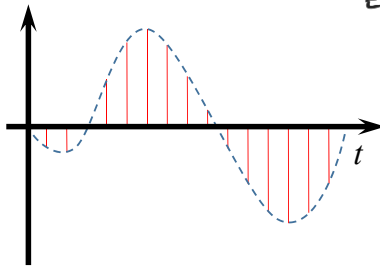
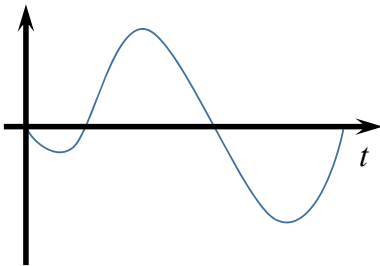
$$x(t) = X_M \sin \omega t$$

$$x(\omega(t+T)) = x(\omega t)$$

$$f = \frac{1}{T}$$

Hz      1/s

- 표본화 주파수(sampling frequency) 샘플링 진동수 사람의 가청 주파수 20,000 Hz  
 ↓  
 표본화 (sampling) → 양자화 (quantization) → 부호화 (encoding)  
 일반적으로 signed int      샘플링 진동수 = 40,000 Hz



Time Domain에서의 주기 신호를 기본적인 주기 함수의 합으로 나타냄 (signal) sin, cos

## Fourier Series

$\frac{\pi}{2}$  만큼 차이 남

$$e^{j\theta} = \underbrace{\cos \theta}_{\text{실수부}} + j \underbrace{\sin \theta}_{\text{허수부}}$$

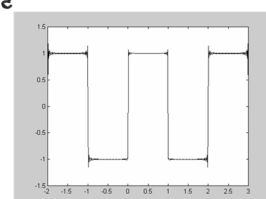
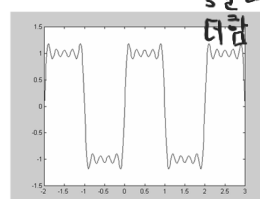
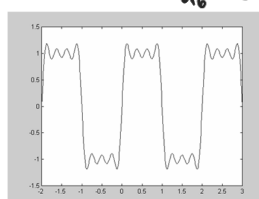
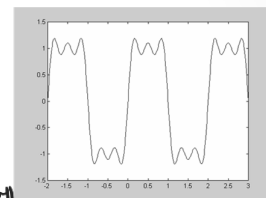
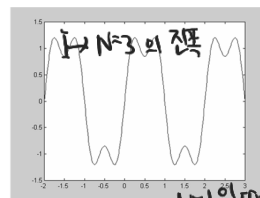
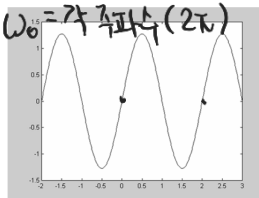
주기함수를 기본 주기함의 합으로 나타냄

$$f(t) = \sum_{k=-\infty}^{\infty} a_k e^{jk\omega_0 t}$$

$$a_k = \frac{1}{T_0} \int_{T_0} f(t) e^{-jk\omega_0 t} dt \rightarrow \text{주기함수의 성분 (여가 계수로 들어감)}$$

$$f_a(t) = \sum_{k=-N}^N a_k e^{jk\omega_0 t}$$

$k\omega_0 = f(t)$ 의 주기  $K = \{R \mid R = \{-\infty, \infty\}\}$  N=1일 때에 3일 때를 더함  
 $\omega_0 = \text{각 주파수}(2\pi)$  N=99  
N=2n+1일 때 N은 0.

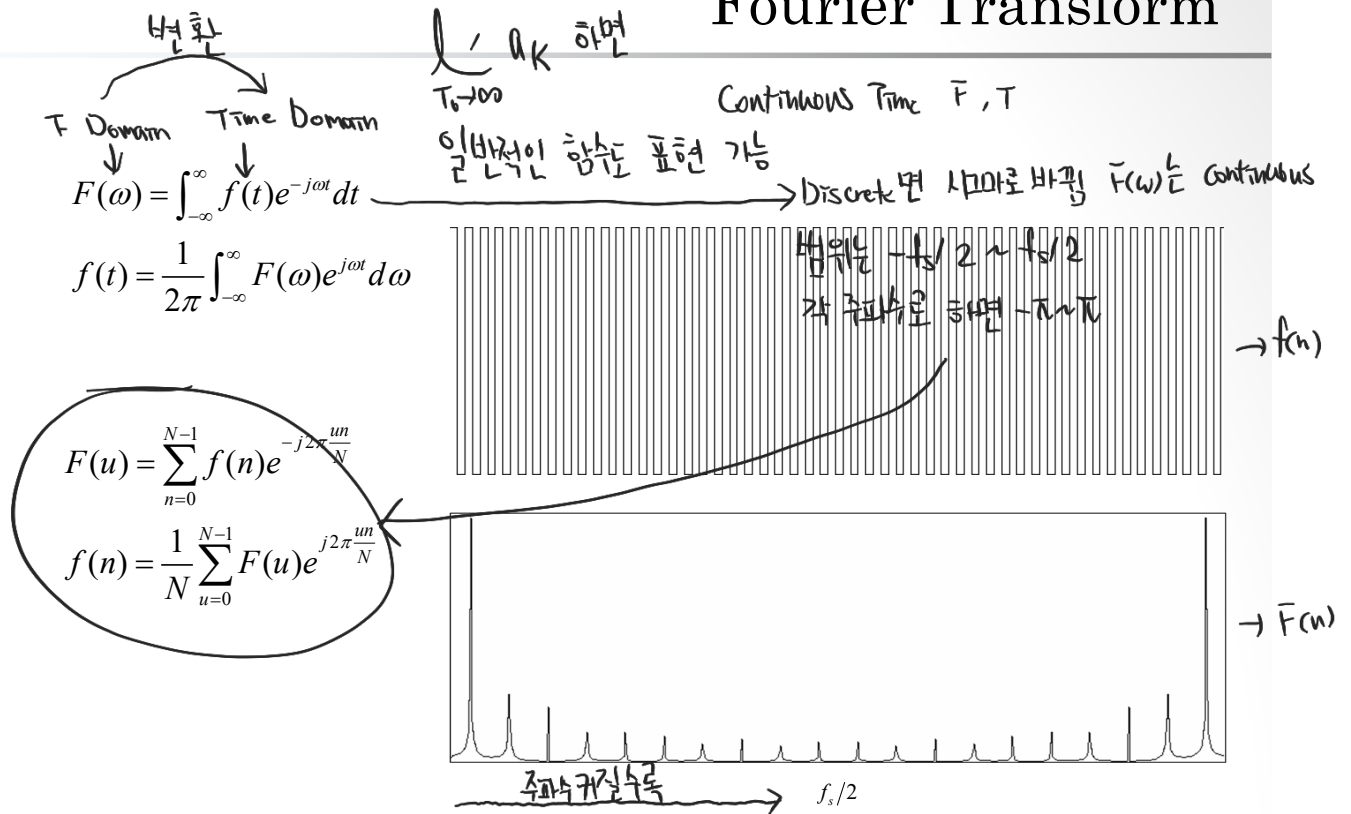


→ 결론적으로 대역폭이

$f(t)$ 가 주기함수일 때로 한정되어 있음

주기 함수를 주할 수 있음

# Fourier Transform



Discrete 일때 010101...  
 $\rightarrow$  최대 주파수 = Sampling Frequency / 2

# Fourier Transform Properties

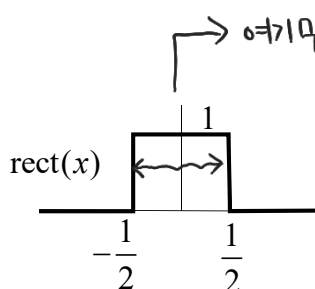
$x(t) \leftrightarrow X(\omega), y(t) \leftrightarrow Y(\omega)$  Convolution

$x(t - t_0) \leftrightarrow e^{-j\omega t_0} X(\omega)$

$e^{j\omega t_0} x(t) \leftrightarrow X(\omega - \omega_0)$

Time  $\leftrightarrow$  Frequency  
 $x(t)y(t) \leftrightarrow X(\omega) * Y(\omega)$

Convolution  
 $x(t) * y(t) \leftrightarrow \frac{1}{2\pi} X(\omega) Y(\omega)$

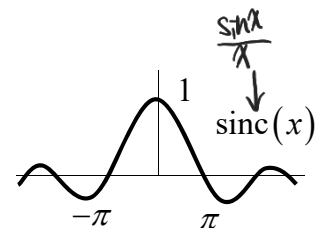


$\delta(t) \leftrightarrow 1$

$1 \leftrightarrow 2\pi\delta(\omega)$

F.T  
 $\text{rect}(t/\tau) \leftrightarrow \tau \text{sinc}\left(\frac{\omega\tau}{2}\right)$

$\frac{W}{\pi} \text{sinc}(Wt) \leftrightarrow \text{rect}\left(\frac{\omega}{2W}\right)$



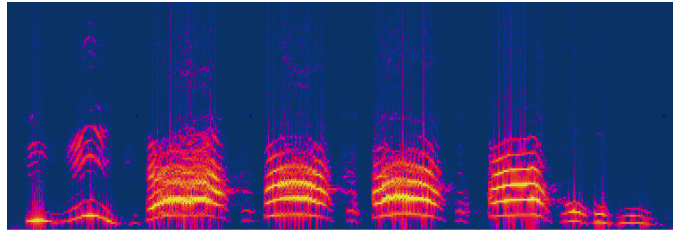
# Spectrogram

Short-time Fourier transform  
Windowed Fourier transform

||  
STFT



↳ 부분적인 F.T. 복수를 크기와 각도로 표현



↳ 가짜 색상

## FFT I

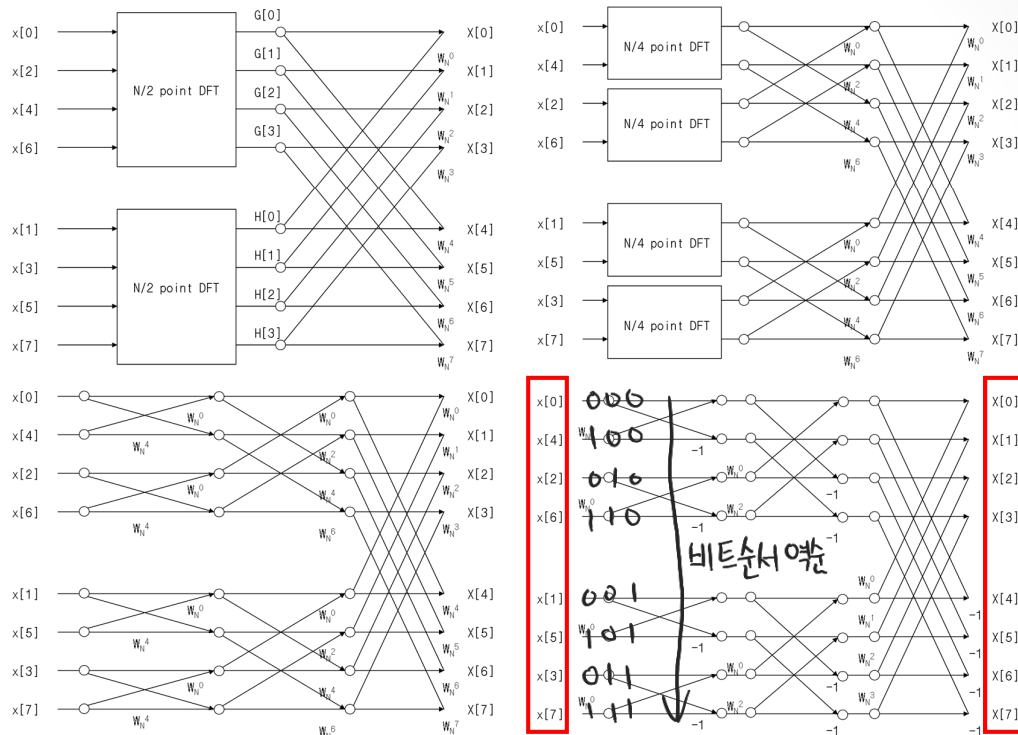
그냥 F.T.는 너무 오래 걸림.

$$\begin{aligned}
 F(u) &= \sum_{n=0}^{N-1} f(n) e^{-j2\pi \frac{un}{N}} = \sum_{n=0}^{N-1} f(n) W_N^{un} \\
 &= \sum_{n=\text{even}} f(n) W_N^{un} + \sum_{n=\text{odd}} f(n) W_N^{un} \\
 &= \sum_{r=0}^{N/2-1} f(2r) W_N^{2ru} + \sum_{r=0}^{N/2-1} f(2r+1) W_N^{(2r+1)u} \\
 &= \sum_{r=0}^{N/2-1} f(2r) (W_N^2)^{ru} + W_N^u \sum_{r=0}^{N/2-1} f(2r+1) (W_N^2)^{ru} \\
 &= \sum_{r=0}^{N/2-1} f(2r) W_{N/2}^{ur} + W_N^u \sum_{r=0}^{N/2-1} f(2r+1) W_{N/2}^{ur}
 \end{aligned}$$

짝수 데이터 FT      홀수 데이터 FT

한번만 쪼개는 게 아니라 계속 쪼갬

# FFT II



$$N^2 \rightarrow N \log_2 N$$

# FFT III

```
void FFT2Radix(double *Xr, double *Xi, double *Yr, double *Yi,
int nN, bool bInverse){
    int i, j, k;
    double T, Wr, Wi;

    if(nN <= 1) return;
    for(i = 0 ; i < nN ; i++) {
        Yr[i] = Xr[i];
        Yi[i] = Xi[i];
    }

    j = 0;
    for (i = 1 ; i < (nN-1) ; i++) {
        k = nN/2;
        while(k <= j) {
            j = j - k;
            k = k/2;
        }
        if (i < j) {
            T = Yr[j];
            Yr[j] = Yr[i];
            Yr[i] = T;
            T = Yi[j];
            Yi[j] = Yi[i];
            Yi[i] = T;
        }
    }

    double Tr, Ti;
    int iter, j2, pos;
    k = nN >> 1;
    iter = 1;
```

16/2=8 <= j

0000				
1000	16/2			
0100	8-8	+0100		
1100	+1000			
0010	12-8	4-4	+0010	
1010	+1000			
0110	10-8	+0100		
1110	+1000			
0001	14-8	6-4	2-2	+0001
1001	+1000			
0101	9-8	+0100		
1101	+1000			
0011	13-8	5-4	+0010	
1011	+1000			
0111	11-8	+0100		
1111	+1000			

# FFT IV

```

while(k > 0) {
    j = 0;
    j2 = 0;
    for(i = 0 ; i < nN >> 1 ; i++) {
        Wr = cos(2.*Pi*(j2*k)/nN);
        if(bInverse == 0)
            Wi = -sin(2.*Pi*(j2*k)/nN);
        else
            Wi = sin(2.*Pi*(j2*k)/nN);

        pos = j+(1 << (iter-1));
        Tr = Yr[pos] * Wr - Yi[pos] * Wi;
        Ti = Yr[pos] * Wi + Yi[pos] * Wr;

        Yr[pos] = Yr[j] - Tr;
        Yi[pos] = Yi[j] - Ti;

        Yr[j] += Tr;
        Yi[j] += Ti;
        j += 1 << iter;
        if(j >= nN) j = ++j2;
    }
    k >>= 1;
    iter++;
}

```

①// iter: (1), (2), (3), ...

③

⑤// j2: (0, 0, 0, ...), (0, 0, 0, ..., 1, 1, 1, ...), ...

//  $j2*k/nN = j2*nN/2^{iter}/nN$

// (j+1), (j+2), (j+4), ...

// Tr

// Ti

//  $x[j] - T$

//  $x[j] = x[j] + T$

④// (0, 2, 4, ...), (0, 4, 8, ..., 1, 5, ...), ...

// +j2

②// nN/2, nN/4, nN/8, nN/16

# FFT V

```

if(bInverse){
    for(i = 0 ; i < nN ; i++) {
        Yr[i] /= nN;
        Yi[i] /= nN;
    }
}
}

```

```
void CKhuGleSignal::MakeSpectrogram() {
    if(!m_Real) m_Real = dmatrix(m_nFrequencySampleLength, m_nWindowSize);
    if(!m_Imaginary) m_Imaginary
        = dmatrix(m_nFrequencySampleLength, m_nWindowSize);

    double *OrgReal = new double[m_nWindowSize];
    double *OrgImaginary = new double[m_nWindowSize];

    for(int t = 0 ; t < m_nFrequencySampleLength ; t++) {
        int OrgT = t*m_nSampleLength/m_nFrequencySampleLength;
        for(int dt = 0 ; dt < m_nWindowSize ; dt++) {
            int tt = OrgT+dt-m_nWindowSize/2;
            if(tt >= 0 && tt < m_nSampleLength)
                OrgReal[dt] = m_Samples[tt];
            else
                OrgReal[dt] = 0;
            OrgImaginary[dt] = 0;
        }

        FFT2Radix(OrgReal, OrgImaginary, m_Real[t], m_Imaginary[t],
            m_nWindowSize, false);
    }
    delete [] OrgReal;                delete [] OrgImaginary;
}
```

## Wave & bitmap files

## KhuGleSignal.h (1)

```
#pragma once
typedef struct tagWAV_HEADER_ {
    ...
} WAV_HEADER_;
typedef struct tagCHUCK_ {
    ...
} CHUCK_;
#pragma pack(push, 1) → 1byte 단위로 조개서 붙여라
typedef struct tagBITMAPFILEHEADER_ {
    ...
} BITMAPFILEHEADER_;
typedef struct tagBITMAPINFOHEADER_ {
    ...
} BITMAPINFOHEADER_;
typedef struct tagRGBQUAD_ {
    ...
} RGBQUAD_;
#pragma pack(pop)
#define BI_RGB_ 0L
```

## KhuGleSignal.h (2)

```
class CKhuGleSignal {
public:
    short int *m_Samples;
    int m_nSampleRate;
    int m_nSampleLength;

    double **m_Real, **m_Imaginary;
    int m_nWindowSize;
    int m_nFrequencySampleLength;    // Spectrogram에서 계산한 sample 수
                                     m_Real = dmatrix(m_nFrequencySampleLength, m_nWindowSize);

    int m_nW, m_nH;
    unsigned char **m_Red, **m_Green, **m_Blue;

    CKhuGleSignal();
    ~CKhuGleSignal();

    void ReadWave(char *FileName);          bool SaveWave(char *FileName);
    void ReadBmp(char *FileName);           bool SaveBmp(char *FileName);

    void MakeSpectrogram();
};
```

## KhuGleSignal.cpp (1)

```
#include "KhuGleSignal.h"
#include "KhuGleBase.h"
#include <cstdio>

CKhuGleSignal::CKhuGleSignal() {
    m_Samples = nullptr;

    m_Real = nullptr;
    m_Imaginary = nullptr;

    m_nWindowSize = 256;
    m_nFrequencySampleLength = 1024;

    m_Red = m_Green = m_Blue = nullptr;
}
```



## KhuGleSignal.cpp (2)

```
CKhuGleSignal::~CKhuGleSignal() {
    if(m_Samples) delete [] m_Samples;
    if(m_Real) free_dmatrix(m_Real, m_nFrequencySampleLength,
        m_nWindowSize);
    if(m_Imaginary) free_dmatrix(m_Imaginary,
        m_nFrequencySampleLength, m_nWindowSize);

    if(m_Red) free_cmatrix(m_Red, m_nH, m_nW);
    if(m_Green) free_cmatrix(m_Green, m_nH, m_nW);
    if(m_Blue) free_cmatrix(m_Blue, m_nH, m_nW);
}
void CKhuGleSignal::ReadWave(char *FileName){}
bool CKhuGleSignal::SaveWave(char *FileName){}

void CKhuGleSignal::ReadBmp(char *FileName){}
bool CKhuGleSignal::SaveBmp(char *FileName){}

void CKhuGleSignal::MakeSpectrogram() {}
```

## Playing wave

## SoundPlayWin.cpp

```
#include <windows.h>
#include <mmsystem.h>
#pragma comment(lib, "Winmm.lib")

HWAVEOUT hPlay;
bool bSoundPlaying = false;
WAVEFORMATEX WaveFormat;
WAVEHDR WaveHdr;

void CALLBACK waveOutProc(HWAVEOUT hWO, UINT uMsg, DWORD dwInstance,
    DWORD dwParam1, DWORD dwParam2) {
    ...
}

void PlayWave(short int *Sound, int nSampleRate, int nLen) {
    ...
}

void StopWave() {
    ...
}

void GetPlaybackPosotion(unsigned long *pPosition) {
    ...
}
```

```
#include <windows.h>
#include "KhuGleBase.h"
#include "KhuGleSprite.h"
#include "KhuGleLayer.h"
#include "KhuGleScene.h"
#include "KhuGleComponent.h"

void PlayWave(short int *Sound, int nSampleRate, int nLen);
void StopWave();
void GetPlaybackPosotion(unsigned long *Rate);

...
```

```
class CKhuGleSoundLayer : public CKhuGleLayer {
public:
    CKhuGleSignal m_Sound;
    int m_nViewType;           // 0: time, 1: time-frequency,
                               // 2: time-frequency (log scale)

    CKhuGleSoundLayer(int nW, int nH, KgColor24 bgColor,
        CKgPoint ptPos = CKgPoint(0, 0));
    void DrawBackgroundImage();
};

CKhuGleSoundLayer::CKhuGleSoundLayer(int nW, int nH, KgColor24 bgColor,
    CKgPoint ptPos)
    : CKhuGleLayer(nW, nH, bgColor, ptPos) {

    m_nViewType = 0;
}
```

```
void CKhuGleSoundLayer:: DrawBackgroundImage() {
    for(int y = 0 ; y < m_nH ; y++)
        for(int x = 0 ; x < m_nW ; x++) {
            m_ImageBgR[y][x] = KgGetRed(m_bgColor);
            m_ImageBgG[y][x] = KgGetGreen(m_bgColor);
            m_ImageBgB[y][x] = KgGetBlue(m_bgColor);
        }
    if(m_nViewType == 0 && m_Sound.m_Samples) {
        int xx0, yy0, xx1, yy1;
        for(int i = 0 ; i < m_Sound.m_nSampleLength ; ++i) {
            xx1 = i*m_nW/m_Sound.m_nSampleLength;
            yy1 = m_nH-(m_Sound.m_Samples[i]+32768)*m_nH/65536-1;
            if(i > 0)
                CKhuGleSprite::DrawLine(m_ImageBgR, m_ImageBgG,
                    m_ImageBgB, m_nW, m_nH,
                    xx0, yy0, xx1, yy1, KG_COLOR_24_RGB(255, 0, 255));
            xx0 = xx1;
            yy0 = yy1;
        }
    }
}
```

```
if(m_nViewType == 1 && m_Sound.m_Real && m_Sound.m_Imaginary) {
    double Max = 0;
    for(int y = 0 ; y < m_nH ; y++)
        for(int x = 0 ; x < m_nW ; x++) {
            int yy = (m_nH-y-1)/2*m_Sound.m_nWindowSize/m_nH;
            int xx = x*m_Sound.m_nFrequencySampleLength/m_nW;

            double Magnitude = sqrt(m_Sound.m_Real[xx][yy]*m_Sound.m_Real[xx][yy] +
                m_Sound.m_Imaginary[xx][yy]*m_Sound.m_Imaginary[xx][yy]);
            if(Magnitude > Max) Max = Magnitude;
        }
}
```

```
for(int y = 0 ; y < m_nH ; y++)
    for(int x = 0 ; x < m_nW ; x++) {
        int yy = (m_nH-y-1)/2*m_Sound.m_nWindowSize/m_nH;
        int xx = x*m_Sound.m_nFrequencySampleLength/m_nW;

        m_ImageBgR[y][x] =
            (int) (sqrt(m_Sound.m_Real[xx][yy]*m_Sound.m_Real[xx][yy] +
                m_Sound.m_Imaginary[xx][yy]*m_Sound.m_Imaginary[xx][yy])*255/Max);
        m_ImageBgG[y][x] = m_ImageBgR[y][x];
        m_ImageBgB[y][x] = m_ImageBgR[y][x];
    }
}
```

```
if(m_nViewType == 2 && m_Sound.m_Real && m_Sound.m_Imaginary) {
    double Max = 0, Min = 0;
    for(int y = 0 ; y < m_nH ; y++)
        for(int x = 0 ; x < m_nW ; x++) {
            int yy = (m_nH-y-1)/2*m_Sound.m_nWindowSize/m_nH;
            int xx = x*m_Sound.m_nFrequencySampleLength/m_nW;

            double Magnitude = sqrt(m_Sound.m_Real[xx][yy]*m_Sound.m_Real[xx][yy] +
                m_Sound.m_Imaginary[xx][yy]*m_Sound.m_Imaginary[xx][yy]);
            Magnitude = 10*log10(Magnitude*Magnitude+1.);
            if(x == 0 && y == 0) {
                Min = Magnitude;
                Max = Magnitude;
            }

            if(Magnitude > Max) Max = Magnitude;
            if(Magnitude < Min) Min = Magnitude;
        }
}
```

```
for(int y = 0 ; y < m_nH ; y++)
    for(int x = 0 ; x < m_nW ; x++) {
        int yy = (m_nH-y-1)/2*m_Sound.m_nWindowSize/m_nH;
        int xx = x*m_Sound.m_nFrequencySampleLength/m_nW;

        double Magnitude = sqrt(m_Sound.m_Real[xx][yy]*m_Sound.m_Real[xx][yy] +
            m_Sound.m_Imaginary[xx][yy]*m_Sound.m_Imaginary[xx][yy]);
        Magnitude = 10*log10(Magnitude*Magnitude+1.);

        m_ImageBgR[y][x] = (int)((Magnitude-Min)*255/(Max-Min));
        m_ImageBgG[y][x] = m_ImageBgR[y][x];
        m_ImageBgB[y][x] = m_ImageBgR[y][x];
    }
}
```

```
class CSoundProcessing : public CKhuGleWin {
public:
    CKhuGleSoundLayer *m_pSoundLayer;
    CKhuGleSprite *m_pSoundLine;

    CSoundProcessing(int nW, int nH, char *SoundPath);
    void Update();
};
```

```

CSoundProcessing::CSoundProcessing(int nW, int nH, char *SoundPath)
: CKhuGleWin(nW, nH) {
    m_pScene = new CKhuGleScene(640, 480, KG_COLOR_24_RGB(100, 100, 150));

    m_pSoundLayer = new CKhuGleSoundLayer(600, 200,
        KG_COLOR_24_RGB(150, 150, 200), CKgPoint(20, 30));
    m_pSoundLayer->m_Sound.ReadWave(SoundPath);
    m_pSoundLayer->DrawBackgroundImage();
    m_pScene->AddChild(m_pSoundLayer);

    m_pSoundLayer->m_Sound.MakeSpectrogram();

    m_pSoundLine = new CKhuGleSprite(GP_STYPE_LINE, GP_CTYPE_KINEMATIC,
        CKgLine(CKgPoint(0, 0), CKgPoint(0, m_pSoundLayer->m_nH)),
        KG_COLOR_24_RGB(255, 0, 0), false, 0);
    m_pSoundLayer->AddChild(m_pSoundLine);
}

```

```

void CSoundProcessing::Update() {
    if(m_bKeyPressed['S']) StopWave();

    if(m_bKeyPressed['T'] || m_bKeyPressed['F'] || m_bKeyPressed['L']) {
        if(m_bKeyPressed['T']) m_pSoundLayer->m_nViewType = 0;
        if(m_bKeyPressed['F']) m_pSoundLayer->m_nViewType = 1;
        if(m_bKeyPressed['L']) m_pSoundLayer->m_nViewType = 2;
        m_pSoundLayer->DrawBackgroundImage();
    }
    if(m_bKeyPressed['M']) {
        int nLength = 3;
        for(int i = 0 ; i < m_pSoundLayer->m_Sound.m_nSampleLength-nLength ; ++i) {
            for(int ii = 1 ; ii < nLength ; ++ii)
                m_pSoundLayer->m_Sound.m_Samples[i]
                    += m_pSoundLayer->m_Sound.m_Samples[i+ii];
            m_pSoundLayer->m_Sound.m_Samples[i] /= nLength;
        }
        m_pSoundLayer->m_Sound.MakeSpectrogram();
        m_pSoundLayer->DrawBackgroundImage();
        m_bKeyPressed['M'] = false;
    }
}

```

```

if(m_bKeyPressed['P'])
    PlayWave(m_pSoundLayer->m_Sound.m_Samples,
            m_pSoundLayer->m_Sound.m_nSampleRate,
            m_pSoundLayer->m_Sound.m_nSampleLength);

unsigned long nPosition;
GetPlaybackPosotion(&nPosition);
if(nPosition > 0)
    m_pSoundLine->MoveTo(nPosition*600/m_pSoundLayer->m_Sound.m_nSampleLength,
            m_pSoundLayer->m_nH/2);

m_pScene->Render();
DrawSceneTextPos("Sound Processing", CKgPoint(0, 0));
CKhuGleWin::Update();
}

```

```

int main() {
    char ExePath[MAX_PATH], SoundPath[MAX_PATH];
    GetModuleFileName(NULL, ExePath, MAX_PATH);

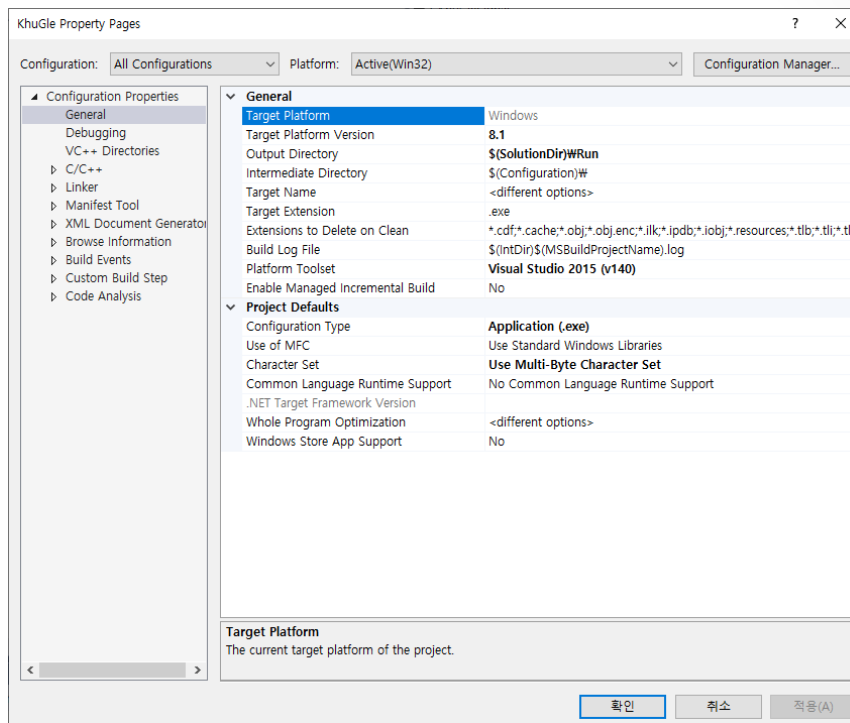
    int i;
    int LastBackSlash = -1;
    int nLen = strlen(ExePath);
    for(i = nLen-1 ; i >= 0 ; i--) {
        if(ExePath[i] == '\\') {
            LastBackSlash = i;
            break;
        }
    }
    if(LastBackSlash >= 0)
        ExePath[LastBackSlash] = '\\0';
    sprintf(SoundPath, "%s\\%s", ExePath, "ex.wav");

    CSoundProcessing *pSoundProcessing = new CSoundProcessing(640, 480, SoundPath);
    KhuGleWinInit(pSoundProcessing);

    return 0;
}

```

# Project setting



# Speech data

<http://www.cstr.ed.ac.uk/projects/eustace/download.html>



**The Centre for Speech Technology Research**  
The University of Edinburgh

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## Full set of sentence recordings for downloading

**Directory structure of the speech files**

There are two version of the EUSTACE downloadable speech corpus, one containing speech files in .wav (RIFF) format and one containing speech files in .sd (ESPS) format. For each version, the top directory contains a README file, with outline information about the corpus and a directory, **speech.wav** or **speech.esps**, which itself contains a directory for each subject, **subject1** to **subject3**. Within each subject directory are two directories **right-headed** and **left-headed**, containing the speech files, named according to the conventions outlined in the description of the organisation of recorded sentences into sound files (see PROCEDURE page, where a full list of the available files is also available).

- [SPEECH DIRECTORIES](#): Diagram illustrating the directory structure of downloadable speech files.

The downloadable speech files are available as:

- a single gzipped tar file, containing speech files in .sd (ESPS) format.
- a single zip file, containing speech files in .wav (RIFF) format.
- a set of six gzipped tar files, each containing the speech files in .sd (ESPS) format for one of the six speakers.
- a set of six zip files, each containing the speech files in .wav (RIFF) format for one of the six speakers.



- FIR
  - Finite impulse response

$$y[n] = b[0]x[n] + b[1]x[n-1] + \cdots + b[N-1]x[n-N+1]$$

- IIR
  - Infinite impulse response

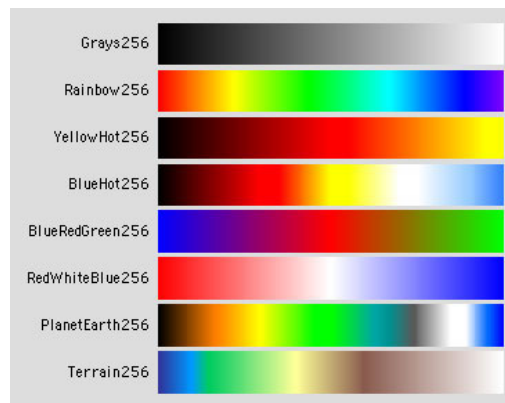
$$y[n] = b[0]x[n] + b[1]x[n-1] + \cdots + b[N-1]x[n-N+1] \\ - a[1]y[n-1] + a[2]y[n-2] + \cdots + a[M]y[n-M]$$

## Practice III

- Cepstrum
- Sound generation

# Advanced Courses (1)

- FIR filter design
  - Window method
- IIR filter design
  - Bilinear
- Pseudo color



[https://www.wavemetrics.com/sites/www.wavemetrics.com/files/images-imported/256-color-versions\\_3.jpg](https://www.wavemetrics.com/sites/www.wavemetrics.com/files/images-imported/256-color-versions_3.jpg)

- Window types
  - Hamming window

$$w_h(n) = \begin{cases} 0.54 - 0.46 \cos\left(\frac{2\pi n}{N-1}\right) & 0 \leq n \leq N-1 \\ 0 & \text{otherwise} \end{cases}$$

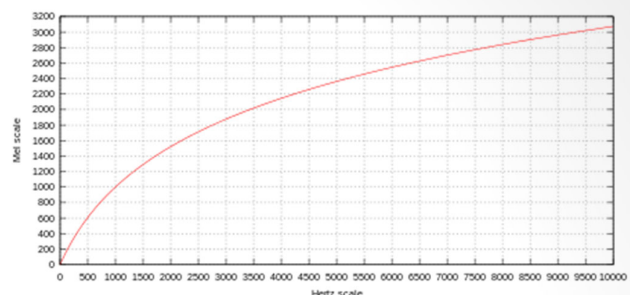
# Advanced Courses (2)

- Sound features
  - Short time energy

$$E_m = \sum_{n=0}^{N-1} \left| x(n)w(m-n) \right|^2$$

- ZCR (zero cross rate)
  - Speech/music classification

$$Z_m = \frac{1}{2N} \sum_{n=0}^{N-1} \left| \text{sign}(x(n)) - \text{sign}(x(n-1)) \right| w(m-n)$$



[https://upload.wikimedia.org/wikipedia/commons/thumb/a/aa/Mel-Hz\\_plot.svg/450px-Mel-Hz\\_plot.svg.png](https://upload.wikimedia.org/wikipedia/commons/thumb/a/aa/Mel-Hz_plot.svg/450px-Mel-Hz_plot.svg.png)

- MFCC (mel-frequency cepstral coefficients)
  - MFC: representation of the short-term power spectrum and the frequency bands are equally spaced on the mel scale