Digital Audio Conversion Distortions: What They Are and Why They Matter

Tom Martin of *The Absolute Sound* argues that **"A/D and D/A processing leads to mathematical errors that are quite unlike the basic music signal and thus are both obvious and distracting"** **theabsolutesound.com**. In other words, digital conversion can introduce distortions that don't occur in natural acoustic sound. Below we break down the key types of distortions in analog-to-digital (A/D) and digital-to-analog (D/A) conversion – including quantization noise, jitter, aliasing, and filter artifacts – along with newer or debated issues. We also examine why some listeners find these distortions so apparent, and what evidence or expert views support or refute that perception.

Key Distortions in A/D and D/A Conversion

Digital audio conversion errors stem from the discrete, mathematical nature of sampling and quantization. Unlike the mostly harmonic distortions of analog gear, digital errors often produce *non-harmonic* artifacts (signals not musically related to the source)theabsolutesound.combenchmarkmedia.com. These can grab our attention because the human brain is wired to notice unusual, "unnatural" soundstheabsolutesound.com. Below are the main types of distortions and how they arise:

1. Quantization Noise and Errors

What it is: When an analog waveform is converted to digital, its amplitude must be rounded to the nearest quantization level. This rounding introduces an error between the original signal and the quantized value. In essence, **quantization is a numeric rounding process** that adds distortion or noise to the signal benchmarkmedia.com. Without special techniques, these errors can manifest as a **"very non-musical and distorted version of the input audio"**benchmarkmedia.com – essentially a gritty, garbled sound on low-level details.

Audible effects: In early or poorly implemented digital systems (e.g. early CD recordings), quantization distortion could **"exceed the level of the music" during quiet passages and fade-outs** **benchmarkmedia.com**, making subtle details like reverb tails break up or disappear. This contributed to the infamous "harsh" or "grainy" character of early digital audio. Audiophiles noted that fades at the end of tracks sounded rough or had a sudden drop-out of detail, which was indeed quantization distortion (sometimes called "low-level resolution loss"). Properly dithered 16-bit audio has a noise floor around –96 dBFS; if undithered, the quantization error isn't random noise but correlated with the signal, producing audible low-level artifacts. **John Siau** (engineer at Benchmark) explains that in a well-designed system these quantization errors *can* be rendered as benign white noise "held to inaudible levels"benchmarkmedia.com – but in a bad design, they remain as distortion. Quantization errors tend to **"mute audio details, or amplify audio details,"** making them especially harmful to soft signalsbenchmarkmedia.com.

How it's managed: Fortunately, **engineers fully understand quantization error** and have developed solutions. **Dither** (adding a tiny bit of noise before quantization) randomizes the error, **turning distortion into low-level hiss**benchmarkmedia.com. Noise shaping can push that hiss energy to frequencies less audible to humans. With **sufficient bit depth**, the quantization noise can be made **"well below the threshold of hearing, even at very loud listening levels"**benchmarkmedia.com. For example, 24-bit audio offers ~144 dB dynamic range – far beyond what any listening scenario demandsbenchmarkmedia.com – making quantization noise essentially a non-issue. In short, **quantization distortion is real, but it is well understood and can be effectively eliminated** from audibility through proper dithering and high-resolution processing. Modern high-end audio production always uses 24-bit (or more) internal processing to avoid the accumulation of quantization errorbenchmarkmedia.com. This is why the glaring quantization artifacts of early 16-bit digital (which *"produced some of the worst sounding digital artifacts"* and gave digital a bad reputation benchmarkmedia.com) are largely a thing of the past.

2. Jitter (Clock Timing Errors)

What it is: Jitter is a slight variation in timing of the digital sampling clock. In A/D conversion, the analog signal must be sampled at perfectly regular intervals; in D/A, the samples must be converted back to analog at exact time instants. **If those timing intervals fluctuate (even by nanoseconds), it introduces phase modulation**

of the signal – effectively a tiny warble or frequency shift error superimposed on the audio benchmarkmedia.com. Importantly, jitter in the digital data **does nothing until the D/A stage** – it's when converting to analog that timing errors become analog distortiontheabsolutesound.com. The artifacts of jitter appear as **sideband frequencies** in the output: extra tones or noise close to the signal's frequencies. Depending on whether the jitter is random or periodic, these sidebands may or may not relate harmonically to the music.

Audible effects: Jitter artifacts are often described as **"enharmonic" distortion – tones not harmonically related to the music's overtones**theabsolutesound.com. This kind of distortion, being unnatural to musical instruments, can be audible even in minute amounts. Subjectively, excessive jitter is said to cause a **blurring or "smearing" of detail, a loss of soundstage focus, or a harsh/metallic character** to the sound. John Siau notes that **jitter can cause a "harsh, cluttered and unnatural sound" long before it reaches gross levels** benchmarkmedia.com. In analog terms, jitter is sometimes likened to **wow-and-flutter** (the pitch variation in wobbly tape decks or turntables), except jitter can occur at much higher frequencies than the slow wow of a tape – even multiple frequencies simultaneouslybenchmarkmedia.com. Audiophiles have attributed issues like "flat imaging" or weak bass to jitter, and indeed **some consumer digital devices have had jitter on the order of several nanoseconds**, causing distortion about **–78 dB relative to the music signal** benchmarkmedia.com. Such distortion, if not masked by the music itself, could be audible as a subtle veil or grit.

However, **how obvious jitter is in practice remains debated**. In the high-end audio press, many swear that reducing jitter yields a clearly tighter, more "analog" sound – hence the market for reclocking devices, precision master clocks, special cables, etc. Martin and colleagues emphasize that these time-based errors draw extra attention because the brain treats unexpected, non-musical artifacts as important theabsolutesound.com. For example, *ScottB* (an engineer/audiophile) wrote that timing errors create distortion that **"is enharmonic, and thus audible at much lower magnitudes than the harmonic distortion introduced by analog components"**theabsolutesound.com. This supports the idea that even tiny jitter might be perceptible under the right conditions, since the ear isn't used to those types of artifacts.

Engineering perspective: Technically, jitter is well-researched and **not a fundamental flaw but a solvable engineering issue**. As one design note flatly states: **"Jitter is not a fundamental limitation of digital systems, it is simply a defect."** With good design, **the timing can be made accurate enough that jitter-induced distortion is *below* audibility**benchmarkmedia.com. How much jitter is too much? Laboratory tests and listening trials have found that the threshold for audibility is quite low – jitter needs to be on the order of picoseconds (1e-12 s) or a few nanoseconds at worst, depending on its spectrum, to be inaudible. In fact, due to masking by the music itself, **even surprisingly large jitter may go unnoticed if it's uncorrelated**. Audio engineer **Ethan Winer** performed a controlled test adding progressively extreme jitter to music: he found that **"even 10 times more jitter than is usual for inexpensive audio devices is innocuous and doesn't harm the music"**audioxpress.com. At that outrageous jitter level (10 µs, which limits SNR to only ~35 dB), subjects still struggled to hear any specific degradation, apart from the music sounding slightly "thinner" at *worst*. At typical real-world jitter levels (nanoseconds or less), **objective tests indicate the impact on audio is negligible** – certainly not the dramatic changes to bass or soundstage that some audiophile anecdotes claim audioxpress.com. In short, modern DACs often have jitter in the picosecond range and **can essentially be considered "jitter-free" in terms of audible effects**benchmarkmedia.com. But not all gear is equal: cheap players or poorly implemented interfaces (like early S/PDIF receivers) did allow higher jitter to creep in, which **could** be audible. This disparity fuels the ongoing debate. On one hand, **subjectivist listeners insist jitter's subtle distortion is audible as a graying or glare** in the sound; on the other, **objectivist engineers find that once jitter is below certain thresholds, it's effectively impossible to identify in double-blind tests**. The consensus in engineering circles is that **jitter *can* be audibly transparent – but careful clock design or jitter-rejection is required**, especially in high-end playback where listeners are paying close attention.

3. Aliasing Artifacts

What it is: *Aliasing* occurs when a signal with frequency content above the system's Nyquist frequency (half the sampling rate) is sampled, causing those higher frequencies to "fold back" and appear as false lower-frequency signals. In A/D conversion, any content above ~22.05 kHz (for 44.1 kHz sampling) must be removed by an anti-alias filter, or else it will produce aliased components in the audible band. Similarly, if a D/A converter does not properly filter its output, the mirrored ultrasonic images of the audio can potentially mix and create spurious tones. Aliasing is a purely digital artifact – in analog, frequencies don't just turn into other frequencies. In digital, however, **"ultrasonic tones may be aliased such that they are reproduced at audible frequencies"**, resulting in tones that have **"no relationship to the music"**benchmarkmedia.com (i.e. not harmonically related). For example, a 30 kHz component in the original signal might fold down and reappear as a 14.1 kHz tone in a 44.1 kHz system – clearly something that was never intended.

Audible effects: Aliasing typically produces **obviously wrong artifacts** if it's severe – imagine high-pitched whistles or inharmonic "undertones" that shouldn't be there. Tom Martin gives an example: very high frequency sounds (say, a sharp cymbal crash with energy above 20 kHz) can generate alias tones. While good filtering removes most of it, **"certain non-linearities in the process can [let] remaining very high frequency alias signals"** interact and **"generate tones shifted into the midrange where they are easily heard"** theabsolutesound.com. In other words, two ultrasonic alias components could subtract and produce a false tone smack in the middle of the audible spectrum – a clear impurity to anyone listening. He notes these alias-derived tones are *unlike anything in the original music*, so they stick out. A real-world symptom of aliasing might be a sort of metallic sheen on complex high-frequency sounds, or weird quiet tones during intense treble moments. Thankfully, **blatant aliasing is rare in modern recordings** – it was more of a problem in the early days of digital or in poorly designed systems. (For instance, early CD machines with analog "brick-wall" filters couldn't perfectly attenuate everything above 22 kHz, so a bit of ultrasonics slipped through and folded down as low-level distortionbenchmarkmedia.com.)

Prevention: Engineers combat aliasing by using **steep low-pass filters** at the input of ADCs and the output of DACs. In modern practice, oversampling converters use digital filters that can **"brick-wall" at Nyquist far more effectively** than older analog filtersbenchmarkmedia.com. Thus, properly designed A/D and D/A converters should produce negligible aliasing artifacts. One controversial exception is **"NOS" (non-oversampling) DACs** popular among some audiophiles – these omit the oversampling digital filter. A NOS DAC running at 44.1 kHz technically allows images and alias components to pass through (e.g. audio at 10 kHz will repeat at 34.1 kHz, 54.1 kHz, etc., and those beyond the analog output filter could reflect back into audible range via intermodulation). Proponents of NOS DACs claim a more "natural" sound (often attributing it to lack of filter ringing), but the trade-off is *measurable aliasing distortion*. Most high-end DACs, however, do apply oversampling and filtering precisely to **avoid alias distortions which would "clutter the audible band"** benchmarkmedia.com. In summary, aliasing is a well-understood phenomenon that **engineers consider unacceptable** – any decent converter's design addresses it. If aliasing artifacts were present at audible levels, they would indeed be *distracting*, as Martin suggests, because they are like ghost notes that don't belong. The good news: with proper filtering, aliasing can be essentially eliminated from the listener's perspective.

4. Digital Filter Artifacts (Ringing and Pre-Ringing)

What it is: Digital audio relies on filters both in recording (to prevent aliasing) and playback (to reconstruct a smooth waveform). These filters – often Finite Impulse Response (FIR) filters – have an inherent side effect: **they "ring" in response to sudden transients**. Ringing means the filter causes a damped oscillation before and after a sharp impulse. With a standard *linear-phase* filter (the most common type used to maintain phase accuracy and flat frequency response), the ringing is symmetric: it starts *before* the transient (pre-ringing) and continues after (post-ringing) for a brief time. This pre-ringing is purely a digital artifact; in nature, you never hear a bell ring *before* it's struck, nor does any physical event produce a precursor sound. Thus, many in the audiophile community consider pre-ringing particularly **"a-musical"**. Martin calls **"pre-ringing" one of his favorite examples of a-musical digital distortion** – because it literally creates a **"distorted version of the music signal… *before* the 'real' signal happens"**theabsolutesound.com. No acoustic instrument does that, so it's argued to be an obvious anomaly.

Audible effects: The audible consequence of filter ringing is a matter of debate. Those critical of conventional digital filters claim that **pre-ringing smears transients**, reducing the crispness or attack of instruments. For example, a stick hit on a cymbal or snare drum might lose some "snap" because the filter's pre-echo blurs the onset. In extreme cases, one might almost hear a faint pre-echo. Martin notes that once listeners become aware of this, **the notion that a signal is happening before it should is clearly "completely unnatural"** theabsolutesound.com – it can make the sound seem uncanny if one is listening for it. He and others point out that **cymbal crashes and other sharp transients** on many digital systems can sound *"odd and unnatural"*, possibly due to these filter-induced artifacts on the leading edgetheabsolutesound.com theabsolutesound.com. Subjectively, some audiophiles describe a *hardness* or *glassy quality* to treble, or a sense that the music's "pace" or dynamics are slightly off – they blame the digital filter's time-domain behavior.

Engineering perspective: Digital filter ringing is an intrinsic result of trying to sharply cut off frequencies (like everything above 22 kHz in a CD) while preserving phase linearity. However, whether this ringing is audible is not so clear. Many engineers argue that **the ringing occurs at frequencies near the cutoff (e.g. around 20 kHz) and is of short duration, so human hearing hardly notices it**. Psychoacoustic studies have suggested that pre-echoes shorter than a few milliseconds at very high frequencies are difficult to perceive, especially if immediately followed by a loud transient (masking will occur). In a forum discussion, one audio developer noted that if a filter's cutoff is high enough, **"the ringing itself will not be audible for sufficiently high frequency filter cut-offs."**audiophilestyle.com In practice, modern oversampling filters are often designed to minimize energy of ringing in the audible band (for instance, some use gentle roll-offs or higher sample rates so that any ringing happens above 20 kHz). **Blind listening tests** have so far shown minimal ability for people to distinguish linear-phase filters (with pre-ringing) from minimum-phase filters (which eliminate pre-ringing by allowing phase distortion instead). For example, blogger **Archimago** conducted such tests and concluded that while filters differ in measurement, their audible differences were negligible for music – cautioning not to **"make a mountain out of a molehill"** regarding the *"much detested 'ringing'"*archimago.blogspot.com. In fact, Archimago's analysis of an actual music transient (a loud cannon shot) showed that the **linear-phase filter preserved the transient's waveform more accurately** than a minimum-phase filter, despite the latter's lack of pre-ringingarchimago.blogspot.comarchimago.blogspot.com. This suggests that the *theoretical* pre-ringing issue may be sonically benign when the whole music signal is considered.

That said, the high-end audio industry has taken the concern seriously. Several DAC manufacturers offer selectable filter settings (sharp vs. slow roll-off, linear vs. minimum phase) so listeners can choose what sounds best. **"Apodizing" filters** have been introduced to tackle not only the DAC's own ringing, but also any ringing artifacts already printed into recordings by the original ADC. (An apodizing filter slightly relaxes the cutoff and uses minimum-phase behavior to ensure no pre-ringing at all, at the cost of some high-frequency response; Meridian and others have championed this approach to make digital playback more natural.) The controversial MQA format also bases part of its philosophy on minimizing time-domain smearing via short, minimum-phase filters. While proponents claim audible improvements in clarity and "analog-like" timing, skeptics note these changes often reside at or below the threshold of typical hearing.

In summary, **filter artifacts like ringing are indeed "mathematical errors" of the conversion process**, as they have no direct analog in live sound. They *could* be "obvious and distracting"theabsolutesound.com if severe, but in well-designed systems their audibility is subtle. Listeners deeply invested in sonic perfection (e.g. audiophiles comparing high-end DACs) often swear they can hear a difference in transient reproduction with different filtering approaches. More casual listeners or engineers looking at controlled tests find that **these effects are minor at best**. The divergence in opinion here is a classic case of subjective impression vs. measured reality.

5. Other Controversial Digital Distortions

Beyond the big four above, a few other digital audio artifacts are sometimes discussed, though they tend to be more niche or largely solved:

- **Noise-Shaping Ultrasonic Noise:** Most modern A/D and D/A converters use delta-sigma modulation with noise shaping to achieve high resolution. This pushes quantization noise up into ultrasonic frequencies (e.g. 30–100 kHz). In theory this noise is inaudible, but if a lot of ultrasonic energy is present, it could potentially intermodulate in analog circuitry (amplifiers, tweeters) and produce subtle distortion in the audible range. Some audiophiles worry that formats like DSD, which contain strong ultrasonic noise, might impart a "hard" or fatiguing quality via such intermodulation. Engineers counter that properly designed equipment will filter or ignore ultrasonics (indeed many amps have limited response above 50 kHz), and experiments have not conclusively shown audible problems. Still, this remains a point of debate: whether the **"noise moved to inaudible frequencies"** really stays inaudible in all systemsbenchmarkmedia.com.

- **Inter-Sample Overs:** Digital audio is usually limited to 0 dBFS (full scale), but due to filtering, the reconstructed analog waveform can sometimes peak slightly above the digital max if multiple samples at 0 dBFS occur in a row. Poorly designed DACs or DSPs might clip these "inter-sample peaks," causing distortion on loud transients. This is not a fault of the digital format per se, but of implementation. Good mastering practice and DAC design avoid this, yet it's a subtle issue that came to light in the 2000s and has since been addressed in high-end devices (many DACs now have a little headroom or soft clipping for such cases).
- **Clocking and Interface-induced Noise:** Apart from jitter, digital interfaces (like USB or S/PDIF) can introduce noise coupling into audio. For example, some argue that USB power noise or ground noise can pollute the DAC's analog output stage, leading to a subtle grain. Others talk about **"packet noise" or buffer-induced timing variation** in networked audio. These issues stray from pure A/D D/A math errors into analog interference, but audiophiles do debate them. From an engineering standpoint, a well-isolated DAC should reject such noise, but in practice not all designs are equal, which might explain why some listeners report differences between, say, USB cables or streamers. The **ScottB** blog piece points out that even cables in synchronous digital transmission can alter the waveform shape and hence timing slightly, **"contribut[ing] to timing errors (jitter)"**, though the audibility of those differences depends on the whole system's jitter rejection and noise floortheabsolutesound.com theabsolutesound.com.
- **Dynamic Range Compression in the Digital Domain:** This is not a conversion error, but it's worth noting because some people confuse the "CD sound" with the loudness wars. If a digital master is heavily peak-limited or compressed, it can sound tiring or distorted compared to an LP of the same music that might be less compressed. This is an *artistic/mastering distortion* rather than a technical one from A/D or D/A, but it has contributed to the perception that "digital is fatiguing." When evaluating digital vs analog, one must ensure the comparison is fair – e.g., a high-quality, well-mastered digital track versus the same on vinyl. Often, complaints about "obvious digital glare" have as much to do with production choices as with conversion technology.

Why Are These Distortions Described as "Obvious and Distracting"?

From the above, one can see a common thread: **many digital distortions produce signals that are *unlike* natural musical overtones**. The ear/brain mechanism has evolved to find meaning in harmonic structures (that's how we identify instruments and voices) and to ignore benign noise (like the hiss of tape or air conditioning). But it also evolved to **"attend to certain kinds of signals — surprising or unnatural signals get extra attention"**theabsolutesound.com. Digital artifacts like quantization distortion (if undithered) or jitter sidebands or alias tones often fall into the "surprising/unnatural" category. They don't mask into the music as easily as gentle analog harmonic distortion does. Instead, they can **stick out like proverbial sore thumbs**, drawing the listener's focus away from the music and onto the distortion. This is what Martin means by *"distracting"* – your brain, even subconsciously, is saying *"hey, that crunchiness or that pre-echo isn't part of the music!"* and thus you're taken out of the musical experience.

For example, consider a well-recorded cymbal crash: on a top analog system or high-resolution digital, it might sound brassy and smooth, with a natural decay. On a poorer digital chain, maybe you hear a hashy aftersound or an odd swishing in the treble – something that makes the cymbal seem less real. A trained listener, especially one who frequently hears live unamplified music (the **"absolute sound" reference standard theabsolutesound.com**), will quickly notice those deviations. Martin specifically notes that **"many recordings heard today on digital have easily heard distortions if you are familiar with the sound of real**

instruments", citing cymbals as a prime exampletheabsolutesound.com. A **cymbal's transient attack** contains extremely high frequencies, potentially beyond 20 kHz, which can **challenge the Nyquist limit** – if the digital chain isn't perfect, aliasing or ringing might make that cymbal strike sound subtly wrong theabsolutesound.comtheabsolutesound.com. To such a listener, these problems are indeed *obvious*. They manifest as a kind of glare, or a "zing" that shouldn't be there, or an unnatural dullness if the transient is smeared. In short, the distortions are audible enough to pull one's attention.

Another angle is **distracting in comparison to what**? If one is used to an excellent analog setup or a state-of-the-art digital system, stepping down to a mediocre digital chain will reveal these flaws starkly. Martin and other high-end reviewers often use the term "a-musical digital distortions" to emphasize that once you've heard a converter that minimizes these errors, you realize how much typical digital was *adding* stuff that isn't music. In one review, Martin wrote that the Berkeley Alpha DAC Reference Series 3 was **"the first DAC I've heard that crosses the threshold where a-musical digital distortions are significantly reduced as an important distraction"gcaudio.com**. In other words, with that DAC, he could finally forget about the digital artifacts and just enjoy the music – implying that with other DACs, those artifacts had been an "important distraction." This perspective is common in the audiophile world: as digital technology improves, people report that the sound becomes more "natural, fluid, analog-like," suggesting the removal or reduction of whatever irritants were present before.

Are These Distortions Audible? – Expert Perspectives

The big question is whether these digital conversion distortions are *truly* audible and problematic to the average listener, or if they are overstated. The answer varies depending on whom you ask, and how the listening is done.

- **High-end audiophile viewpoint:** Many audiophile experts (reviewers, seasoned listeners) maintain that digital's flaws are indeed audible and can be addressed with careful equipment choices. We've already seen Tom Martin's stance that digital distortions are real and bothersome unless minimized. Others echo this: terms like *"harsh, brittle, lifeless, cold"* have often been used to describe early digital audiobenchmarkmedia.com. These descriptors point to the absence of natural warmth (because digital adds non-harmonic junk instead of euphonic harmonics) and presence of annoying treble artifacts. Audiophiles often recount how a better DAC or a better clock "opened up the soundstage" or made the music more *relaxing* to listen to. It's common to find assertions that *"digital done right"* is musical, but *"bad digital"* triggers listening fatigue – the ear's way of telling you it's annoyed by distortions and errors. As evidence, they point to their listening experiences, and occasionally to waveform analyses that show differences in time-domain behavior. A subjective but illustrative example: mastering engineer Bob Katz once noted that he could often distinguish 16-bit audio from 24-bit in a good monitoring environment by listening to the decay of reverb tails – 16-bit (if undithered or just at the edge of resolution) would truncate the very low-level detail, whereas 24-bit preserved a smoother decay. This kind of critical listening supports the idea that quantization noise and other issues aren't entirely benign if you know what to listen for.

Audiophile publications also share anecdotal "evidence" from demonstrations: e.g., playing a high-res track through a high-quality DAC versus a standard CD-quality through an older DAC and having listeners overwhelmingly prefer the former for its realism. While not always scientifically controlled, these testimonies reinforce each other in the high-end community. Additionally, companies like **Benchmark, dCS, Chord, and others** have designed specialty solutions (advanced upsampling filters, FPGA-based DAC algorithms, jitter-elimination circuits) precisely to address these errors, and reviewers often validate their effectiveness with listening tests. The existence of **"DAC shootouts"** where differences are heard implies that not all DACs handle these distortions equally – some indeed leave more "digital hardness" in the sound than others.

- **Engineering/scientific viewpoint:** On the other side, many engineers and researchers argue that if a digital system is competently designed — with sufficient bit depth, proper dither, low jitter, and good filtering — it can be **audibly transparent**, meaning any distortions are below thresholds of human perception. Objective studies lend support to this. For instance, the AES (Audio Engineering Society) published a famous study by Meyer and Moran (2007) where hundreds of trials showed that people could not reliably distinguish high-resolution audio from the same audio down-sampled to CD quality under double-blind conditions, provided the conversion was done properly. This suggests that the common digital limits (16-bit, 44.1 kHz) *do not* necessarily impose obvious distortions – at least not by 2007, when converters and processing had improved greatly from the 1980s. Likewise, controlled tests of jitter audibility (such as Ethan Winer's experiment or an older one by Julian Dunn) have found that below certain high levels, jitter's effect is vanishingly small. We saw Winer's result that even grotesque 10 μs jitter did not "destroy" the musical experienceaudioxpress.com. As he points out, jitter noise (when not extreme) is always *masked by the music itself* – unlike a tape hiss which you can hear in quiet pauses, jitter only produces sidebands when music is playing, and then they tend to be swamped by the music's louder componentsaudioxpress.com. This masking principle is key: many digital artifacts occur concurrently with the music (not as separate noise floor hiss or hum), so the brain often cannot separately detect them unless they are fairly large.

John Siau's detailed article "The Unique Evils of Digital Audio" ultimately concludes on an optimistic note: each of these "evils" (jitter, quantization, aliasing) can be **"defeated" such that digital audio achieves transparency**. He demonstrates, for example, that adding 16 bits of dither noise enables even a 1-bit system (like DSD) to perform at 20-bit fidelitybenchmarkmedia.com. He also notes that **"24-bit systems" and modern DSP mean even complex processing can be done without audible error**benchmarkmedia.com. Essentially, the engineering consensus is that **digital conversion *is* capable of delivering error-free (audibly) audio**, and if listeners still find it lacking, it might be due to implementation faults or even psychological bias. In forums like *Audiophile Style*, some technically-minded users argue that many supposed digital "distortions" are invoked as *boogeymen* to explain any disliked sound, when in fact they are below audibility. For example, when listeners claim one DAC "has better PRaT (pace, rhythm and timing)" or more depth, skeptics will ask: *where are the measurements of jitter or frequency response that correlate?* Often, modern DACs all measure flat and low-distortion; differences heard could be placebo or subtle analog output stage differences, rather than core conversion errors.

- **Contrasting views and bridging the gap:** There are also balanced perspectives. The *Guest Blog* by ScottB on *The Absolute Sound* (itself a subjectivist outlet) tries to reconcile that **both extremes have points**: yes, digital cables and such can in theory change jitter, but also yes, tiny differences may be heavily masked and require extraordinary circumstances to heartheabsolutesound.com theabsolutesound.com. Some mastering engineers acknowledge that while much of digital's issues have been solved, early experiences left a lingering distrust. Notably, analog veteran Bob Ludwig has said that today's best digital is **"finally good enough"** to not be the bottleneck – hinting that for years it wasn't. So part of Martin's stance comes from historical context: there really were problems in the past (like missing dither, bad filters, high jitter clocks in 80s gear) that were audible, and some audiophiles maintain those problems *still persist in many consumer devices*. Engineers respond that *properly engineered* devices no longer have those flaws in any audible amount. The truth likely depends on the exact equipment and listener. A resolving sound system in a quiet room, guided by a trained ear, can reveal subtle differences that a casual listener on average gear might never notice.

In conclusion, the **"obvious and distracting" nature of digital conversion distortions** is a subject of active discussion. **Objectively**, technologies like dithering, oversampling, and precision clocking mean that digital audio, when done right, *should* introduce only vanishingly small errors – far below the levels that typically annoy the ear. **Subjectively**, however, experienced listeners insist they can hear something "unnatural" in many digital playback chains, which they attribute to these very errors. Expert opinions span the spectrum: from **Tom Martin's** camp arguing that digital's math imperfections *clearly* compromise musical realism, to **Ethan Winer and many in pro audio** saying that modern digital, at normal listening levels, is essentially flawless and any perceived issues are due to other factors (room acoustics, transducers, or imagination).

The takeaway is that **quantization noise, jitter, aliasing, and filter artifacts are real phenomena** – they are not "myths" – but **their audibility depends on their severity and context**. In high-end audio, where the goal is to approach a lifelike experience, even a whiff of "a-musical" distortion can be bothersome. This is why such distortions get a lot of attention and are described in strong terms. Meanwhile, scientific evaluations remind us that when these errors are pushed beneath psychoacoustic thresholds, music *can* sound transparently clean. As digital audio continues to evolve, the gap between the two perspectives may narrow. Indeed, we now see many audiophiles acknowledging that today's best digital sources are far more *believable* than those of decades past, indicating progress in taming these once "distracting" distortions.

Comparison of Digital Conversion Distortions

For a clearer summary, the table below compares the various distortions discussed, highlighting their causes, how they affect sound, and what experts say about their audibility:

Distortion	Cause & Nature	Audiophile View (Concern)	Engineering View (Solutions & Audibility)
Quantization Noise (quantization distortion)	Rounding of analog signal to discrete digital steps introduces error. Without dither, error correlates with signal (nonlinear distortion); with dither, it becomes random noise.	Can **"add very high distortion to low-level signals,"** obscuring fine details benchmarkmedia.com. Early 16-bit digital often sounded gritty on fades or reverb tails due to quantization errors. Considered **unnatural** because it can truncate or garble quiet musical nuances.	**Well-understood & fixable:** Dither turns distortion into benign hiss benchmarkmedia.com. 24-bit depth and noise shaping push quantization noise far below audibility. Modern systems achieve > 120 dB SNR, so quantization noise is effectively inaudible. (If audible distortion occurs, it's a design flaw or lack of dither.)
Jitter (clock timing error)	Variations in timing of sample intervals cause small timing errors in playback, leading to sideband frequency modulation of the signal (often non-harmonic).	Can cause **"harsh, cluttered and unnatural"** sound if significant benchmarkmedia.com. Blamed for reduced imaging precision and "cold" treble. Audiophiles consider even tiny jitter potentially audible as a loss of clarity or emotional engagement. Enharmonic jitter artifacts draw attention disproportionately theabsolutesound.com.	**Can be minimized to inaudibility:** Good clock design yields jitter in picoseconds, producing distortion ~ –120 dB or lower. Random jitter is masked by music; moderate levels (<1 ns) rarely affect sound audibly. Engineering tests show typical jitter in **competent DACs is inaudible**audioxpress.com. Audible only if poorly implemented (e.g. some old CD players or cheap DACs with >2 ns jitter causing –78 dB sidebands benchmarkmedia.com). Thresholds are debated, but consensus is that **jitter can be made negligible** with today's technology benchmarkmedia.com.
Aliasing (folded frequencies)	Occurs when input contains frequencies above Nyquist or when output is not properly filtered. High-frequency content "folds" into false	**Highly offensive if present:** Creates spurious tones with **"no relationship to the music"** benchmarkmedia.com, which	**Prevented by design:** Anti-alias and reconstruction filters remove frequencies above Nyquist. Oversampling and digital filtering have virtually

Distortion	Cause & Nature	Audiophile View (Concern)	Engineering View (Solutions & Audibility)
	lower frequencies. Produces tones that were not in original signal (inharmonic).	are heard as strange whistles or warbles. For example, improper handling of a hot cymbal's overtones could generate unnatural low-mid artifacts theabsolutesound.com. Even low-level aliasing is viewed as corrupting the purity of the music.	eliminated aliasing in modern AD/DA. Audible aliasing was mostly an early-digital artifact benchmarkmedia.com. In normal use, aliasing distortion indicates a faulty process (or a deliberate NOS DAC choice). Properly designed systems have **no audible aliasing** (any residual ultrasonics are filtered or too low to matter).
Filter Ringing (pre- and post-ringing)	Linear-phase digital filters (used for anti-aliasing/reconstruction) introduce ringing around transients. **Pre-ringing** = a small oscillation *before* the main impulse (unnatural in time domain). Post-ringing follows the impulse.	**Seen as a major "digital signature":** Pre-ringing is considered *entirely unnatural* **– sound occurring before cause**theabsolutesound.com. Believed to smear attacks and diminish "snap" or realism of percussion and other fast transients. Audiophiles find that eliminating pre-ringing (via minimum-phase or other filters) yields more organic, analog-like sound. Thus, conventional brick-wall filters are often viewed with suspicion for causing listening fatigue or a "sterile" quality.	**Trade-off managed through filter design:** Linear-phase filters ensure accurate frequency response but do ring. Many modern DACs offer alternative filters (slow roll-off, minimum-phase, etc.) to address concerns. Research indicates that typical filter ringing at 44.1 kHz is very short (<1 ms) and at high frequencies, thus **often inaudible**. Blind tests show minimal audible difference between filter types in music. Some experts argue the **audibility of pre-ringing is overstated** archimago.blogspot.com, especially with high sample rates or gentle filtering. In practice, designers choose filter shapes balancing ringing vs. frequency response, and any remaining ringing artifacts are subtle.
Other digital artifacts (noise shaping, inter-sample peaks, etc.)	**Noise shaping** in delta-sigma converters shifts quantization noise to ultrasonic range (potential IMD issues in analog domain if not handled). **Inter-sample overs** can clip if DAC isn't designed with headroom. Interface-related issues (e.g. ground noise, EMI) can introduce spuriae.	These are on the fringes of audibility, but audiophiles with highly resolving systems and ears claim they can contribute to a "digital glare" or fatigue. For example, some prefer multibit DACs over 1-bit delta-sigma, suspecting the ultrasonics from noise shaping make things edgy. Likewise, a poorly handling of inter-sample overs might make a	Engineers mitigate these: DACs often include an analog filter to tame ultrasonics; proper dithering and modest levels ensure no audible IMD from noise-shaped spectra. Many DACs have ~3 dB headroom for inter-sample peaks now. These issues are **acknowledged but largely solved** in good equipment. Any remaining effects are typically down in the

Distortion	Cause & Nature	Audiophile View (Concern)	Engineering View (Solutions & Audibility)
		loud passage sound unexpectedly compressed or crackly.	noise floor. Objective tests haven't shown convincing audible differences attributable solely to these factors under controlled conditions.

Table: Common digital conversion distortions, their origin, perceived sonic impact, and expert consensus on audibility.

Final Thoughts

In summary, analog-to-digital and digital-to-analog conversion can introduce various types of distortions, each rooted in the mathematics of sampling and quantization. **Quantization error, jitter, aliasing, and filtering artifacts** are the main culprits often cited. These distortions **differ from analog distortions** in that they frequently create *new*, non-musical frequency components (anharmonic tones, noise bursts, pre-echoes) that are not part of the original signal. This is why writers like Tom Martin emphasize their **"a-musical"** character theabsolutesound.com. To a discerning ear, these anomalies can indeed be **"obvious and distracting"** theabsolutesound.com – once you learn to detect the subtle pre-ringing before a snare hit, or the slight roughness on a quiet piano fade-out, it's hard to ignore it.

However, the extent to which these issues **degrade everyday listening** is a matter of scale. **Objective engineering advancements** have made digital conversion remarkably transparent: the use of dither, better clocks, higher sampling rates, and oversampling filters means that in most well-designed digital audio systems, noise and distortion from conversion are below the audible threshold. **Subjectively**, though, experienced listeners may still hear differences that they attribute to these conversion artifacts, especially when comparing ultra-high-end solutions aimed at mitigating them. There is some truth on both sides. As one audio blogger neatly put it after extensive testing: digital conversion artifacts exist, but one should avoid hyperbole – it's possible we sometimes **"focus on the wrong things"** and miss that good digital audio, even with its imperfections, is incredibly faithfularchimago.blogspot.com.

In the high-end audiophile domain, the conversation continues to evolve. New formats and techniques (like MQA's time-domain optimization, or novel DAC architectures) are often marketed as conquering the last vestiges of "digital sound." Skeptics often demand proof, while enthusiasts report positive listening experiences. Ultimately, whether one finds digital distortions *obvious* may depend on listening habits and context. To the average music lover enjoying a 320 kbps stream, claims of pre-ringing or jitter probably seem esoteric or irrelevant – the music sounds just fine. But to the dedicated audiophile who has spent hours comparing a D/A filter at 192 kHz versus 44.1 kHz, those tiny differences form part of a never-ending quest for the *absolute sound*.

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