ヘッドホンではコード 1 の音を再生しないように!

コード 1: ホワイトノイズ, genwav.c

```
#include <stdio.h>
#include <stdlib.h>
#define SIZE  (4*44100)     // 1秒の音データ.
int main() {
  FILE *fp2;
  unsigned char
   h[44]={0x52,0x49,0x46,0x46,    /* 識別子RIFFをascii codeで*/
          0x84,0x56,0x8,0x0,      /* 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*/
          0xb8,0x55,0x8,0x0};     /* データ部分のサイズ.[可変]*/
  unsigned char data[SIZE];
  int c;
  int i;
  int filesize = (44+SIZE)-8;
  int datasize = SIZE;
  fp2 = fopen("mysound.wav","w");
  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/0
    x1000000) % 0x100;

  /* set data in the array data */
  for (i=0; i<SIZE; i++) data[i] = random()%0x100;

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

/*
  WAVE ファイルの形式 [WAVE file format] で検索.
*/
```

コード 2: sin 波,genwav0.c

```
#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};     /* データ部分のサイズ. [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;
```

```
23    int i;
24    int filesize = (44+SIZE)-8;
25    int datasize = SIZE;
26    int w;
27    double t;
28
29    /* data に音データを書き込む, 440Hz の sin 波 */
30    for (i=0; i<SIZE; i += 4) {
31      t = ((double)(i/4))/44100.0;
32      w = (int) 3000*sin(2*3.1415*440*t);//440 Hz の sin 波
33      if (w < 0) w = w + 0x10000;    //補数表現へ
34      //data[i] が左, data[i+2]が右
35      data[i] = data[i+2] = w % 0x100;
36      data[i+1] = data[i+3] = w / 0x100;
37    }
38
39    /* データをファイル mysound.wav へ書き込む. */
40    fp2 = fopen("mysound.wav","w");
41    for (i=0; i<44; i++) fputc(h[i],fp2);
42    for (i=0; i<SIZE; i++) fputc(data[i],fp2);
43    fclose(fp2);
44  }
45
46  /*
47    WAVE format     :   http://www.kk.iij4u.or.jp/~kondo/wave/
48  */
```

Risa/Asir ドリル 2022, 27 章 “音ファイルの生成”.