COMPUTER AUDIO DESIGN

Sampling Frequency & Bit Depth

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Definitions

The first thing to realise is that these two parameters, “sampling rate” and “bit depth”, are completely independent.

As an example:

A CD has a sampling frequency of 44.1kHz and a bit depth of 16 bits.

Sampling frequency is how many times per second a continuous signal (analog) was "recorded" or sampled to make a discrete or digital signal. With a CD this is 44,100 times per second.

Bit depth is how many values are available for each sample. Bit depth is calculated by 2^N. Where N is the number of bits. So a CD has 2^{16} = 65,536 different values available for each sample.

Are both Sampling Frequency and Bit Depth equally important to sound quality in home audio systems? Let’s start with bit depth.

Bit Depth

A *Bit* is the abbreviation for a single “binary digit”, represented by a 0 or a 1. For example, here is a 16-bit binary number:

0110111110111010

The right most bit is called Bit 0 and the left most Bit is called Bit 15. 0 through 15 equals a total of 16 bits.

The left most bit is called the "Most Significant Bit" (MSB) and is equal to 2^{N-1}, where N = the bit number, which in this case N = 16.

The second Bit is the 2nd left most bit and is equal to 2^{N-2}

The third Bit is the 3nd left most bit and is equal to 2^{N-3}

The fourth Bit is the 4nd left most bit and is equal to 2^{N-4}

And so on…….

The 16th Bit is the right most bit and is equal to 2^{N-N} or $2^0 = 1$

In the above 16-bit binary number Bits 15, 12, 6, 2 and 0 are all zero, so these Bit values are equal to zero and contribute nothing to the output.

So the 16-Bit Binary number above is equal to:

$$2^{14} + 2^{13} + 2^{11} + 2^{10} + 2^9 + 2^8 + 2^7 + 2^5 + 2^4 + 2^3 + 2^1$$

Which equals:

16,384 + 8192 + 2048 + 1024 + 512 + 256 + 128+ 32 + 16 + 8 + 2 = 28,602

If all the Bits in the 16-Bit binary number were equal to 1's like this: 1111111111111111 That would be the maximum output of the device, which is 2^{16} = 65,536.

If all the bits in the 16-Bit binary number where equal to 0's like this: 0000000000000000 That would be the minimum output of the device, which is 0.

This means with a 16 bit system we have 65,536 individual values available. In an "ideal" 16 Bit DAC the DAC can output 65,536 different values.

Here are how the Bit numbers relate to the DAC output values:

1) The "Most Significant Bit" (MSB) is equal to half of the maximum output of the DAC.

2) The next (2nd) significant bit will be half of the MSB.

3) The third will be half of the 2nd MSB and so on.

4) The "Least Significant Bit" (LSB) can be calculated by this equation: max output of DAC / 2^N, where N is the number of bits the DAC has.

Think of the MSB as the coarse tuning knob on a radio and the LSB as the fine tuning knob on a radio.

To make it easier assume we have a 16 bit "ideal" DAC Integrated Circuit (IC). Some DACs output voltage, others output current. Let's assume our "ideal" DAC outputs current.

Let’s also assume the maximum output of our 16 bit "ideal" DAC is 5 milliamps (a typical value). The value of each bit is then:

Bit 15 (MSB) = 2.50 mA
Bit 14 = 1.25 mA
Bit 13 = 0.625 mA
Bit 12 = 0.3125 mA
Bit 11 = 0.015625 mA
Bit 10 = 78.125 uA
Bit 9 = 39.0625 uA
Bit 8 = 19.5313 uA
Bit 7 = 9.76563 uA
Bit 6 = 4.88281 uA
Bit 5 = 2.44141 uA
Bit 4 = 1.22070 uA
Bit 3 = 0.610352 uA
Bit 2 = 0.305176 uA
Bit 1 = 0.152588 uA
Bit 0 (LSB) = 0.0762939 uA

So the LSB involves a 0.0762929 uA of current change! That is 76.3 x 10^{-9} AMPS!

In a 24 bit system where the maximum output of the DAC IC is 5mA the LSB will be 2.98 x10^{-10} amps.

T o help you understand how small this value is say our maximum output is a distance equal to 10 miles. In a 24 bit system the LSB is equal to: 0.03777 inches! We are talking *very* small here!

I have chosen 5mA as the maximum output of the DAC. Some DACs will have a higher maximum output which of course increases the value of the LSB. But as the value of the LSB increases the resolution of the DAC decreases. Many DACs have a maximum output that is less than 5mA - meaning the LSB value will be even smaller.

Assume we have a 24 bit DAC with a maximum output level of 15mA. The LSB will still have a value of 8.94 x 10^{-10} amps.

In a perfect world the LSB should be as small as possible because it would allow a higher resolution and better sound quality. Unfortunately, the world is not perfect, there is noise and in addition to that, unfortunately, “jitter”. I do not want to go into what jitter is in this discussion, but it is related to timing errors. These timing errors also decrease bit resolution.

An issue I see with bit depths that are 24 bit (or higher) is that the LSBs are so small that the LSBs can easily drop below system noise level. In other words, it is difficult enough to get full 16 bit resolution.

The one other issue of 24 bit over 16 bit files is that a 24 bit file takes more space to store and computer power and resources to playback than a 16 bit file.

I do not want to get any more technical in this discussion but as soon as you start looking at how noise, jitter and frequency affect the LSBs you can quickly see how we can get in trouble.

Being an electrical engineer myself I would not want to be the person responsible for designing a circuit where the system noise and jitter levels must be low enough to take full advantage of the LSB’s in a 24 bit system. The LSBs of a 16 bit system is hard enough!

Sampling Frequency

For a DAC, ***Sampling Frequency*** is an incorrect term, even though we all use it. An Analogue to Digital Converter (ADC) takes "samples" at specific intervals (frequencies). A DAC has a maximum rate (frequency) that it can accept data. What we are really talking about when we say, "DAC Sampling Frequency" is the "speed" or "throughput rate" at which the DAC can accept data.

DAC chips are specified for what their maximum throughput rate is, and this varies greatly. But the key issue here is that unlike bit depth, throughput rates up to 192kHz and higher do not pose a difficult technical issue for the audio DAC designer if the appropriate DAC is chosen. Higher throughput rates can of course cause issues such as increased noise and jitter, but nothing that is comparable to the difficulties cause by the LSBs of 24 bit depth.

Some down sides of Higher Sampling Rates:

It is a fact that higher sample rate music requires more computer power and “resources” to play. This is because the computer has more processing work to do. It is CAD’s belief than when you minimize the amount of power and processing that is required for playback of music the amount of high frequency noise and jitter is also minimized and that the sound quality changes. The DAC chip itself will also have to dissipate more heat itself for higher sampling rate music.

The most important thing of them all is to please trust your ears!

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