```
#include <stdio.h>
#include <math.h>
#define SIZE  (8*44100)
int main() {
  FILE *fp2;
  unsigned char
       h[44]={0x52,0x49,0x46,0x46,   /* 識別子 RIFF を ascii code で */
              0x44,0x62,0x5,0x0,     /* ファイルサイズ-8. [可変] */
              0x57,0x41,0x56,0x45,   /* 識別子 WAVE を ascii code で */
              0x66,0x6d,0x74,0x20,   /* fmt を ascii code で */
              0x10,0x0,0x0,0x0,      /* linear PCM */
              0x1,0x0,               /* linear PCM */
              0x2,0x0,               /* stereo */
              0x44,0xac,0x0,0x0,     /* sampling rate = 44100 = 0xac44 */
              0x10,0xb1,0x2,0x0,     /* byte per second, 44100*4 */
              0x4,0x0,               /* 16 bit, stereo */
              0x10,0x0,              /* bit/sample, 16 bit */
              0x64,0x61,0x74,0x61,   /* 識別子 data を ascii code で */
              0x20,0x62,0x5,0x0};    /* 以下のデータ部分のファイルサイズ. [可変] */
                                     /* 2 秒分のデータなので決め打ち */
  unsigned char data[SIZE];  // 2秒のデータ.
  int c;
  int i;
  int filesize = (44+SIZE)-8;
  int datasize = SIZE;
  int w;
  double t;

  /* data に音データを書き込む, 440Hz の sin 波 */
  for (i=0; i<SIZE; i += 4) {
    t = ((double)(i/4))/44100.0;
    w = (int) 3000*sin(2*3.1415*440*t); // 440 Hz の sin 波
    if (w < 0) w = w + 0x10000;       // 補数表現へ
    data[i] = data[i+2] = w % 0x100;  // data[i] が左, data[i+2]が右
    data[i+1] = data[i+3] = w / 0x100;
  }

  /* データをファイル mysound.wav へ書き込む. */
  fp2 = fopen("mysound.wav","w");
  for (i=0; i<44; i++) fputc(h[i],fp2);
  for (i=0; i<SIZE; i++) fputc(data[i],fp2);
  fclose(fp2);
}
```



```
/*
  WAVE format     :   http://www.kk.iij4u.or.jp/~kondo/wave/
*/
```

```c
 /* cc genwav4c.c -lm,  和音 */
#include <stdio.h>
#include <math.h>
/* 5 秒 */
#define SIZE  (4*44100*5)
int cegL(unsigned char data[],int level,int from,int length);

int main() {
  FILE *fp2;
  char h[44]={0x52,0x49,0x46,0x46,  //RIFF
              0x84,0x56,0x8,0x0,        // ファイルのサイズ-8 ***
              0x57,0x41,0x56,0x45,  //WAVE
              0x66,0x6d,0x74,0x20,  //fmt
              0x10,0x0,0x0,0x0,     // linear PCM
              0x1,0x0,              // linear PCM
              0x2,0x0,              // stereo
              0x44,0xac,0x0,0x0,    // sampling rate 0xac44=44100
              0x10,0xb1,0x2,0x0,    // byte per second, 44100*4
              0x4,0x0,              // 16 bit stereo
              0x10,0x0,             // bit/sample
              0x64,0x61,0x74,0x61,  // data
              0xb8,0x55,0x8,0x0};   // data部分のサイズ.
  unsigned char data[SIZE];
  int c;
  int i;
  int filesize = (44+SIZE)-8;
  int datasize = SIZE;
  fp2 = fopen("mysound.wav","w");
  /* ファイルサイズを自動計算 */
  h[4] = filesize % 0x100; h[5] = (filesize/0x100) % 0x100;
  h[6] = (filesize/0x10000) % 0x100; h[7] = (filesize/0x1000000) % 0x100;
  h[40] = datasize % 0x100; h[41] = (datasize/0x100) % 0x100;
  h[42] = (datasize/0x10000) % 0x100; h[43] = (datasize/0x1000000) % 0x100;

  /* set data in the array data */
  for (i=0; i<SIZE; i++) data[i]=0;
  cegL(data,2000,0,4*44100);  //0秒から4秒間, ドミソの和音

  for (i=0; i<44; i++) fputc(h[i],fp2);
  for (i=0; i<SIZE; i++) fputc(data[i],fp2);
  fclose(fp2);
}
```



```c
int cegL(unsigned char data[],int level,int from,int length) {
  int i,p;
  int w;
  if (4*(from+length) >= SIZE) {
    fprintf(stderr,"Error\n"); return(-1);
  }
  for (i=from; i< from+length; i++) {
    /* 和音を作るには ドの周波数 523.25, ミの周波数 659.26, ソの周波数 783.99
       を重ねた sin 波を作れば良い.
    */
    w = (int) level*(sin(2*3.14*523.25*((double) i)/44100.0)
                    +sin(2*3.14*659.26*((double) i)/44100.0)
                    +sin(2*3.14*783.99*((double) i)/44100.0));
    p = i*4;
    /* printf("%d\n",w); */
    if (w < 0) w = w+0x10000;
    data[p] = w % 0x100;
    data[p+1] = w/0x100;
  }
  return(0);
}

/*
  音階と周波数の関係を調べるには, [音階 周波数] 検索
*/
```