How digital filters affect analog audio signal levels

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Introduction

Digital audio processing provides a great amount of flexibility to system designers. Multiple filter structures can be cascaded to form equalization (EQ), low-pass, high-pass, shelf, and many other filter combinations with relatively low power consumption and little PCB space. Infinite-impulse-response (IIR) filters can be used to easily simulate filter functions performed by analog counterparts.

Digital audio signals are represented as an array of bits with a fixed resolution. This means that the signal is discrete in nature, both in amplitude and in time. If the source of this data is analog, it is quantized and sampled at fixed intervals (sampling periods) by an analog-to-digital converter (ADC). An audio engineer has to be careful to ensure that the signal being recorded is not clipped, while maintaining the signal as loud as possible to maximize the signal-to-noise ratio (SNR). An ADC has an amplitude limit, which may be defined as a full-scale voltage, meaning that any signal above a certain amount of volts at the converter's input may result in clipping. Also, the signal is quantized into a number with a certain fixed resolution (which might also be close to the clipping point).

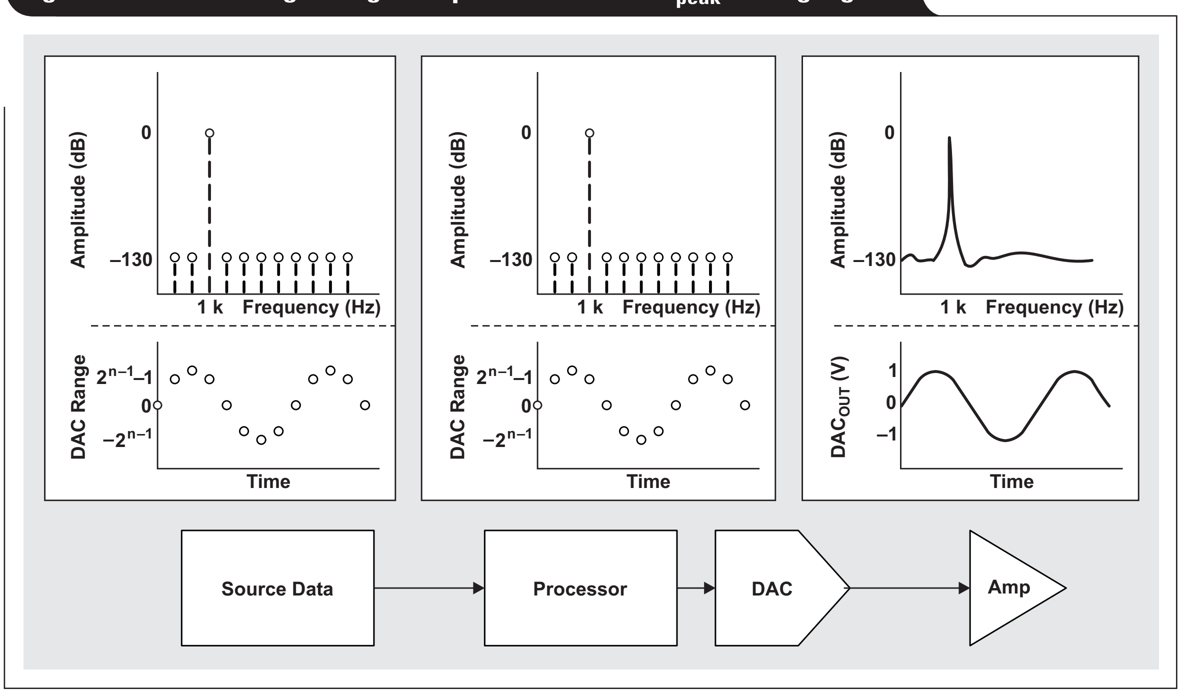
A mastering engineer also has to be careful when working with music in the digital domain. Some EQ can be added to boost certain frequencies as well as to achieve other effects. If there is an extravagant amount of headroom, the engineer can boost, boost, and boost. However, the final medium for the music is a CD (which is limited to 16-bit audio), so trade-offs have to be made because the rest of the music sounds too "quiet" compared to the SNR of the medium.

Boosting frequency bands in the digital domain can create certain problems when the bands are converted into the analog domain. Digital-to-analog converters (DACs) can also clip the signal if their digital input is larger than their full-scale voltage. Most processors allow a certain amount of headroom to work with intermediate values, but ultimately a DAC expects a certain data width bounded by maximum and minimum values. If the signal is scaled down and certain frequencies are then boosted to accommodate the higher peaks, then the SNR at the flat region(s) will suffer from a lower SNR.

Understanding the problem

A very important aspect to consider when digital filters are used is how the signal level is affected upon its conversion from the digital to the analog domain. Suppose that a system provides a digital signal to a processing unit and converts it to analog using an ideal DAC without any processing being applied, as shown in Figure 1. In this example, a 0-dBFS digital signal is provided to the DAC and converted

Figure 1. Full-scale digital signal represented as a 1-V_{peak} analog signal



into the analog domain. A relationship between the digital code and the analog output amplitude is provided in the specification of a codec as the full-scale amplitude. If the specification of the full-scale amplitude is 0.707 V_{RMS} (or 1 V_{peak}), this means that a full-scale 0-dBFS digital sinusoid will result in a 1-V_{peak} sinusoid, as shown in the figure.

If a DAC is bounded by -2^{n-1} and $2^{n-1} - 1$, amplifying a signal beyond these limits will distort the signal by clipping it at its output (assuming saturation logic), as shown in Figure 2. Note that it is typical for most signal processors to allow some amount of headroom before providing data to a DAC. It is important for the data within the processor's memory to remain undistorted. Figure 2 illustrates the DAC input limits where output clipping may occur if exceeded.

A solution to this problem would be to ensure that the signal is not amplified beyond the DAC's limits (i.e., ensure that positive gain is not applied to the source signal). However, there are cases for which the solution is not as obvious. Performing a boost relative to the full-scale amplitude of the DAC input at a specific frequency range will also cause adverse effects. In Figure 3, a 500-Hz signal is boosted by 6 dB. The distortion observed in the analog output is due to DAC clipping.

Figure 2. Excessive gain in the digital domain can make the signal exceed the DAC's upper and lower limits

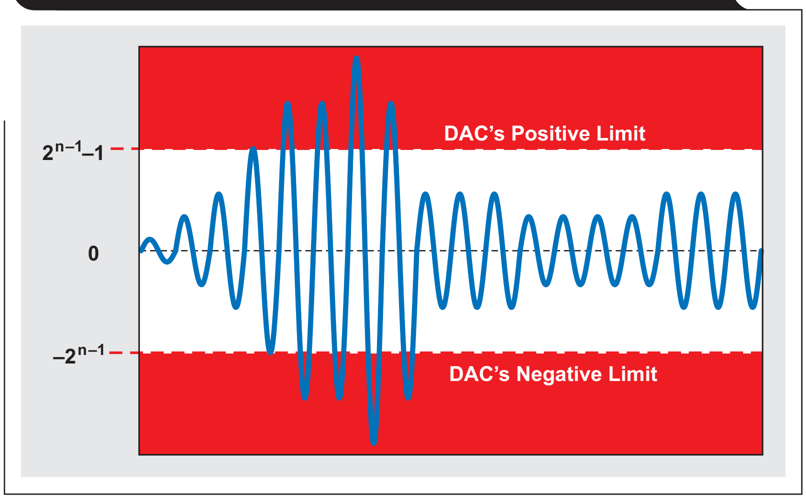
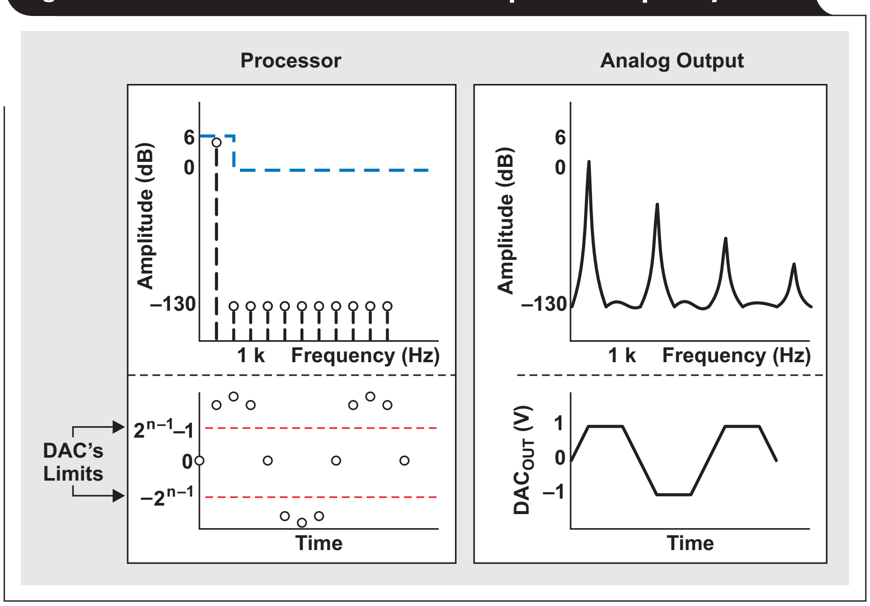
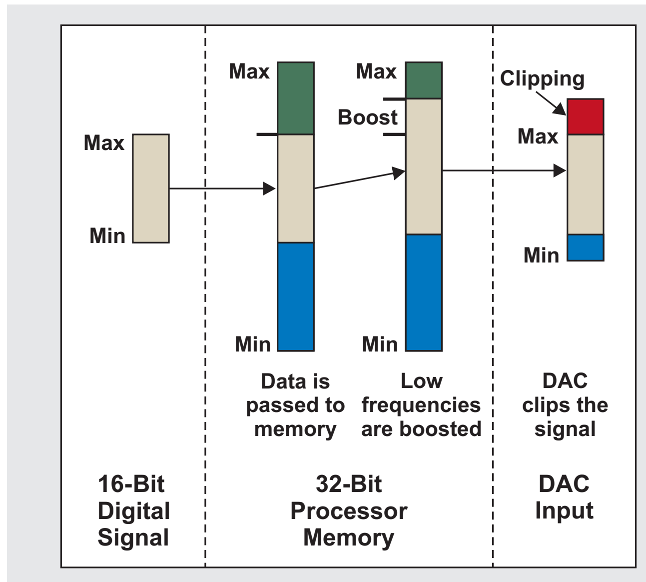


Figure 3. Possible effects of boost on a specific frequency band



This concept is also illustrated in Figure 4. Note that the noise from the source data is inherited when passed to the larger bus width of the processor's memory. As mentioned previously, data could be scaled down by the maximum amount of total boost to accommodate the boosted regions. However, as seen in Figure 5, even if the boost reference point is in a good position, the DAC signal might be affected by the output SNR. If the amount of boost does not compromise overall system SNR significantly, then simple scaling might be a viable solution. Some low-power codecs provide 100 dB of SNR, which allows some amount of scaling without sacrificing the SNR of the original 16-bit source.

Figure 4. Signal content throughout the digital signal chain



Quantization and number representation

In digital processing, a real number is represented as an integer value with a fixed precision. This is called quantization, and the quantized value is an approximation of the original value. The integer value can be represented as a fixed-point number or a floating-point number. An integer value represented as a fixed-point number is composed of magnitude bits and fractional bits. An integer value represented as a floating-point number is composed of exponent bits and mantissa bits. This discussion will henceforth be restricted to fixed-point numbers and fixed-point arithmetic.

A fixed-point number is represented as a twos-complement integer with a fixed number of digits after the radix point (or the decimal point). These digits make up the fractional part of the number. The digits before the radix point are the magnitude part and denote the range of the number. The magnitude part also contains the sign of the number.

Digital data coming into an audio processor is considered to be a real number lying between –1 and 1 – 1LSB. Assuming that the real value is represented as a 16-bit fixed-point number, the number –1 will be represented as 1000000000000000 in binary (or 0x8000 in hexadecimal). In twos-complement arithmetic, 0x8000 corresponds to an integer value equal to –32768. This means that dividing the integer number by 32768 will result in the quantized

Figure 5. Signal content with the use of scaling

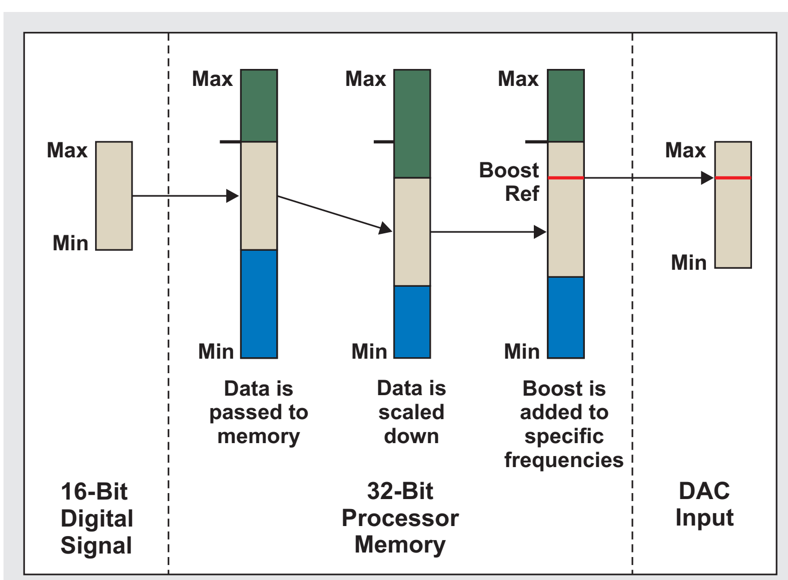
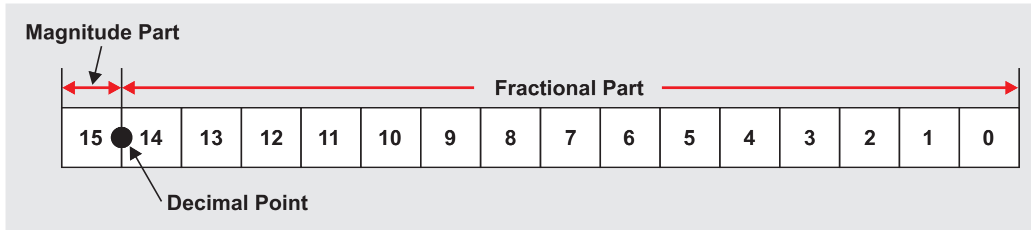


Figure 6. Fixed-point representation of a real number



approximation of the real value. The largest positive number in 16 bits is 0111111111111111 in binary (or 0x7FFF in hexadecimal). The corresponding integer value is 32767. Dividing this by the scale factor of 32768 produces the largest real number that can be represented in this format. The number is 32767/32768 = 0.999969482421875. The fixed-point representation is shown in Figure 6.

In this representation, there are 15 fractional bits and 1 magnitude bit, which is also the sign bit. This means that a real number must lie between –1 and 0.999969482421875 before quantization. If the real number is above or below this range, it cannot be represented in the given format because the 16-bit register will overflow. To accommodate larger real numbers, the magnitude part needs to be increased at the expense of a reduced fractional part. This format is also known as the 1.15 format (1 = magnitude bits, and 15 = sign bits). Input to a digital processor is always represented in 1.n format, where n is the number of fractional bits (15, 19, 23, or 31). A value of 0 dBFS corresponds to the RMS value of a full-scale sine wave whose amplitude is $(2^n – 1)/2^n$. The largest real number in the given format is represented by 2^n. The number of bits that are used to represent a signal is called the signal bit width or the data bit width.

Overflow and saturation

Overflow occurs when a processing unit's computation results in a value greater in magnitude than the data bit width. Overflow is typically associated with computation in the accumulator, where successive numbers of the same sign are added and stored. Accumulators usually keep accumulating even after overflow because the final result, if within bounds, will still come out correctly.

The output of the accumulator is saturated before it is stored as a signal value. Saturation is a process where a positive overflow is converted to the maximum positive number and a negative overflow is converted to the minimum negative number. Saturation is a nonlinear operation and results in severe harmonic distortion of the output. Headroom bits are used to prevent saturation.

Signal bits

Signal and noise bits affect the performance of a system. The digital audio processor adds quantization noise, and the overall performance will be the effect of both the analog circuit noise and the quantization noise. Assuming that both noise sources are the outcome of independent random processes, the overall system noise performance can be defined as

$$\mathrm{SNR} = 10\log_{10}\left(\frac{S^2}{N_C^2 + N_Q^2}\right),$$

where S is a uniformly distributed random signal, N_C is DAC circuit noise, and N_Q is quantization noise. Using a 100-dB DAC and a 120-dB signal processor will result in an overall SNR of 99.96 dB.

It should be noted that the overall SNR is also limited by the source—the input to the digital audio processor. If the input is provided as a 16-bit number, then the signal-to-quantization-noise ratio (SQNR) of the system can, at best, be 96 dB (assuming a uniformly distributed random signal, unweighted). So, even a higher-bit internal representation (lower N_Q) will not provide much improvement in this case.

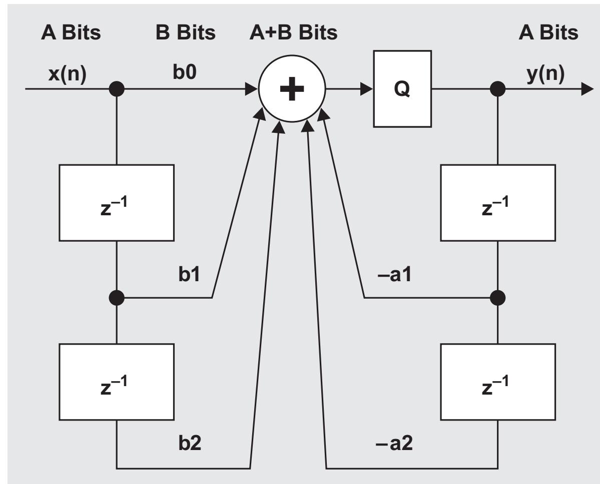
Noise bits

Earlier it was mentioned that the number of signal bits determines the performance of the digital audio system. More bits are sometimes needed for filter-response calculations.

A filter implementation consists of a data path through which the signal flows and is stored as delay elements for the filter. The signal and delay values are multiplied by the coefficients associated with the filter taps. Coefficient quantization also plays an important role in system performance. The product of the signal and coefficient values is stored in the accumulator, which usually has a higher bit width than that of the signal. Subsequent products are added in the accumulator (at a higher bit width), and the final filter output is then stored back in signal precision (at a lower bit width).

Consider the biquad filter implementation in Figure 7. In this figure, the input and output signals are represented by "A bits." The a and b coefficients are represented by "B bits." The input signal and its delay elements are multiplied by the coefficients and added at the accumulator. The multiplier and accumulator together are A + B bits wide. The output signal is then quantized by the Q block and stored as an A-bit number. This introduces a quantization error, which is a noise source for the digital filter; therefore extra bits are needed to ensure that the noise contribution from the digital filter is below the target SNR. These extra bits are called the noise bits. The effect of noise is more pronounced with IIR filters than with finite-impulse-response (FIR) filters. The number of noise bits also depends on the sampling frequency and the cutoff frequency of the digital filter. As the sampling frequency increases, the number of noise bits required increases. As the cutoff frequency decreases, the number of noise bits required increases. For a 48-kHz operation, 14 to 16 noise bits are enough to maintain the target SNR for a 40-Hz IIR filter.

Figure 7. Biquad filter implementation



Headroom bits

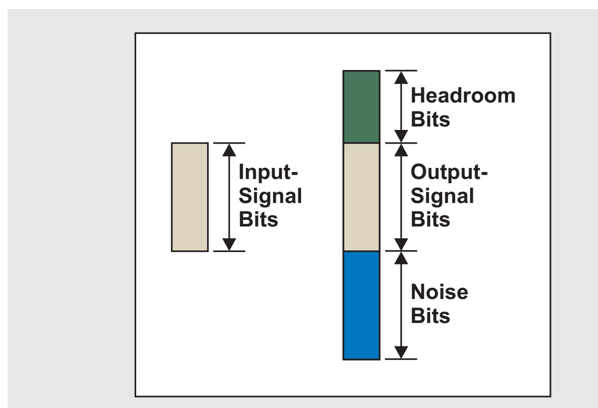
Other than signal and noise bits, additional bits are needed to prevent overflow. These bits are called headroom bits. An end-to-end audio-processing chain will usually preserve the signal level. This means that if a 0-dB signal is input to the signal chain, the output will measure 0 dB or less. (Usually there is a signal compressor that will limit the signal swing to a few decibels below zero.) If boost filters are used to amplify specific signal bands, the remaining bands are usually attenuated to prevent the signal from going above 0 dB. For the latter case, when the input-signal level is at 0 dB (also known as the neutral signal level), the output signal will be lower than 0 dB, and only the amplified bands will reach 0 dB at the output. This will reduce the average volume level of the audio signal.

In spite of the signal level being maintained at 0 dB, the signal can overflow at intermediate processing points. To prevent overflow, headroom bits—i.e., bits in addition to signal and noise bits—are needed.

There can be two sources of overflow:

1. An audio-processing chain can have a filter whose gain (at some specific frequency values) is greater than 0 dB. The filter can be part of a cascaded filter chain (e.g., low-pass, high-pass, and/or band-pass filters) whose overall gain is 0 dB, or it can be a frequency-selective filter that amplifies a specific frequency band relative to the neutral signal level (e.g., shelf and EQ filters). Note that if a real number is represented in 1.n format (where n is the number of fractional bits), the magnitude of the number is always less than 1. So, if a filter with a gain of more than 0 dB (a real number greater than 1) is used, then the output value from the filter is going to overflow if the input value is 0 dB (a real number equal to 1). To prevent overflow in such cases, more headroom bits are needed.

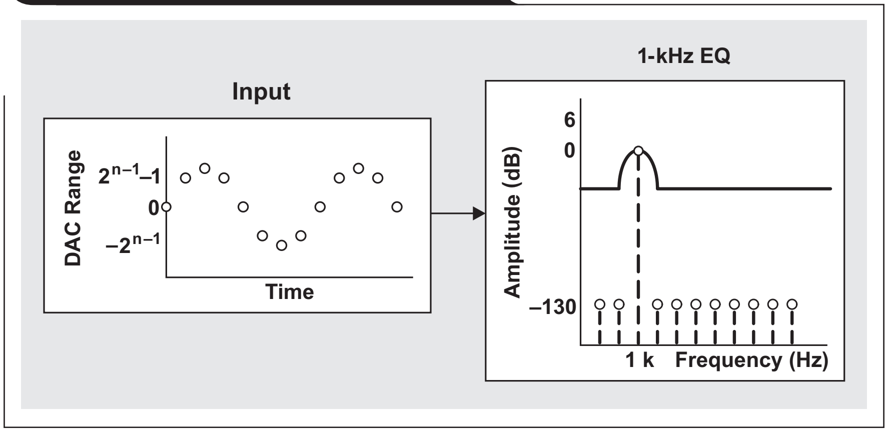
Figure 8. Representation of signal with headroom bits



2. A filter with a gain of less than or equal to 0 dB can have instantaneous real values greater than 1. To ensure that these instantaneous values do not overflow, headroom bits are needed.

A pictorial representation of the signal in the audio processor is shown in Figure 8. An important point to note is that headroom bits are primarily used to accommodate intermediate signal growth. It is expected that at the end of the final processing block, the output will fit within the signal bit width. Otherwise, for low-signal amplitudes, the output will still be within limits and not distort; but, for high-signal amplitudes, the output will get saturated and cause distortion. To prevent distortion, it is best to attenuate the signal prior to the final output.

Figure 9. Scaling the transfer function



Scaling

Scaling can be used to avoid saturation for filters with gain exceeding 0 dB. A boost filter can be used intentionally that will gain a particular frequency. Even a multisection low-pass filter can have a biquad section that actually gains certain frequencies that are higher than the available headroom (the overall response will still be 0 dB). In such a case, whether or not to use scaling can be determined by multiplying the input-signal level by the maximum gain of the total filter response. If the product is greater than the available DAC headroom, then scaling could be used to avoid saturation.

One method of scaling is to attenuate the system's transfer function by an amount equal to the maximum amplitude of the filter's transfer function. The scaling factor can be defined as

$$S = \max\left|H(e^{j\omega})\right|,$$

where $0 \le \omega \le \pi$. A second method is to scale the input signal by S. Figure 9 demonstrates the effect of scaling the transfer function. A full-scale sinusoid is input to the transfer function, which attenuates the flat frequencies by 6 dB. Relative to –6 dBFS, the 1-kHz signal is boosted by 6 dB.

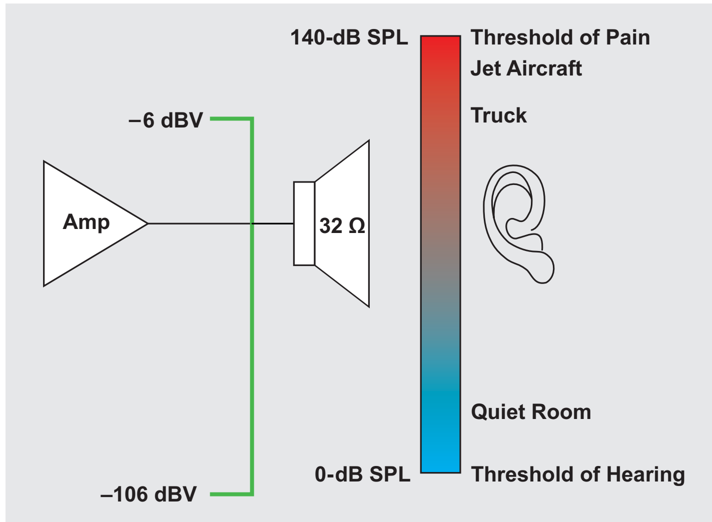
In some cases, due to the filter structure and the instantaneous signal sequences, the output of a filter can be more than 0 dB even though it does not have a gain of more than 0 dB. An FIR filter can increase the gain of a signal by a sum of the absolute value of the filter taps if the individual memory elements are at 0 dB with a sign opposite to that of the taps. The response of the filter may not exceed 0 dB, so additional headroom may be applied. Computing additional headroom for IIR filters is complex because they have feedback elements, and finding a closed-form expression to determine the upper limit for instantaneous gain is complex. In fact, one of the reasons signal processors provide additional headroom (above the DAC limit) is to allow headroom for instantaneous values. Measurements might be needed to compute additional headroom. In some cases, the SNR may need to be traded off for distortion that is due to saturation, and then an analog gain may need to be added to get the signal back to 0 dB.

When scaling is used, it is sometimes desirable to add additional gain (boost) in the analog output stage to compensate. Special care should be taken to ensure that the signal of the boosted regions does not saturate the output amplifiers as well, resulting in a distorted signal. Boost is also provided at the final output stage of the processor to compensate for scaling. This is required for multisection, 0-dB filters where scaling has been done to prevent overflow for one or more of the individual sections. For filters that gain frequencies above 0 dB (EQ and shelf filters), the neutral signal level is scaled below 0 dB. In this case, the final-stage boost is not required. The result is a loss of SNR for the flat regions.

A more elegant solution is to limit the amount of filter gain based upon the volume gain applied at the digital processor, which is very well-suited for headphone use. At higher volume levels, the frequency boost can be lowered and, ultimately, be flat at full volume.

In some cases, the frequency boost is kept constant, while the signal is compressed when the volume is high. This is the anti-clipping dynamic-range compressor (DRC) function: At low volume levels, the original SNR is maintained; but, as volume increases, the scaling is proportionately increased to prevent distortion.

Figure 10. Headphone volume relative to human hearing



Regardless of the method used, it is important to consider how humans perceive sound and noise. Human hearing has an outstanding dynamic range. Headphone amplifiers trade between noise floor and output power to best accommodate this range. For instance, the TLV320AIC3254 audio codec can deliver a very high sound-pressure level (SPL) with just 500 mV_{RMS} into a typical 32-Ω or 16-Ω headphone load, and at the same time can have a noise floor of 100 dB (A-weighted) below full scale, which can be below the threshold of hearing (see Figure 10). Sometimes, it is not even necessary to add additional amplification after scaling is performed, since the output power could be very well above the comfortable listening level.

References

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2. W. Marshall Leach, Jr., *Introduction to Electroacoustics and Audio Amplifier Design,* 3rd ed. (Dubuque, IA: Kendall/Hunt Publishing Co., 2003), p. 7.

Related Web sites

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