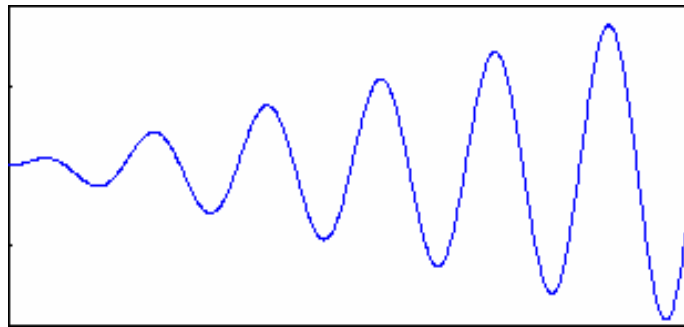


BITS IN TIME

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Reading the various forums and Internet browsing, I have seen that many audio enthusiasts are not very clear in all processes of the digital data transmission and playing of the digital records. This short article will provide a very short description for the non technical peoples, as much as possible simply stated, without the "triple integrals and Fourier-analysis. Its purpose is to give a general idea of what is "bit perfect" and "jitter", because of a lot of myths and esoteric affairs, has been surrounding the audio high end.

As we know, an analog audio signal can be present as a function of time or the graph for the voltage, current, sound pressure, etc. When we are talking about the signals transmission, usually we consider the voltage:

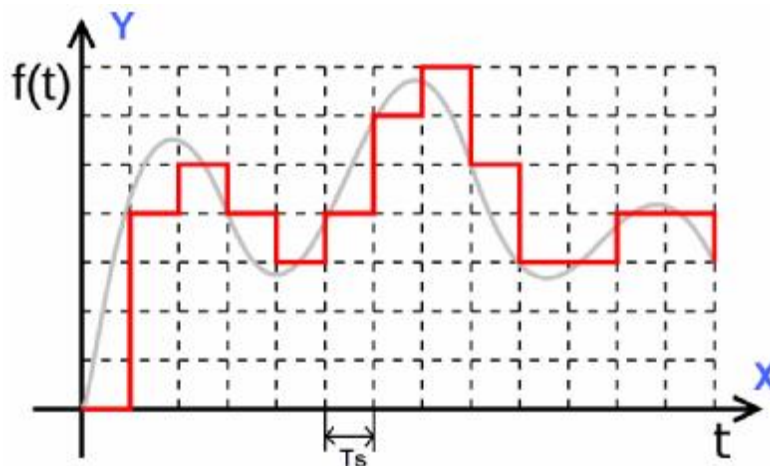


<pic. 1 Analog Signal Graph>

Digital audio signal is a similar graph, but with discrete data. Each point of this graph is a specific data value, equal the voltage at a particular point in time. This is called "count".

The frequency of these count usually called F_s , and the time between the count – sample interval, or T_s ($T_s = 1/F_s$).

So, we can see that both digital and analog signals are characterized by two coordinates - time (t), and the value ($f(t)$) at that time (count), or the X and Y direction.



<pic. 2 Discrete Signal Graph >

The main advantage of the digital (discrete), is the fact that they can be stored, copied and transmit to any distance absolutely without distortion, in an unchanged form. It should be understood the difference between the "storage", "transfer" ("copy") and "play".

For storage, it is sufficient to record the data on the storage media (floppy disk, hard disk drive, flash memory, RAM, etc.) keeping the exact data (Y) and to put it in a strict order. The X coordinate is not stored, it is assumed that the time between the two neighbouring data is equal to T_s . Because the same data can be played back at different speeds, it is essential to know in advance the frequency sampling rate (F_s). Typically, it is stored in the file header (when stored on disk) or it is expected the default (the Audio-CD it equals 44.1kHz).

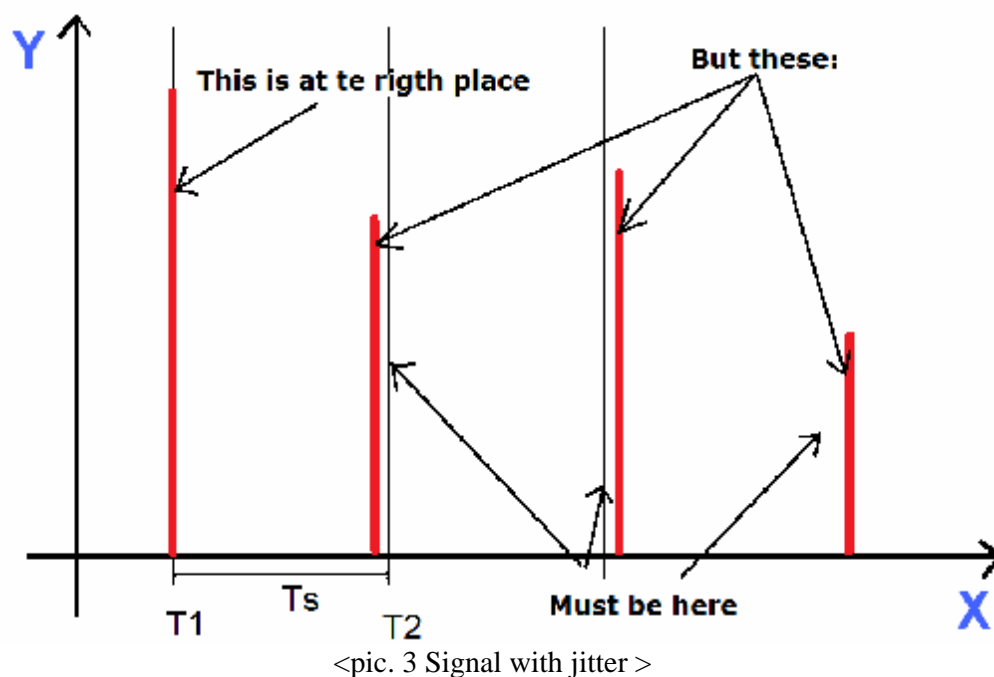
To copy the data, it is sufficient only to copy them from one place to another, while maintaining the value and the order. You can heard mane esoteric stories, when the “audio” is corrupting after the long distance transmission (internet) , or when you save it at one PC to the flash disc and then read the flash on another PC. Comments are unnecessary....

But, everything have changed when you play the digital record! That means that you transfer the digital discrete signal to the analog form, and it is not so easy! Of course, the data you must to transfer correctly - if at this moment the data should be 1247, you should transmit (and receive!) also 1247. Not 1246, not 1248, not 2759 - only 1247!

This is called "bit prefect". But this is not enough! You need than this digit come to the digital-to-analog converter (DAC) exactly at the right time, as defined by sampling rate.

People forget about this, when asking the questions like "why the sound is corrupted, there are the same 0 and 2?!"

Yes, there are the same “0 and 1” but that have to be converter at the right time. If one come at the time T_1 , then the following should do at the time $T_2 = T_1 + T_s$



So, we have 2 requirements to the T_s :

- 1) T_s should be equal to your original T_s with high accuracy. Usually, a time interval is generated by master clock, with the required frequency dividers. Its duration must be equal to the duration interval, used during the record. Since two identical generators do not often, their frequency should be as close as possible, otherwise, there will be will a audible changes of pace and tone. (Who remember the tape recorders - if the record was done at 9.53cm/s and played at 19.05cm/s , the sound will be twice faster and have a twice higher tone, but this is an extreme case. In a poorly adjusted tape recorder, even if the speed is slightly different - this is well audible).

Fortunately, the frequency accuracy of even the bad ceramic/quartz oscillator is no more than 100-200ppm, i.e. in worst case the difference 2 generator's frequencies (deviations are maximum and with opposed sign) is not more than 400ppm.

Those who are not aware that is a "ppm" - 1000ppm this is only 0.1 %, and even a professional musician or the conductor, with the absolute pitch, cannot detect the difference in tone less the few percents (10000ppm). Most oscillators have 20-50ppm frequency accuracy (that for the audio purposes is more than enough), although you can find the special "audiophile" generators at the market, with advertised accuracy 0.1 -1ppm.

(What I can say, to earn money is not prohibited...)

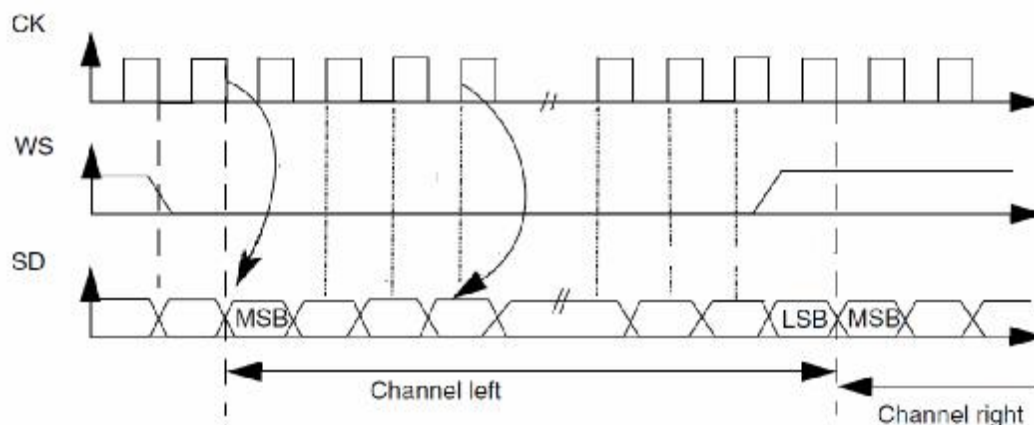
2) T_s must be a constant, i.e. , all the same. In practice, the oscillator's accuracy characterizes the mean value of the frequency for a while. But no matter how precise was the alternator, it's always a certain frequency deviation, which random changes:

$$T_s = T_{s_const} + T_{s_var},$$

where T_{s_const} - the average value of the frequency, T_{s_var} - the variable component.

This is called "phase jitter" (again about the tape recorders - jitter slightly recalls detonation), and in contrast to the simple deviation (accuracy), it is no longer about tens-hundreds of ppm, or tenths of one per cent. Jitter audibility is characterized by not only by its value (in hertz, nanoseconds, etc.) but also the spectral distribution. A lower frequency jitter components are more audible than the higher. The maximum jitter depends on the sampling rate and the signal resolution. For standard AudioCD (44.1 / 16) format non audible jitter can reach tens of nanoseconds. For high-resolution formats (96/24, 192/24), the maximum non audible jitter value is reduced to units/parts of the nanoseconds.

There is one small nuance - the jitter impact at the time of the transmission and during the digital-to-analog conversion. The transfer is usually done on a synchronous bus, where data is transmitted at data lines and strobed by the clock signal (all audio buses, like I2S, LJ, RJ, in fact, are the kind of SPI bus).



<pic. 4 I2S>

As can be seen in the pic.4, the data (SD) is clocked by positive edge of the clock signal (CK). In order to transmit the data correctly, without distortion, the edge must be "inside" of the data interval. If the jitter's value is more then $T_s/2$ it will clock another bit. This is a very huge jitter, that does not in reality occurs.

However, when playing, as mentioned above, the maximum allowed jitter is much less. At the last stage, the data coming to the DAC and are converted to the analog.

In fact, the two processes are connected together, so the large jitter here is not allowed.

Nevertheless, many digital sound sources have not good jitter characteristics – the cheap quartz oscillators can be used, it is often uses one quartz, then the necessary frequencies are obtained from the PLL circuit. A large jitter injection can be occurred at the digital transmission with self clocked bi-phase coded, via SPDIF (coax) and Toslink (optical) interfaces, because the clock is recovered from this signal by PLL circuit at the receiver.

Now, let's look at a few methods of the jitter reduction or elimination. For the full descriptions of all methods a lot of pages needs for each of them, therefore, I will make only the small observation:

- 1) The data transmission from the source to the DAC through the I2S bus, bypassing the SPDIF/Toslink encoder-decoder. All errors associated with the signal encoding and signal reconstruction will be absent, and the jitter will depends only on the clock source quality (which, if necessary, can be replaced by a more qualitative). Using a standard CD/DVD players and PC sound cards, sufficiently serious hardware changes might be necessary.
- 2) The data transmission with the I2S bus or SPDIF/Toslink, with the external clocking by the high quality oscillator. The better place for this oscillator is near the DAC (because it is the DAC, who is sensitive to jitter) and to send this clock to the transmitter. This excludes the source generators' influence to the source signal, as well as the jitter in the input/output buffers, cable connection. Of course, it also requires the sources hardware changes, not for all sources it is possible.
- 3) Using of more modern SPDIF receivers with reduced jitter (for example, DIR9001 instead of CS84xx). Is not the best method, but it is one of the simplest, not require the source signal device penetration.
- 4) Re-clock - digital data at the DAC input are transferred through the trigger, where they are latched by the high quality oscillator's clock. Does not require the source device penetration and hardware changes, also very often is present in the high quality DACs.
- 5) Using of the special chips (for example SRC4192, AD1890), which are designed to convert the signals from one sampling rate to another. Their special feature in that output data are clocked from the DAC oscillator, asynchronously to the input clock. Input and output frequency can be the same - in this case, no re-sampling occurs, but there is a jitter reduction. The disadvantage – the “bit-perfect” can be corrupted during the digital signal processing inside the chip.
- 6) There are special "clock signal cleaners" chips that reduced the jitter. But they are expensive, and their effectiveness for audio technology is a big question.
- 7) Using of the modern types of media and playback devices - media players, network players, solid state (HDD, flash) players, USB interfaces to computers. Many of them provide an external clocking from the DAC oscillator.
- 8) Using of the buffer memory (FIFO) technology - signal from the source (they can be any type of signal - SPDIF, TOSLINK, USB, I2S) is stored in the circle memory. Reading the memory occurs according to high quality clock signal from DAC oscillator with some delay. Memory is needed to compensate for the frequency difference of the DAC and source oscillators. As in all methods, in which the transfer is clocked by the DAC's oscillator, there is a total source jitter elimination.

As you can see, some of these methods are suitable only for the new types of sound source devices, others may be used also with the old ones (and FIFO method will not require even its penetration and modification). Some methods can only to reduce the source's signals jitter, the other has completely eliminates the source jitter influence, minimizing the jitter to the high quality oscillator's ones. For the modern DS DAC chips only MCLK signal is important, to which they are particularly sensitive.