Music Synthesizer Report

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- See My Github [Repository of This Project]()

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1. 简单的音乐合成 `Question_1/`

1.1 播放东方红的片段(有'啪啪声') `Oriental_Red.m`

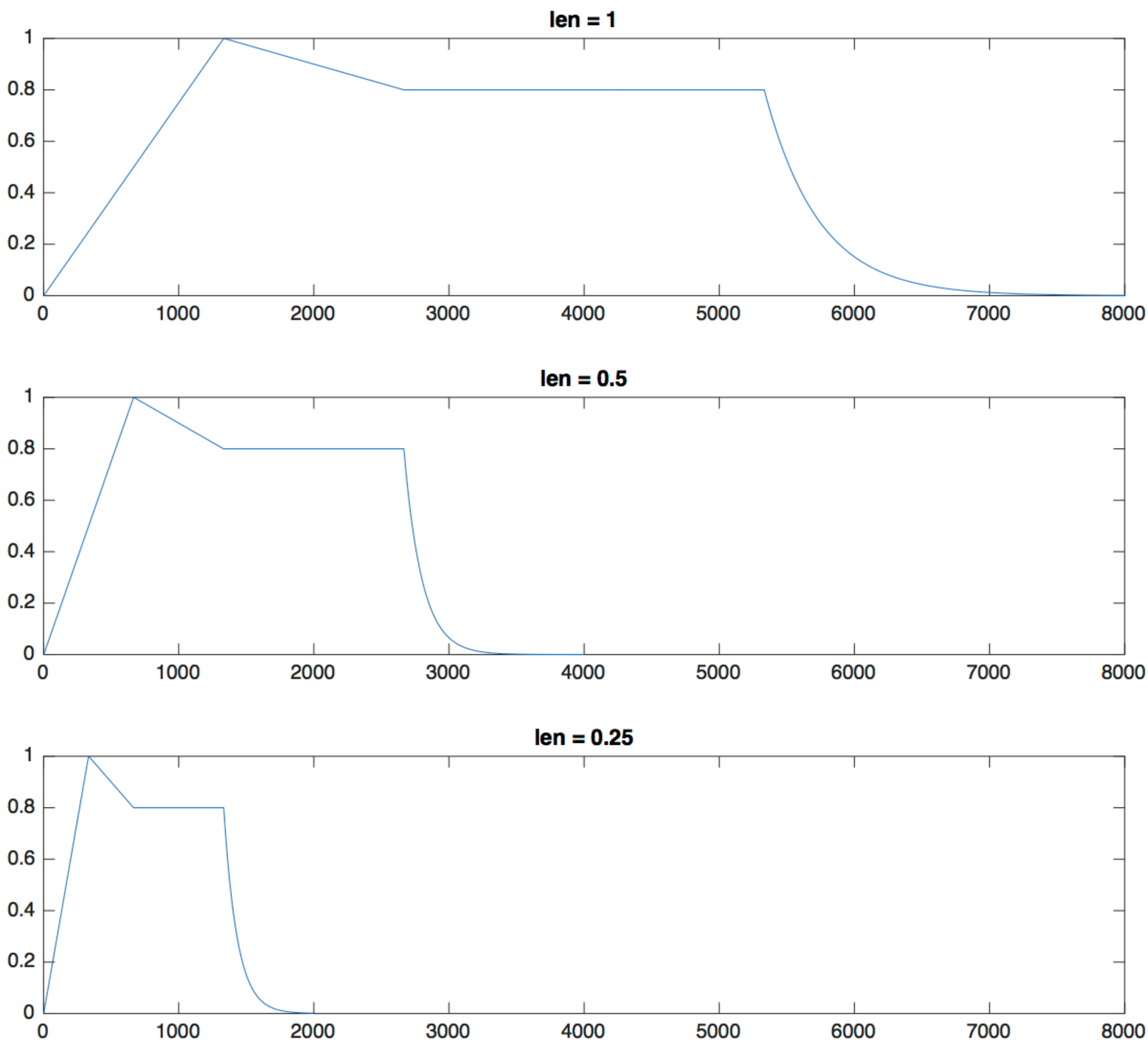
- 一开始摸索了一整子, 因为在我的刻板印象里, Matlab 可以用来播放音乐!? Excuse me? 后来试着用 `sound` 函数播放了一段东西, 觉得 Matlab 实在是太神奇了…
- 本小题当中使用的主要方法就是将要发出的声音 `y`, 在每次循环 (每个音符) 当中扩大 `y = [y, sin(2 * pi * tone(i) * t )];`
- 最后使用 sound 函数, 以 `sample_rate= 8000` 播放
- Comments:
    - 一开始我播放的时候觉得播放的速度有点慢… 所以我加入了一个 `speed` 变量, 让播放的速度增加一倍
    - 东方红的这个片段的第 6 个音符是降了一个八度的 D(6), 这里因为这个片段其他音符的 range 并没有这么广, 因此我直接用 `f(6)/2` 来实现降了一个八度的 D(6)
    - 听起来的确是有'啪啪声'
- 以下为本题的源代码:

```
% The First Problem: Oriental_Red
function Oriental_Red
    speed = 2;
    sample_rate = 8000;
    len = [1,0.5,0.5,2,1,0.5,0.5,2];
    len = len / speed;

    %      F(1), G(2), A(3),  B-(4),  C(5),   D(6),   E(7)
    f = [349.23, 392,  440,  466.16, 523.25, 587.33, 659.25];
    tone = [f(5),f(5),f(6),f(2),f(1),f(1),f(6)/2,f(2)];

    % Generate Sin Signal
    y = [];
    for i = 1:length(tone)
        t = linspace(0,len(i),len(i)*sample_rate);
        y = [y, sin(2 * pi * tone(i) * t )];
    end
    % Make sound
    sound(y, sample_rate);
end
```

1.2 用包络修正 -> 消除'啪啪声' `Oriental_Red_2.m` , `generate_volume.m`

- 本小题的思路都与上一个小题相同, 区别之处仅在于生成的 `y` 多乘了一个包络 `volume_array`
- 这里产生包络的函数为 generate_volume, 使用它生成的包络波形、代码如下:

```
% return an array of volume strength
function volume_array = generate_volume(len_divide_speed,sample_rate)
    unit = sample_rate * len_divide_speed;
    x1 = linspace(0,len_divide_speed/6,unit/6);
    x2 = linspace(0,len_divide_speed/6,unit/6);
    x3 = linspace(1,1,unit/3);
    x4 = linspace(0,len_divide_speed/3,unit-length([x1,x2,x3]));
    volume_array = [6/len_divide_speed*x1, 1-1.2/len_divide_speed*x2,...
        0.8*x3, 0.8*exp(-(100-80*len_divide_speed)*x4)];
    plot(volume_array);
end
```



`Oriental_Red_2.m` 里加入的关键代码如下:

~~~~
```
% Volume
volume_array =[];
for i=1:length(len)
   volume_array = [volume_array, generate_volume(len(i),sample_rate)];
end

% y suppressed by volume
y = y .* volume_array;
```
~~~~

- 播放出来之后果然没有了'啪啪声'

1.3 声调、降调 `Oriental_Red_3.m`

1. 方法1: 直接修改音调

    - 一开始看到这题的时候, 我不太懂老师的意思...
        - 升一个八度, 降一个八度,不就直接将 `tone` 除以2就好? 如下:
        - `tone = [f(5),f(5),f(6),f(2),f(1),f(1),f(6)/2,f(2)]; tone = tone /2;`
        - 上升半个音阶就 `tone = tone * 2^(1/12)` 就好啦
        - 播放出来也没有什么问题...

    - 为什么老师还要在后面加一句 (提示:音乐播放的时间可以变化, 用 `resample` 函数) 呢?
    - 后来才知道老师的意思是不能直接修改输入的音调

2. 方法2: 修改 `sample_rate`

    - 比较简单的一个方法是直接修改 `sample_rate` , 将 `sample_rate` *2 或 /2
    - 即让 Matlab `sound` 函数采样的频率上升/下降一倍即可以做到 升一个八度/降一个八度
    - 关键代码如下:

    ```
    % sound(y, 2*sample_rate);    % up a key
    % sound(y, 1/2*sample_rate);  % down a key
    ```

3. 方法3: 使用 `resample` 函数

    - `resample` 函数, 顾名思义, 即对输入序列进行重新采样
    - 因为 `resample` 函数的特殊用法, 首先需要使用 `rat` 函数得到 `2^(1/12)` 的近似, 然后将 `p, q` 输入 `resample` 函数
    - 这样子相当于就将 `y` 增加/减少了一些值
    - 关键代码如下:

```
% up for a half degree %%%%%%%
[p,q] = rat(2^(1/12),0.00000001);
y = resample(y,q,p);
```

4. 总结

   - 方法2 & 方法3 其实有类比性
     - 方法2 是 提高/降低 了 `sample_rate` , 即让 `sound` 函数每秒 多/少 读一些值
     - 方法3 是 减少/增加 了 `y` , 即让 `sound` 函数要读比较 少/多 值才能结束
   - 殊途同归, 使用 方法2, 方法3 都会使输出的声音 播放时间变快/变慢

1.4 增加谐波分量 `Oriental_Red_4.m`

- 这里按照题目里给的要求, 增加了 0.2*二次谐波, 0.3 *三次谐波
- 播放出来的声音…还蛮像风琴的…吧
- 增加谐波分量的关键代码如下:

```
% Generate Harmonic Sin Signal
y = [];
for i = 1:length(tone)
    t = linspace(0,len(i),len(i)*sample_rate);
    y = [y, [1, 0.2, 0.3] * ...
        [sin(2*pi*tone(i)*t);sin(2*pi*2*tone(i)*t);sin(2*pi*3*tone(i)*t)]];
end
```

1.5 合成贝多芬第五交响乐开头两小节 `Beethoven_5.m`

- 贝多芬第五交响乐就是 命运！
- 前两小节的节拍、音符如下:

```
len = [1,1/3,1/3,1/3,2, 1,1/3,1/3,1/3,2];
tone = [0, f(4),f(4),f(4),f(2),0, f(3),f(3),f(3),f(1)];
```

- 播放出来真的还蛮像的!

2. 用傅立叶级数分析音乐 `Question_2/`

2.1 播放 fmt.wav `load_data.m`

- `load` 并 播放 `fmt.wav` 的代码如下:

```
% !! wavread() deprected, cannot be used.
music = audioread('fmt.wav');
sound(music, 8000);
% Indeed more genuine
```

- 听起来的确是就是真实的吉他声...

2.2 预处理 -> 生成 wave2proc 信号 `preprocessing.m`

1. 思路
   - 首先观察 `realwave` 波形, 发现有十个周期。
   - 这里与处理的主要思路就是将这十个周期的波形求平均, 然后再重复十遍
2. 过程:
   - 步骤1: 发现 `realwave` 信号有243个值, 并不是10的倍数, 因此先用 `resample` 函数将 `realwave` 进行重采样, 采样成 2430 个值, 关键代码如下:

   ```
   % First *10, output a 2430 elements array
   input_array_10 = resample(input_array', cycle, 1);
   ```

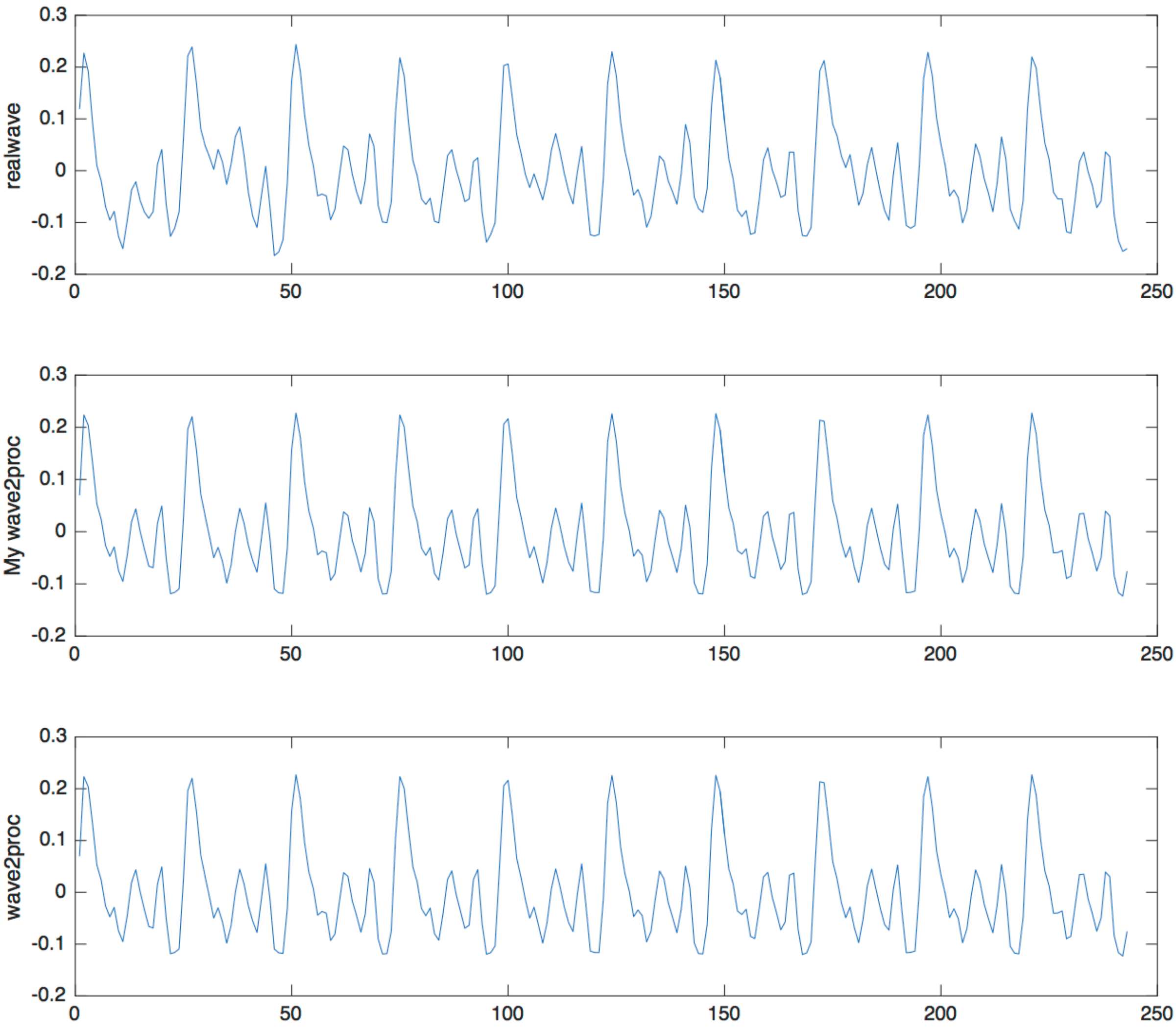
   - 步骤2: 然后将波形等分10份, 并求这十份的平均, 然后拼在一起, 关键代码如下:

   ```
   % Calculate the mean of the 10 cycles
   unit = mean(...
       reshape(input_array_10', [length(input_array),cycle])'...
   );

   % Integrate into one array
   units = [unit,unit,unit,unit,unit,unit,unit,unit,unit,unit];
   ```

   - 步骤3: 拼在一起后一定会有波形不连续的问题, 因此再次使用 `resample` 函数 对 `units` 进行重新采样, 关键代码如下:

   ```
   % Resampling
   preprocessed_array = resample(units, 1, cycle)';
   ```

3. 结果
   - 最后产生的波形如下, 第一张图为原来的 `realwave` , 第二张图为我生成的 `wave2proc` , 第三张图为老师提供的 `wave2proc`
   - 从图中可以看出我生成的波形与老师提供的波形几乎没有什么差别。



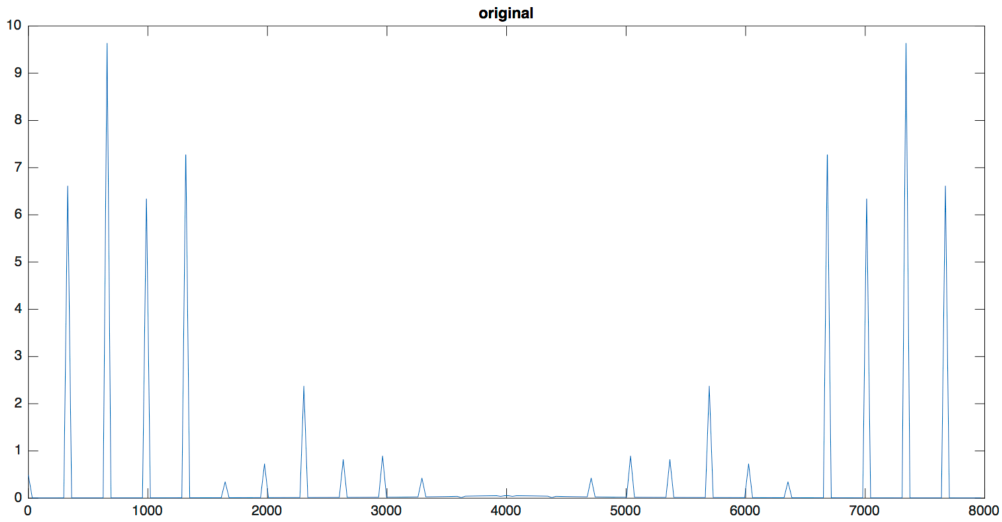
2.3 分析音乐的基频、谐波分量 `Freq_Analyze.m`

1. 不知不觉就到了本次作业当中最难的部分了, 首先先说一下自动分析基频的思路

    1. 第一步:

        - 题目要分析音乐的频率的情况, 因此首先我使用了 `fft` 函数, 将函数从时域转换到频域, 代码、产生信号如下:

```
freqtarget = abs(fft(target));
```



- 发现其实左右是对称的, 因此只取前半部分

```
% half
x = x(1:ceil(length(x)/2));
freqtarget = freqtarget(1:ceil(length(freqtarget) / 2));
```

- 理所当然, 下一步是找到频域里的波峰, 我的做法是找出大于最大波峰幅度*0.2 的部分

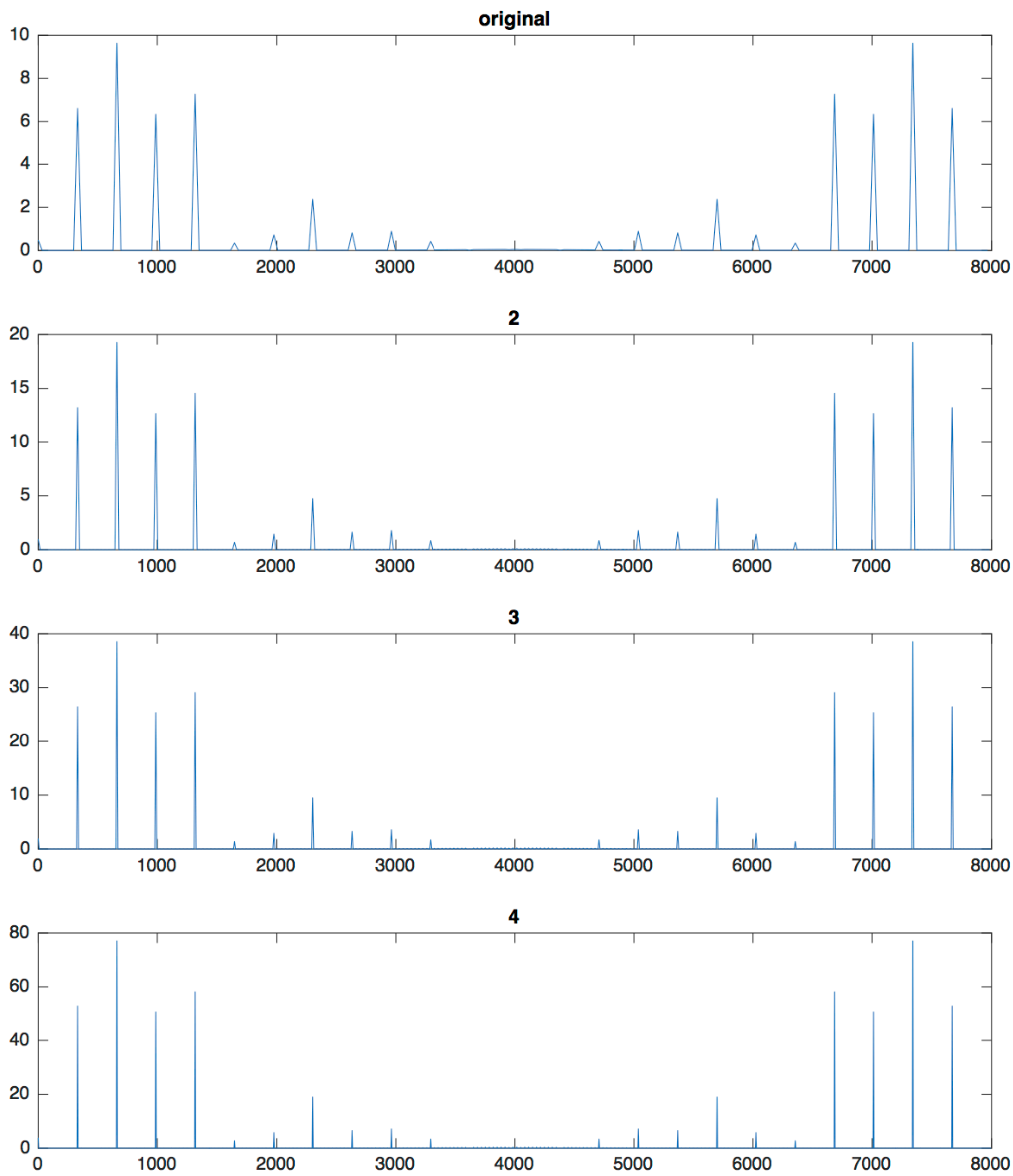
```
% Start Analyzing
% 'freqtarget'
freqtarget = abs(fft(target));

% filter top
maxx = max(freqtarget);
f = find(freqtarget > maxx*0.2);
```

2. 第二步:

- 就跟题目里说的一样, 果然波峰并不是接近于冲激函数, 因此我使用了 `repmat` 函数, 将原波形在时域重复若干次

```
unit = input_array;
target = input_array;
for i = 2:parameter

    target = repmat(target, 2, 1);
    % x axis
    x = [ 0 : length(target)-1 ]/length(target)*8000;

    if(problem_index==8)
        % plot
        subplot(parameter,1,i);
        plot(x, abs(fft(target)));title(i);
    end
end
```

- 以下为每次重复后的结果, 可以看出每个波峰真的越来越接近冲激函数:



3. 第三步:

- 错误:
  - 一开始, 我以为如果你是基频, 那你一定就会有二次、三次、四次谐波分量, 所以我的代码思路是, 先筛选波峰, 找出大于最大波峰幅度*0.2 的部分, 然后从最大的频率往回找,如果这个频率有二次、三次、四次谐波分量, 则你就是基频的备选
  - 然而我挣扎了好久之后, 发现这么做是不对的, 因为其实一个信号不一定会有二次、三次、四次 谐波分量, 而且一个信号的基频分量并不是一定比他的谐波分量要大。

- 正确:
  - 后来, 我正确的做法是:
    1. 首先找到所有频率中的最高峰, 暂时认为他为基频

2. 检查他的1/2, 1/3, 1/4频率处是否有峰, 如果有的话将那个替换为基频

- 这样做没有问题的原因是:
  1. 一般来讲, 一个正常信号的频谱具有最大分量的频率一定是在基频、二次、三次、 四次频率分量处

- 关键代码如下:

```
% possible top
possible_top = freqtarget(f);
[val,index]=max(possible_top);
base = x(f(index));

err = 3;

% ismember( find( (x>base/2-err & x<base/2+err) ), f )
if sum(ismember( find( (x>=base/4-err & x<=base/4+err) ), f ))
    base = x( find( (x>=base/4-err & x<=base/4+err & ismember(x,x(f)))
) );
elseif sum(ismember( find( (x>=base/3-err & x<=base/3+err) ), f ))
    base = x( find( (x>=base/3-err & x<=base/3+err & ismember(x,x(f)))
) );
elseif sum(ismember( find( (x>=base/2-err & x<=base/2+err) ), f ))
    base = x( find( (x>=base/2-err & x<=base/2+err & ismember(x,x(f)) )
 ) );
end
```

- 实际操作之后, 我发现这样做可能会有的一个问题是, 如果频谱比较杂, 在最大频的1/2, 1/3, 1/4处 可能会有多个大于 最大波峰幅度*0.2 的部分, 因此再多加了一个从这里面筛选出来分量最大的那个 作为基频的几行代码, 如下:

```
% If more than one base exist, give it with the biggest amp
ans = find( ismember(x,base) );
[val, index] = max( freqtarget( ans ) );

base = x( ans(index) );
```

- 在得出来基频之后, 在去二倍、三倍、四倍基频处找到二次、三次、四次谐波分量的大小

```
% Calculate Harmonic Components
one_amp = freqtarget(x == base);

two_amp_index = find(x>base*2-err & x<base*2+err & ismember(x,x(f)) );
[val, index] = max( freqtarget( two_amp_index ) );
two_amp = freqtarget( two_amp_index(index) );

three_amp_index = find(x>base*3-err & x<base*3+err & ismember(x,x(f)) )
;
[val, index] = max( freqtarget( three_amp_index ) );
three_amp = freqtarget( three_amp_index(index) );

four_amp_index = find(x>base*4-err & x<base*4+err & ismember(x,x(f)) );
[val, index] = max( freqtarget( four_amp_index ) );
four_amp = freqtarget( four_amp_index(index) );

% Check if its empty(zero)
if isempty(two_amp)
    two_amp = 0;
end
if isempty(three_amp)
    three_amp = 0;
end
if isempty(four_amp)
    four_amp = 0;
end
```

- 最后输出结果

```
% Output result
report = table(base, one_amp./one_amp,two_amp./one_amp,...
    three_amp./one_amp,four_amp./one_amp ,tone_cell,...
'VariableNames', {'Base' 'base_amp' 'two_amp' 'three_amp' 'four_amp' 'Tone
'})
```

```
    Base      base_amp    two_amp    three_amp    four_amp    Tone
    ______    ________    _______    _________    ________    ____

    329.22    1           1.4572     0.95874      1.0999      'e1'
```

2.4 自动化分析 fmt.wav 的基频、谐波分量 `Analyze_fmt.m`

1. 这里采用手动分割音符的方法 QAQ...代码如下:

```
start_time =[700,  2300 ,14000, 18000, 22000, 25000, 29000,...
    32000, 36000, 40000, 46000, 48000, 56000, 62000, 68000,...
    72000, 76000, 79000, 81000, 83000, 84500, 86500, 90000,...
    94000, 98000,  102000, 106000, 110000, 114500, 119000];
end_time =  [2300, 14000,18000, 22000, 25000, 29000, 32000,...
    36000, 40000, 46000, 48000, 56000, 62000, 68000, 72000,...
    76000, 79000, 81000, 83000, 84500, 86500, 90000, 94000,...
    98000, 102000, 106000, 110000, 114500, 119000, 131000];
```

2. 然后依次对每个音符传入 `Freq_Analyze` 函数进行处理, 并将结果存入变量中

```
for i = 1:length(start_time)
    [base_uut, one_amp_uut,two_amp_uut,three_amp_uut,four_amp_uut ,tone_uut] =
 ...
    Freq_Analyze( music(start_time(i):end_time(i)), 6, 9);

    leng = ( end_time(i) - start_time(i) )*2 / 4000;
    leng = round(leng) / 2;

    len(i,1) = leng;
    base(i,1) = base_uut;
    one_amp(i,1) = one_amp_uut;
    two_amp(i,1) = two_amp_uut;
    three_amp(i,1) = three_amp_uut;
    four_amp(i,1) = four_amp_uut;
    tone{i,1} = (tone_uut);
end
```

3. 最后一起用 `table` 函数输出:

```
report = table(base, len, two_standard,...
    three_standard, four_standard, tone,...
    'VariableNames', {'Base' 'length' 'two_amp' 'three_amp' 'four_amp' 'Tone'})
```

4. 输出结果:

Base	length	two_amp	three_amp	four_amp	Tone
219.86	0.5	0	0	0	'a '
221.52	3	0.29724	0	0	'a '
247.94	1	0	0	0	'b '
221.94	1	0	0	0	'a '
295.9	1	1.2095	0	0	'd1'
329.92	1	1.1482	0.88077	0	'e1'
194.6	1	0.74316	0	0	'g '
221.94	1	0.21927	0	0	'a '
173.96	1	0.35087	0	0	'f '
294.62	1.5	0.75596	0	0	'd1'
207.9	0.5	0	0.20372	0	'bA'
247.97	2	0.26959	0	0	'b '
165.31	1.5	2.7418	0	1.8888	'f '
222.63	1.5	0	0	0	'a '
163.96	1	2.0917	0	4.6697	'f '
219.95	1	2.0084	0.9904	1.0287	'a '
221.26	1	0.2717	0	0	'a '
131.93	0.5	1.7638	3.5543	0	'f '
351.82	0.5	0	0	0	'f1'
330.45	0.5	1.2344	0.89083	0	'e1'
291.85	0.5	0.38878	0	0	'd1'
329.05	1	2.3378	1.1515	1.1278	'e1'
247.94	1	0.22354	0	0	'b '
145.96	1	3.1962	0.956	2.0246	'f '
261.93	1	0.53623	0.21021	0	'c1'
173.96	1	0.33968	0	0	'f '
221.94	1	0	0	0	'a '
165.3	1	1.7435	0	0	'f '
222.17	1	0	0	0	'a '
209.98	3	0	0	0	'bA'

3. 基于傅立叶级数的音乐合成 `Question_3/`

3.1 用 2.3 算出的谐波分量, 再次完成 1.4 `Oriental_Red_with_harm_1.m`, `generate_volume_for3.m`

1. 更改谐波分量大小

    - 回顾 2.3 算出的谐波分量大小

Base	base_amp	two_amp	three_amp	four_amp	Tone
329.22	1	1.4572	0.95874	1.0999	'e1'

- 更改 1.4 中谐波分量的大小, 关键代码如下:

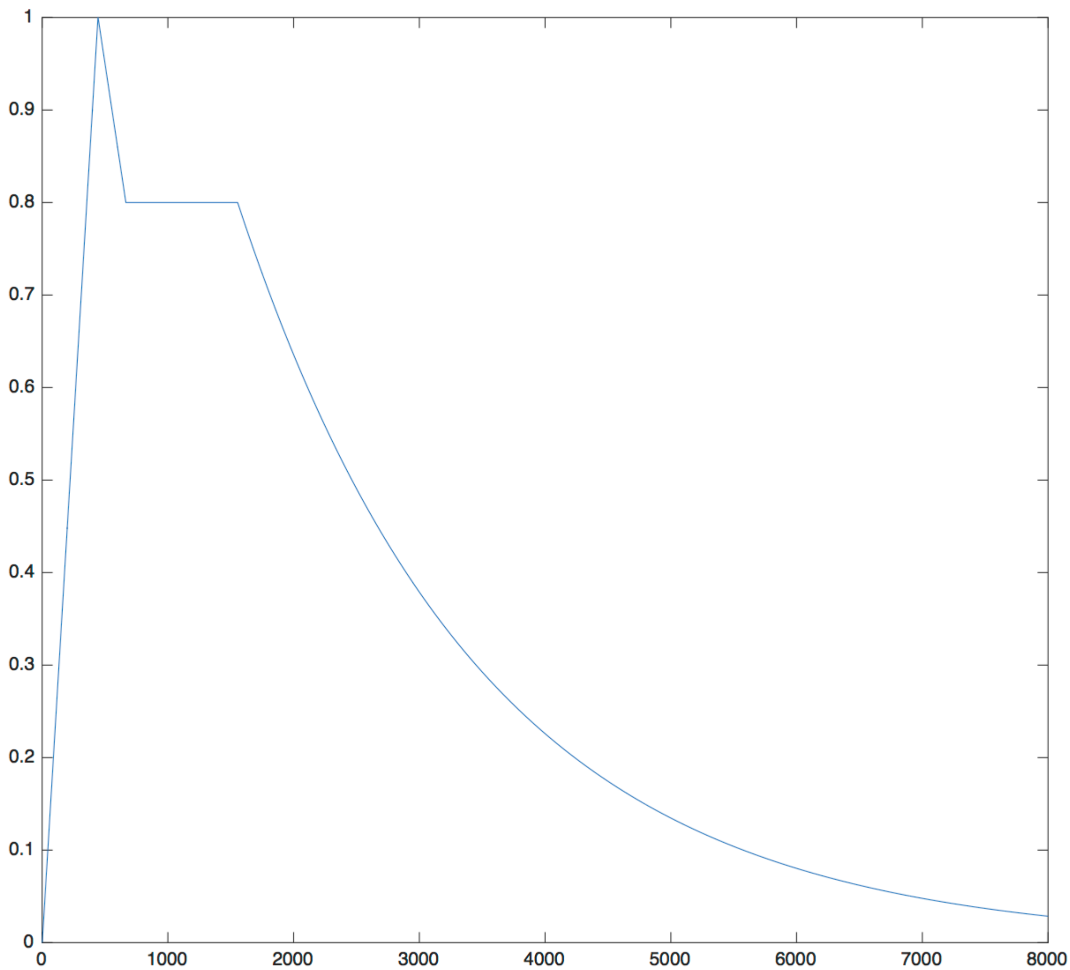
```
% Generate Harmonic Sin Signal
y = [];
for i = 1:length(tone)
    t = linspace(0,len(i),len(i)*sample_rate);
    y = [y, [1, 1.4572, 0.95874, 1.0999] * ...
        [sin(2*pi*tone(i)*t); sin(2*pi*2*tone(i)*t);...
        sin(2*pi*3*tone(i)*t); sin(2*pi*4*tone(i)*t)]];
end
```

- 结果发现生成的音乐一点也不像吉他! 仔细想了一下, 发现好像波形的包络也会影响到音色

2. 更改波形包络

- 因此重新写了一个 `generate_volume_for3` 函数, 生成的波形、代码如下:

```
% return an array of volume strength
function volume_array = generate_volume_for3(len_divide_speed,sample_rate)
    unit = sample_rate * len_divide_speed;
    x1 = linspace(0,len_divide_speed/6,unit/18);
    x2 = linspace(0,len_divide_speed/6,unit/36);
    x3 = linspace(1,1,unit/9);
    x4 = linspace(0,len_divide_speed/3,unit-length([x1,x2,x3]));
    volume_array = [6/len_divide_speed*x1, 1-1.2/len_divide_speed*x2,...
        0.8*x3, 0.8*exp(-(100-90*len_divide_speed)*x4)];
    plot(volume_array);
end
```



- 使用这个包络产生的音乐就像吉他多了, 但离真实的声音还是有一点差距的...

3.2 用 2.4 分析出的谐波分量, 再次完成 1.4 `Oriental_Red_with_harm_2.m`

- 首先跑一遍 `Analze_fmt` , 以得到每个音符的基频, 二次、三次、四次谐波分量

```
% Get info from fmt
[base, two_standard, three_standard, four_standard] = Analyze_fmt;
```

- 然后找到每个音符距离 `Analyze_fmt` 里最近的音符, 使用那个音符的谐波分量来生成乐音

```
y = [];
for i = 1:length(tone)
    t = linspace(0,len(i),len(i)*sample_rate);

    [val, index] = min( abs(tone(i) - base) );

    y = [y, [1, two_standard(index), three_standard(index), four_standard(inde
x)] * ...
        [sin(2*pi*tone(i)*t); sin(2*pi*2*tone(i)*t);...
        sin(2*pi*3*tone(i)*t); sin(2*pi*4*tone(i)*t)]];
end
```

- 最后生成的音乐的确是比第一小题像吉他多了…不过我个人感觉还是有一些差距的…

3.3 将上述功能封装成 GUI `music_syn_gui.fig` , `music_syn_gui.m`

1. 实现:
    - 首先要先将 2.4 分析出数据保存下来, 保存到了 `guitar.mat`
    - 第 3.2 的生成音乐的函数封装了起来, 可以输入音调、节拍, 代码如下:

```
function playmusic(tones, len)
    load guitar.mat

    speed = 2;
    sample_rate = 8000;
    len = len / speed;

    % Volume
    volume_array =[];
    for i=1:length(len)
       volume_array = [volume_array, generate_volume_for3(len(i),sample_ra
te)];
    end

    %      F(1), G(2), A(3),  B-(4),  C(5),   D(6),   E(7)
    f = [349.23, 392,  440,  466.16, 523.25, 587.33, 659.25];
    tone = f(tones);

    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
    % Generate Harmonic Sin Signal

    y = [];
    for i = 1:length(tone)
        t = linspace(0,len(i),len(i)*sample_rate);

        [val, index] = min( abs(tone(i) - base) );

        y = [y, [1, two_standard(index), three_standard(index), four_stand
ard(index)] * ...
            [sin(2*pi*tone(i)*t); sin(2*pi*2*tone(i)*t);...
            sin(2*pi*3*tone(i)*t); sin(2*pi*4*tone(i)*t)]];
    end
    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

    % y suppressed by volume
    y = y .* volume_array;

    % Make sound
    sound(y, sample_rate);
```

- 按钮按下时的触发函数:

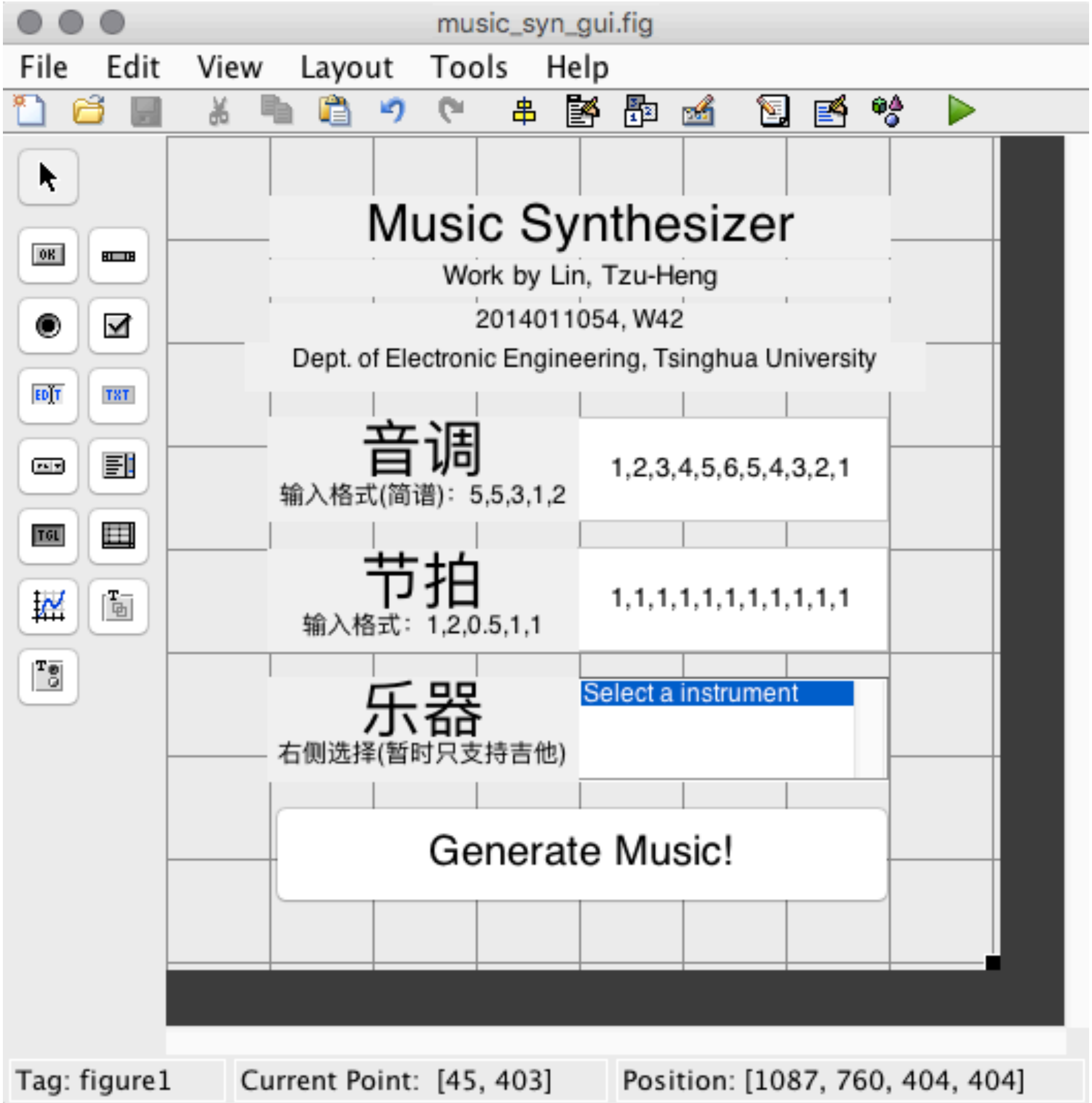
```
% --- Executes on button press in pushbutton1.
function pushbutton1_Callback(hObject, eventdata, handles)
    % hObject    handle to pushbutton1 (see GCBO)
    % eventdata  reserved - to be defined in a future version of MATLAB
    % handles    structure with handles and user data (see GUIDATA)

    tones = str2num(get(handles.edit1,'string'));
    len = str2num(get(handles.edit2,'string'));

    playmusic(tones, len);
```

- 可选要模仿成哪种乐器 (页面上显示: guitar, violin, piano 但目前仅支持吉他):

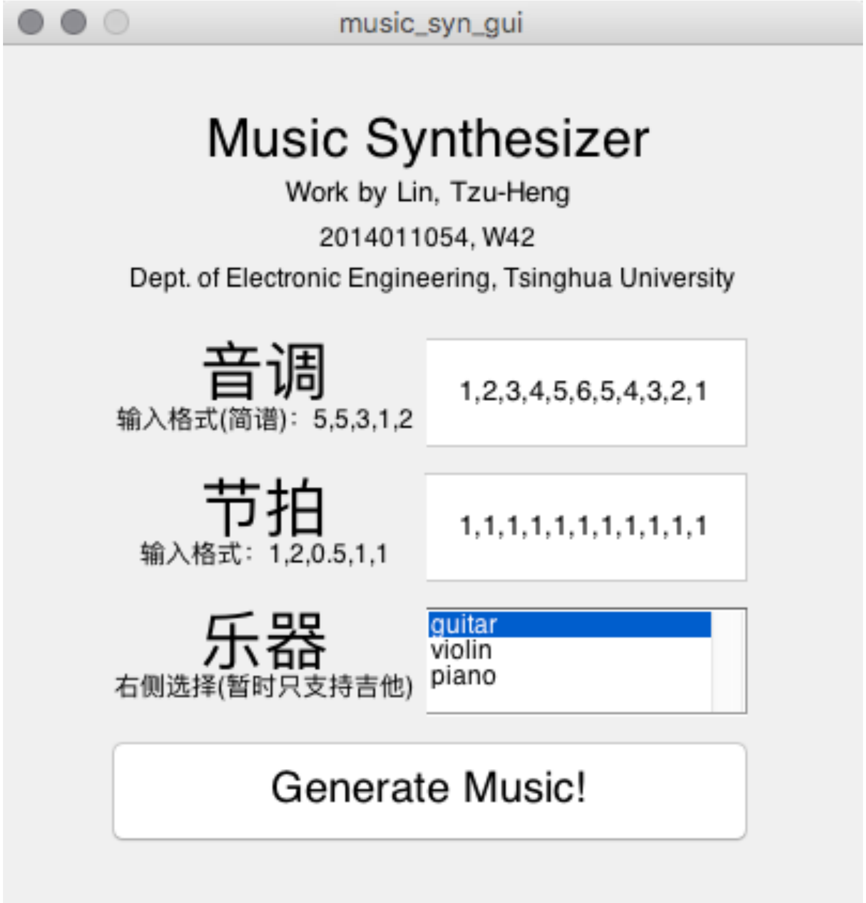
```
% --- Executes just before music_syn_gui is made visible.
function music_syn_gui_OpeningFcn(hObject, eventdata, handles, varargin)
    % This function has no output args, see OutputFcn.
    % hObject    handle to figure
    % eventdata  reserved - to be defined in a future version of MATLAB
    % handles    structure with handles and user data (see GUIDATA)
    % varargin   command line arguments to music_syn_gui (see VARARGIN)

    set(handles.listbox1, 'string', {'guitar', 'violin', 'piano'});

    % Choose default command line output for music_syn_gui
    handles.output = hObject;

    % Update handles structure
    guidata(hObject, handles);

    % UIWAIT makes music_syn_gui wait for user response (see UIRESUME)
    % uiwait(handles.figure1);
```

2. 结果:

    - gui的编辑界面:

- 

- 最后输出的页面，亲测可用：

- 

4. 原创性声明

本报告的所有内容、代码均为本人原创

5. 写在最后

- 这次实验让我对 Matlab 有了另一番的认识，原本我以为他只能做一些简单的数学运算，没想到他竟然还能做这样子的事情…
- 做实验的过程当中也是走了挺多弯路，比如说一开始的时候真的是无从下手，连 `sound` 函数怎么用都不知道…，所以真的是看了很久的 help 也不知道要干嘛…
- 慢慢的后来就好了很多，整个老师出的题目也是循序渐进，后来还遇到很多问题，在上网查答案、 自己读 documentation 之中一一解决，自己debug的能力也提升了不少。
- 我必须承认的一个不足是，在分析 `fmt.wav` 的部分，我知道我分析出来的音频一定有问题，可能是出在我手动标定开始、结束时间，也可能是这段音频里面本身就有一些两个音一起弹的部分，让我得出来的分析结果其实并不精确。这个部分算是在这个实验里我觉得做的最不理想的部分了。
- 总之，这次实验我真的学到了好多好多~~!

6. 源程序

1. 简单的音乐合成 `Question_1/`

1. 播放东方红的片段(有啪声) `Oriental_Red.m`
2. 用包络修正 -> 消除啪啪声 `Oriental_Red_2.m` , `generate_volume.m`
3. 声调、降调 `Oriental_Red_3.m`
4. 增加谐波分量 `Oriental_Red_4.m`
5. 合成贝多芬第五交响乐开头两小节 `Beethoven_5.m`

2. 用傅立叶级数分析音乐 `Question_2/`

1. 播放 fmt.wav `load_data.m`
2. 预处理 -> 生成 wave2proc 信号 `preprocessing.m`
3. 分析音乐的基频、谐波分量 `Freq_Analyze.m`
4. 自动化分析 fmt.wav 的基频、谐波分量 `Analyze_fmt.m`

3. 基于傅立叶级数的音乐合成 `Question_3/`

1. 用 2.3 算出的谐波分量, 再次完成 1.4 `Oriental_Red_with_harm_1.m` , `generate_volume_for3.m`
2. 用 2.4 分析出的谐波分量, 再次完成 1.4 `Oriental_Red_with_harm_2.m`
3. 将上述功能封装成 GUI `music_syn_gui.fig` , `music_syn_gui.m`

- Oriental_Red.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The First Problem: Oriental_Red
function Oriental_Red
    speed = 2;
    sample_rate = 8000;
    len = [1,0.5,0.5,2,1,0.5,0.5,2];
    len = len / speed;

    %      F(1), G(2), A(3),  B-(4),  C(5),   D(6),   E(7)
    f = [349.23, 392,  440,  466.16, 523.25, 587.33, 659.25];
    tone = [f(5),f(5),f(6),f(2),f(1),f(1),f(6)/2,f(2)];

    % Generate Sin Signal
    y = [];
    for i = 1:length(tone)
        t = linspace(0,len(i),len(i)*sample_rate);
        y = [y, sin(2 * pi * tone(i) * t )];
    end
    % Make sound
    sound(y, sample_rate);
end
```

- Oriental_Red_2.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The Second Problem: Oriental_Red without bang
function Oriental_Red_2
    speed = 2;
    sample_rate = 8000;
    len = [1,0.5,0.5,2,1,0.5,0.5,2];
    len = len / speed;

    % Volume
    volume_array =[];
    for i=1:length(len)
       volume_array = [volume_array, generate_volume(len(i),sample_rate)];
    end

    %      F(1), G(2), A(3),  B-(4),  C(5),   D(6),   E(7)
    f = [349.23, 392,  440,  466.16, 523.25, 587.33, 659.25];
    tone = [f(5),f(5),f(6),f(2),f(1),f(1),f(6)/2,f(2)];

    % Generate Sin Signal
    y = [];
    for i = 1:length(tone)
        t = linspace(0,len(i),len(i)*sample_rate);
        y = [y, sin(2 * pi * tone(i) * t )];
    end

    % y suppressed by volume
    y = y .* volume_array;

    % Make sound
    sound(y, sample_rate);
end
```

- generate_volume.m

```
% return an array of volume strength
function volume_array = generate_volume(len_divide_speed,sample_rate)
    unit = sample_rate * len_divide_speed;
    x1 = linspace(0,len_divide_speed/6,unit/6);
    x2 = linspace(0,len_divide_speed/6,unit/6);
    x3 = linspace(1,1,unit/3);
    x4 = linspace(0,len_divide_speed/3,unit-length([x1,x2,x3]));
    volume_array = [6/len_divide_speed*x1, 1-1.2/len_divide_speed*x2,...
        0.8*x3, 0.8*exp(-(100-80*len_divide_speed)*x4)];
    plot(volume_array);
end
```

- Oriental_Red_3.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The Third Problem: Oriental_Red up/down degrees
function Oriental_Red_3
    speed = 2;
    sample_rate = 8000;
    len = [1,0.5,0.5,2,1,0.5,0.5,2];
    len = len / speed;

    % Volume
    volume_array =[];
    for i=1:length(len)
       volume_array = [volume_array, generate_volume(len(i),sample_rate)];
    end

    %      F(1), G(2), A(3),  B-(4),  C(5),   D(6),   E(7)
    f = [349.23, 392,  440,  466.16, 523.25, 587.33, 659.25];
    tone = [f(5),f(5),f(6),f(2),f(1),f(1),f(6)/2,f(2)];

    % Generate Sin Signal
    y = [];
    for i = 1:length(tone)
        t = linspace(0,len(i),len(i)*sample_rate);
        y = [y, sin(2 * pi * tone(i) * t )];
    end

    % y suppressed by volume
    y = y .* volume_array;

    % up for a half degree %%%%%%%
    [p,q] = rat(2^(1/12),0.00000001);
    y = resample(y,q,p);
    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

    % Make sound
    sound(y, sample_rate);
    % sound(y, 2*sample_rate);    % up a key
    % sound(y, 1/2*sample_rate);  % down a key
end
```

- Oriental_Red_4.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The Fourth Problem: Oriental_Red with harmonic
function Oriental_Red_4
    speed = 2;
    sample_rate = 8000;
    len = [1,0.5,0.5,2,1,0.5,0.5,2];
    len = len / speed;

    % Volume
    volume_array =[];
    for i=1:length(len)
       volume_array = [volume_array, generate_volume(len(i),sample_rate)];
    end

    %      F(1), G(2), A(3),  B-(4),  C(5),   D(6),   E(7)
    f = [349.23, 392,  440,  466.16, 523.25, 587.33, 659.25];
    tone = [f(5),f(5),f(6),f(2),f(1),f(1),f(6)/2,f(2)];

    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
    % Generate Harmonic Sin Signal
    y = [];
    for i = 1:length(tone)
        t = linspace(0,len(i),len(i)*sample_rate);
        y = [y, [1, 0.2, 0.3] * ...
            [sin(2*pi*tone(i)*t);sin(2*pi*2*tone(i)*t);sin(2*pi*3*tone(i)*t)]]
;
    end
    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

    % y suppressed by volume
    y = y .* volume_array;

    % Make sound
    sound(y, sample_rate);
end
```

- Beethoven_5.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The Fifth Problem: Beethoven Symphony No.5
% in C Minor
function Beethoven_5
    speed = 2;
    sample_rate = 8000;
    len = [1,1/3,1/3,1/3,2, 1,1/3,1/3,1/3,2];
    len = len / speed;

    % Volume
    volume_array =[];
    for i=1:length(len)
       volume_array = [volume_array, generate_volume(len(i),sample_rate)];
    end

    %      F(1), G(2),  -A(3),   B(4),   C(5),   D(6),   E-(7)
    % f = [174.61, 196,  207.65, 246.94, 261.63, 293.66, 311.13];
    f = [349.23, 392,  415.30,  493.88, 523.25, 587.33, 622.25];
    tone = [0, f(4),f(4),f(4),f(2),0, f(3),f(3),f(3),f(1)];

    y = [];
    for i = 1:length(tone)
        t = linspace(0,len(i),len(i)*sample_rate);
        y = [y, sin(2*pi*tone(i)*t)];
    end

    % y suppressed by volume
    y = y .* volume_array;

    % Make sound
    sound(y, sample_rate);
end
```

- load_data.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The Sixth Problem

clear; clc;
load Guitar.MAT;
figure;
subplot(2,1,1);plot(realwave);ylabel('realwave');
subplot(2,1,2);plot(wave2proc);ylabel('wave2proc');

% !! wavread() deprected, cannot be used.
music = audioread('fmt.wav');
sound(music, 8000);
% Indeed more genuine
```

- preprocessing.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The Seventh Problem

% input_array -> realwave
% cycle -> The amount of cycles we saw
% Standard -> The array teacher has given

function preprocessed_array = preprocessing(input_array, cycle, standard)
    % First *10, output a 2430 elements array
    input_array_10 = resample(input_array', cycle, 1);

    % Calculate the mean of the 10 cycles
    unit = mean(...
        reshape(input_array_10', [length(input_array),cycle])'...
    );

    % Integrate into one array
    units = [unit,unit,unit,unit,unit,unit,unit,unit,unit,unit];

    % Resampling
    % preprocessed_array = resample(units, length(input_array), 1);
    preprocessed_array = resample(units, 1, cycle)';

    % Plot
    figure;
    subplot(3,1,1); plot(input_array); ylabel('realwave');
    subplot(3,1,3); plot(standard); ylabel('wave2proc');
    subplot(3,1,2); plot(preprocessed_array); ylabel('My wave2proc');

end
```

- Freq_Analyze.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The Eighth Problem

% input_array -> wave2proc
% parameter = 5 -> multiplier of time zone
% problem_index = 8 or 9 -> 8:output a table 9:do nothing

function [base, one_amp,two_amp,three_amp,four_amp ,tone]=...
```

```
Freq_Analyze(input_array, parameter,problem_index)

close all;
% x axis
x = [ 0 : length(input_array)-1 ] / length(input_array)*8000;
plot(x, abs(fft(input_array)));title('original');
if(problem_index==8)
    figure;
    subplot(parameter,1,1);
    plot(x, abs(fft(input_array)));title('original');
end

unit = input_array;
target = input_array;
for i = 2:parameter

    target = repmat(target, 2, 1);
    % x axis
    x = [ 0 : length(target)-1 ]/length(target)*8000;

    if(problem_index==8)
        % plot
        subplot(parameter,1,i);
        plot(x, abs(fft(target)));title(i);
    end
end

% Start Analyzing
% 'freqtarget'
freqtarget = abs(fft(target));

% half
x = x(1:ceil(length(x)/2));
freqtarget = freqtarget(1:ceil(length(freqtarget) / 2));

% test plot
if (problem_index==8)
    figure;
    plot(x,freqtarget);
end

% filter top
maxx = max(freqtarget);
f = find(freqtarget > maxx*0.2);

% possible top
possible_top = freqtarget(f);
[val,index]=max(possible_top);
```

```
    base = x(f(index));

    err = 3;

    % ismember( find( (x>base/2-err & x<base/2+err) ), f )
    if sum(ismember( find( (x>=base/4-err & x<=base/4+err) ), f ))
        base = x( find( (x>=base/4-err & x<=base/4+err & ismember(x,x(f))) ) )
;
    elseif sum(ismember( find( (x>=base/3-err & x<=base/3+err) ), f ))
        base = x( find( (x>=base/3-err & x<=base/3+err & ismember(x,x(f))) ) )
;
    elseif sum(ismember( find( (x>=base/2-err & x<=base/2+err) ), f ))
        base = x( find( (x>=base/2-err & x<=base/2+err & ismember(x,x(f)) ) )
);
    end

    % If more than one base exist, give it with the biggest amp
    ans = find( ismember(x,base) );
    [val, index] = max( freqtarget( ans ) );

    base = x( ans(index) );

    % Calculate Harmonic Components
    one_amp = freqtarget(x == base);

    two_amp_index = find(x>base*2-err & x<base*2+err & ismember(x,x(f)) );
    [val, index] = max( freqtarget( two_amp_index ) );
    two_amp = freqtarget( two_amp_index(index) );

    three_amp_index = find(x>base*3-err & x<base*3+err & ismember(x,x(f)) );
    [val, index] = max( freqtarget( three_amp_index ) );
    three_amp = freqtarget( three_amp_index(index) );

    four_amp_index = find(x>base*4-err & x<base*4+err & ismember(x,x(f)) );
    [val, index] = max( freqtarget( four_amp_index ) );
    four_amp = freqtarget( four_amp_index(index) );

    % Check if its empty(zero)
    if isempty(two_amp)
        two_amp = 0;
    end
    if isempty(three_amp)
        three_amp = 0;
    end
    if isempty(four_amp)
        four_amp = 0;
    end
```

```
    keys = [174.61, 196, 220, 246.94, 261.63, 293.66, 329.63, 349.23, 392, ...
        184.99, 207.65, 233.08, 277.18, 311.13, 369.99, 415.30 ...
    ];
    values = ['f '; 'g '; 'a '; 'b '; 'c1';     'd1';   'e1';  'f1';  'g1';...
        'bG'  ; 'bA'  ; 'bB'  ; 'bD'  ; 'bE'  ; 'bG'  ; 'bA'   ...
    ];

    [val,index] = min(abs(keys-base));
    tone = values(index,:);
    tone_cell = cellstr(tone);

    if(problem_index==8)
    % Output result
        report = table(base, one_amp./one_amp,two_amp./one_amp,...
            three_amp./one_amp,four_amp./one_amp ,tone_cell,...
        'VariableNames', {'Base' 'base_amp' 'two_amp' 'three_amp' 'four_amp' '
Tone'})
    end
end
```

- Analyze_fmt.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The Nineth Problem

function [base, two_standard, three_standard, four_standard] = Analyze_fmt
    close all;
    load Guitar.MAT;
    music = audioread('fmt.wav');
    start_time =[700,  2300 ,14000, 18000, 22000, 25000, 29000,...
        32000, 36000, 40000, 46000, 48000, 56000, 62000, 68000,...
        72000, 76000, 79000, 81000, 83000, 84500, 86500, 90000,...
        94000, 98000,  102000, 106000, 110000, 114500, 119000];
    end_time =  [2300, 14000,18000, 22000, 25000, 29000, 32000,...
        36000, 40000, 46000, 48000, 56000, 62000, 68000, 72000,...
        76000, 79000, 81000, 83000, 84500, 86500, 90000, 94000,...
        98000, 102000, 106000, 110000, 114500, 119000, 131000];

    len = [];
    base = [];
    one_amp = [];
    two_amp = [];
    three_amp = [];
    four_amp = [];
    tone = {};
```

```
    for i = 1:length(start_time)

        [base_uut, one_amp_uut,two_amp_uut,three_amp_uut,four_amp_uut ,tone_uu
t] = ...
        Freq_Analyze( music(start_time(i):end_time(i)), 6, 9);

        leng = ( end_time(i) - start_time(i) )*2 / 4000;
        leng = round(leng) / 2;

        len(i,1) = leng;
        base(i,1) = base_uut;
        one_amp(i,1) = one_amp_uut;
        two_amp(i,1) = two_amp_uut;
        three_amp(i,1) = three_amp_uut;
        four_amp(i,1) = four_amp_uut;
        tone{i,1} = (tone_uut);
    end

    two_standard = two_amp./one_amp;
    three_standard = three_amp./one_amp;
    four_standard = four_amp./one_amp;

    report = table(base, len, two_standard,...
        three_standard, four_standard, tone,...
        'VariableNames', {'Base' 'length' 'two_amp' 'three_amp' 'four_amp' 'To
ne'})

end
```

- Oriental_Red_with_harm_1.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The Tenth Problem: Oriental_Red with harmonic calculated by Problem 5
function Oriental_Red_with_harm_1
    speed = 2;
    sample_rate = 8000;
    len = [1,0.5,0.5,2,1,0.5,0.5,2];
    len = len / speed;

    % Volume
    volume_array =[];
    for i=1:length(len)
       volume_array = [volume_array, generate_volume_for3(len(i),sample_rate)]
;
    end

    %      F(1), G(2), A(3),  B-(4),  C(5),   D(6),   E(7)
    f = [349.23, 392,  440,  466.16, 523.25, 587.33, 659.25];
    tone = [f(5),f(5),f(6),f(2),f(1),f(1),f(6)/2,f(2)];

    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
    % Generate Harmonic Sin Signal

    y = [];
    for i = 1:length(tone)
        t = linspace(0,len(i),len(i)*sample_rate);
        y = [y, [1, 1.4572, 0.95874, 1.0999] * ...
            [sin(2*pi*tone(i)*t); sin(2*pi*2*tone(i)*t);...
            sin(2*pi*3*tone(i)*t); sin(2*pi*4*tone(i)*t)]];
    end
    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

    % y suppressed by volume
    y = y .* volume_array;

    % Make sound
    sound(y, sample_rate);
end
```

- generate_volume_for3.m

```
% return an array of volume strength
function volume_array = generate_volume_for3(len_divide_speed,sample_rate)
    unit = sample_rate * len_divide_speed;
    x1 = linspace(0,len_divide_speed/6,unit/18);
    x2 = linspace(0,len_divide_speed/6,unit/36);
    x3 = linspace(1,1,unit/9);
    x4 = linspace(0,len_divide_speed/3,unit-length([x1,x2,x3]));
    volume_array = [6/len_divide_speed*x1, 1-1.2/len_divide_speed*x2,...
        0.8*x3, 0.8*exp(-(100-90*len_divide_speed)*x4)];
    plot(volume_array);
end
```

- Oriental_Red_with_harm_2.m

```
% Work by Lin, Tzu-Heng
% W42, Dept. of Electronic Engineering, Tsinghua University
% All rights reserved

% The Tenth Problem: Oriental_Red with harmonic calculated by Problem 9
function Oriental_Red_with_harm_2
    speed = 2;
    sample_rate = 8000;
    len = [1,0.5,0.5,2,1,0.5,0.5,2];
    len = len / speed;

    % Volume
    volume_array =[];
    for i=1:length(len)
       volume_array = [volume_array, generate_volume_for3(len(i),sample_rate)]
;
    end

    %      F(1), G(2), A(3),  B-(4),  C(5),   D(6),   E(7)
    f = [349.23, 392,  440,  466.16, 523.25, 587.33, 659.25];
    tone = [f(5),f(5),f(6),f(2),f(1),f(1),f(6)/2,f(2)];

    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
    % Generate Harmonic Sin Signal

    % Get info from fmt
    [base, two_standard, three_standard, four_standard] = Analyze_fmt;

    y = [];
    for i = 1:length(tone)
        t = linspace(0,len(i),len(i)*sample_rate);

        [val, index] = min( abs(tone(i) - base) );

        y = [y, [1, two_standard(index), three_standard(index), four_standard(
index)] * ...
            [sin(2*pi*tone(i)*t); sin(2*pi*2*tone(i)*t);...
            sin(2*pi*3*tone(i)*t); sin(2*pi*4*tone(i)*t)]];
    end
    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

    % y suppressed by volume
    y = y .* volume_array;

    % Make sound
    sound(y, sample_rate);
end
```

- music_syn_gui.m

```
function varargout = music_syn_gui(varargin)
% MUSIC_SYN_GUI MATLAB code for music_syn_gui.fig
%      MUSIC_SYN_GUI, by itself, creates a new MUSIC_SYN_GUI or raises the exi
sting
%      singleton*.
%
%      H = MUSIC_SYN_GUI returns the handle to a new MUSIC_SYN_GUI or the hand
le to
%      the existing singleton*.
%
%      MUSIC_SYN_GUI('CALLBACK',hObject,eventData,handles,...) calls the local
%      function named CALLBACK in MUSIC_SYN_GUI.M with the given input argumen
ts.
%
%      MUSIC_SYN_GUI('Property','Value',...) creates a new MUSIC_SYN_GUI or ra
ises the
%      existing singleton*.  Starting from the left, property value pairs are
%      applied to the GUI before music_syn_gui_OpeningFcn gets called.  An
%      unrecognized property name or invalid value makes property application
%      stop.  All inputs are passed to music_syn_gui_OpeningFcn via varargin.
%
%      *See GUI Options on GUIDE's Tools menu.  Choose "GUI allows only one
%      instance to run (singleton)".
%
% See also: GUIDE, GUIDATA, GUIHANDLES

% Edit the above text to modify the response to help music_syn_gui

% Last Modified by GUIDE v2.5 23-Jul-2016 12:47:00

% Begin initialization code - DO NOT EDIT
gui_Singleton = 1;
gui_State = struct('gui_Name',       mfilename, ...
                   'gui_Singleton',  gui_Singleton, ...
                   'gui_OpeningFcn', @music_syn_gui_OpeningFcn, ...
                   'gui_OutputFcn',  @music_syn_gui_OutputFcn, ...
                   'gui_LayoutFcn',  [] , ...
                   'gui_Callback',   []);
if nargin && ischar(varargin{1})
    gui_State.gui_Callback = str2func(varargin{1});
end

if nargout
    [varargout{1:nargout}] = gui_mainfcn(gui_State, varargin{:});
else
    gui_mainfcn(gui_State, varargin{:});
```

```
end
% End initialization code - DO NOT EDIT

% --- Executes just before music_syn_gui is made visible.
function music_syn_gui_OpeningFcn(hObject, eventdata, handles, varargin)
% This function has no output args, see OutputFcn.
% hObject    handle to figure
% eventdata  reserved - to be defined in a future version of MATLAB
% handles    structure with handles and user data (see GUIDATA)
% varargin   command line arguments to music_syn_gui (see VARARGIN)

set(handles.listbox1, 'string', {'guitar', 'violin', 'piano'});

% Choose default command line output for music_syn_gui
handles.output = hObject;

% Update handles structure
guidata(hObject, handles);

% UIWAIT makes music_syn_gui wait for user response (see UIRESUME)
% uiwait(handles.figure1);

% --- Outputs from this function are returned to the command line.
function varargout = music_syn_gui_OutputFcn(hObject, eventdata, handles)
% varargout  cell array for returning output args (see VARARGOUT);
% hObject    handle to figure
% eventdata  reserved - to be defined in a future version of MATLAB
% handles    structure with handles and user data (see GUIDATA)

% Get default command line output from handles structure
varargout{1} = handles.output;

% --- Executes on button press in pushbutton1.
function pushbutton1_Callback(hObject, eventdata, handles)
% hObject    handle to pushbutton1 (see GCBO)
% eventdata  reserved - to be defined in a future version of MATLAB
% handles    structure with handles and user data (see GUIDATA)

tones = str2num(get(handles.edit1,'string'));
len = str2num(get(handles.edit2,'string'));

playmusic(tones, len);

% --- Executes on selection change in listbox1.
function listbox1_Callback(hObject, eventdata, handles)
% hObject    handle to listbox1 (see GCBO)
% eventdata  reserved - to be defined in a future version of MATLAB
% handles    structure with handles and user data (see GUIDATA)
```

```
% Hints: contents = cellstr(get(hObject,'String')) returns listbox1 contents a
s cell array
%        contents{get(hObject,'Value')} returns selected item from listbox1

% --- Executes during object creation, after setting all properties.
function listbox1_CreateFcn(hObject, eventdata, handles)
% hObject    handle to listbox1 (see GCBO)
% eventdata  reserved - to be defined in a future version of MATLAB
% handles    empty - handles not created until after all CreateFcns called

% Hint: listbox controls usually have a white background on Windows.
%       See ISPC and COMPUTER.
if ispc && isequal(get(hObject,'BackgroundColor'), get(0,'defaultUicontrolBack
groundColor'))
    set(hObject,'BackgroundColor','white');
end

function edit1_Callback(hObject, eventdata, handles)
% hObject    handle to edit1 (see GCBO)
% eventdata  reserved - to be defined in a future version of MATLAB
% handles    structure with handles and user data (see GUIDATA)

% Hints: get(hObject,'String') returns contents of edit1 as text
%        str2double(get(hObject,'String')) returns contents of edit1 as a doub
le

% --- Executes during object creation, after setting all properties.
function edit1_CreateFcn(hObject, eventdata, handles)
% hObject    handle to edit1 (see GCBO)
% eventdata  reserved - to be defined in a future version of MATLAB
% handles    empty - handles not created until after all CreateFcns called

% Hint: edit controls usually have a white background on Windows.
%       See ISPC and COMPUTER.
if ispc && isequal(get(hObject,'BackgroundColor'), get(0,'defaultUicontrolBack
groundColor'))
    set(hObject,'BackgroundColor','white');
end

function edit2_Callback(hObject, eventdata, handles)
% hObject    handle to edit2 (see GCBO)
% eventdata  reserved - to be defined in a future version of MATLAB
% handles    structure with handles and user data (see GUIDATA)

% Hints: get(hObject,'String') returns contents of edit2 as text
%        str2double(get(hObject,'String')) returns contents of edit2 as a doub
le
```

```
% --- Executes during object creation, after setting all properties.
function edit2_CreateFcn(hObject, eventdata, handles)
% hObject    handle to edit2 (see GCBO)
% eventdata  reserved - to be defined in a future version of MATLAB
% handles    empty - handles not created until after all CreateFcns called

% Hint: edit controls usually have a white background on Windows.
%       See ISPC and COMPUTER.
if ispc && isequal(get(hObject,'BackgroundColor'), get(0,'defaultUicontrolBack
groundColor'))
    set(hObject,'BackgroundColor','white');
end

% return an array of volume strength
function volume_array = generate_volume_for3(len_divide_speed,sample_rate)
    unit = sample_rate * len_divide_speed;
    x1 = linspace(0,len_divide_speed/6,unit/18);
    x2 = linspace(0,len_divide_speed/6,unit/36);
    x3 = linspace(1,1,unit/9);
    x4 = linspace(0,len_divide_speed/3,unit-length([x1,x2,x3]));
    volume_array = [6/len_divide_speed*x1, 1-1.2/len_divide_speed*x2,...
        0.8*x3, 0.8*exp(-(100-90*len_divide_speed)*x4)];
    % plot(volume_array);

function playmusic(tones, len)
    load guitar.mat

    speed = 2;
    sample_rate = 8000;
    len = len / speed;

    % Volume
    volume_array =[];
    for i=1:length(len)
       volume_array = [volume_array, generate_volume_for3(len(i),sample_rate)]
;
    end

    %      F(1), G(2), A(3),  B-(4),  C(5),   D(6),   E(7)
    f = [349.23, 392,  440,  466.16, 523.25, 587.33, 659.25];
    tone = f(tones);

    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
    % Generate Harmonic Sin Signal

    y = [];
    for i = 1:length(tone)
```

```
        t = linspace(0,len(i),len(i)*sample_rate);

        [val, index] = min( abs(tone(i) - base) );

        y = [y, [1, two_standard(index), three_standard(index), four_standard(
index)] * ...
            [sin(2*pi*tone(i)*t); sin(2*pi*2*tone(i)*t);...
            sin(2*pi*3*tone(i)*t); sin(2*pi*4*tone(i)*t)]];
    end
    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

    % y suppressed by volume
    y = y .* volume_array;

    % Make sound
    sound(y, sample_rate);
```