The Science of Digital Media by Jennifer Burg

Errata

Items marked in yellow have been corrected.

Chapter 1

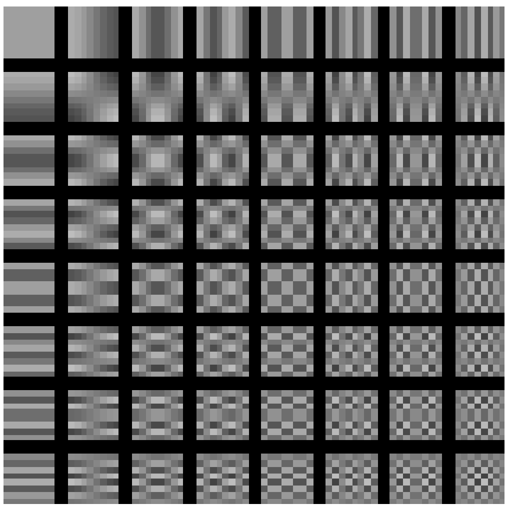
p. 16

> Let f be the frequency of a sine wave. Let r be the minimum sampling rate that can be used in the digitization process such that the resulting digitized wave is not aliased. Then
>
> $$r = 2f$$
>
> f is called the ***Nyquist frequency***.

Chapter 2

p. 76

There are only seven rows of frequency components in Figure 2.25. The figure should look like this:



Chapter 4

p. 219

$$f(t) = \sum_{n=-\infty}^{\infty} F_n e^{in\omega t}$$

$$= \sum_{n=-\infty}^{\infty} \left[F_n \left(\cos(n\omega t) + i \sin(n\omega t) \right) \right] \qquad Step\,1$$

$$= \sum_{n=-\infty}^{-\infty} \left[(a_n - ib_n)(\cos(n\omega t) + i \sin(n\omega t)) \right] \qquad Step\,2$$

The summation in Step 2 goes to infinity

p. 220

Step 8.

The fourth term in the summation should be positive: $+b_{-n} \sin(-n\omega t)$

Step 9. $a_n = a_{-n}$, $-\boldsymbol{b}_n = \boldsymbol{b}_{-n}$, $\cos(-n) = \cos(n)$, and $\sin(-n) = -\sin(n)$.

p. 221

$$a_n = \frac{1}{N}\sum_{k=0}^{N-1} f_k \cos\left(\frac{2\pi nk}{N}\right) \text{ and } b_n = \frac{1}{N}\sum_{k=0}^{N-1} f_k \sin\left(\frac{2\pi nk}{N}\right)$$

b_n is based on the *sin* function, not *cos*.

p. 233

line 26

just a second requires 176,400 bytes

Deleted: half

p. 236

line 22

More commonly, a MIDI cable connects

Deleted: that

p. 245

4th line from the bottom

turn up the volume, or create some other effect

Deleted: or

Chapter 5

p. 259

line 8

color coded three-jack connections where the yellow plug

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p. 261

middle of page

Four common plug-in formats are VST...

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Formatted: Normal

p. 284

Indent last paragraph on the page.

p. 300

```
algorithm FIR_low_pass filter
/*Input:    f_c, the cutoff frequency for the lowpass filter, in Hz
            f_samp, the sampling frequency of the audio signal to be filtered, in Hz
            N, the size of the filter; assume N is odd
  Output:  a low-pass FIR filter in the form of an N-element array */
{
   /*Normalize f_c and ω _c so that π is equal to the Nyquist angular frequency*/
   f_c  =  f_c/f_samp
   ω _c  =  2*π*f_c
   middle = N/2      /*Integer division, dropping remainder*/
   /*Create the filter using the low-pass filter function from Table 1*/
   for n = −N/2 to N2
      if (n = 0) fltr(n+middle) = 2*f_c
      else  fltr(n+middle) = sin(ω _c *n)/(π*n)
   /*Multiply the elements of fltr by a windowing function chosen from Table 2.  We
use the Hanning window function/
  for n = −N-1 to N-1
       fltr(n+middle) = fltr(n+middle) * (0.5 + 0.5*cos((2*π*n)/N))
}
```