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A FRICTIONLESS FUTURE FOR RETAIL?

How technology could drive changes to retail in cinemas.

PROUDLY CARBON NEUTRAL

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IN SEARCH OF THE ALL-DIGITAL B-CHAIN

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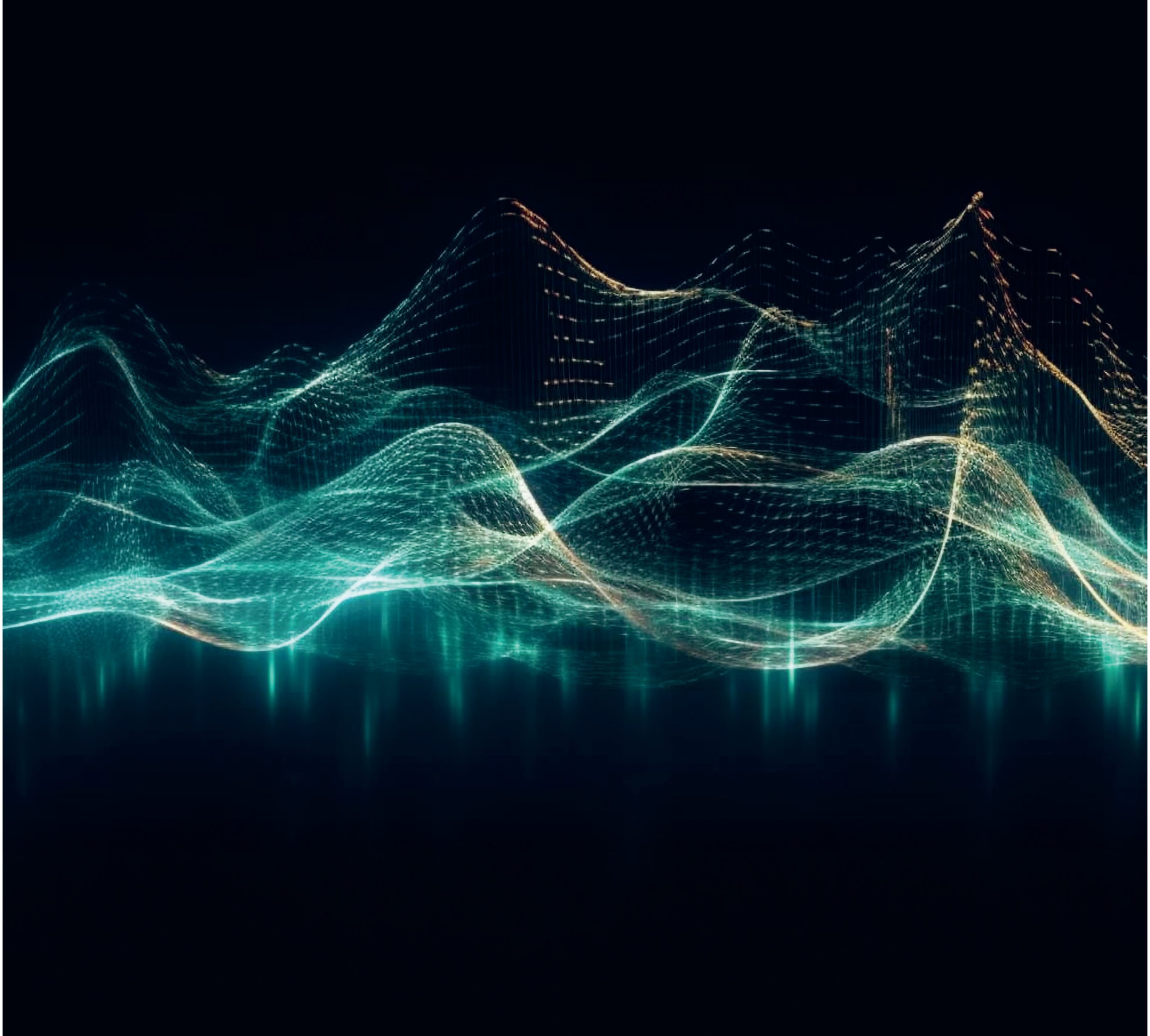


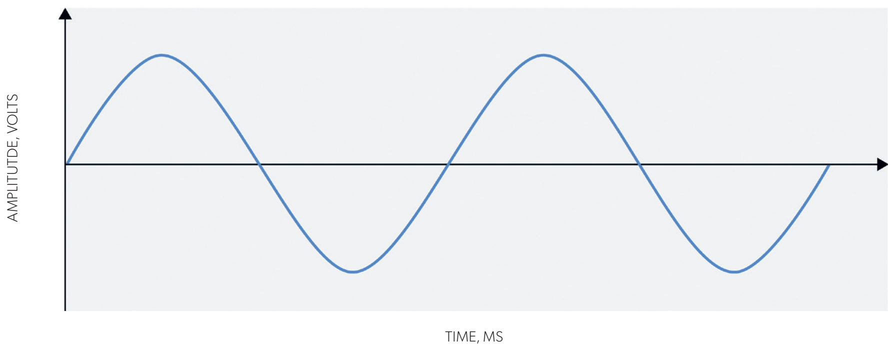
It's been over 25 years since cinema technology entered the digital age, and nearly a decade since most of the world's screens converted to digital projection. Digital technologies have fundamentally changed the way movies are made and played. >

For live action films, it's quite common that an image is only converted from analog to digital once (by a digital camera) before it is converted back to analog as light in the theatre.

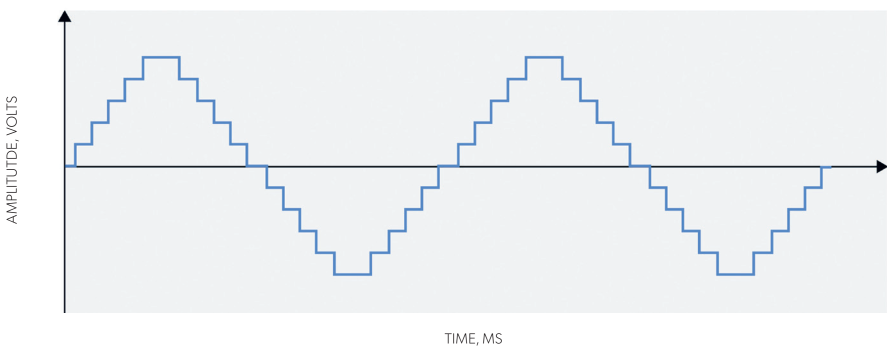
For animated films, the image never needs to be analog until it is projected as light. But for sound, it's a different story. End-to-end digital signal transmission for cinema sound is only just now becoming a reality. In traditional B-chain systems, there are sometimes up to three stages of analog-to-digital (A-to-D, or A2D) and digital-to-analog (D-to-A, or D2A) conversions. Each stage is a potential source of data error, signal loss, and noise - and thus distortion - of the original signal.

An analog audio signal is continuous over time, like a smooth sine wave. Converting a continuous analog signal to digital requires it to be sampled very frequently over time so that it can be converted into a numerical representation of that smooth signal. This is the process of digitization.

1 A CONTINUOUS ANALOG SINE WAVE



2 A 'DIGITIZED' SINE WAVE



Factors Affecting Conversion Quality

There are three main parameters to consider when assessing the effect of A-to-D and D-to-A conversion: sampling rate, quantization, and compression.

Sampling rate is the number of times per second that the analog signal is measured. In the digitized sine wave shown in figure 2, each horizontal "tread" on the steps represents one sample. In the early days of digital audio, a sampling rate of 44,100 times per second was quite common and considered adequate; in fact, this is often called "CD (compact disc) quality". More recently, engineers have found that much higher rates (96 kHz or 192 kHz) can result in even less distortion and better sound quality. The higher the sampling rate, the better the resolution of the audio signal. Theoretically, an infinite number of samples would essentially be a continuous signal, like the original analog signal produced by a natural sound source (human voice, musical instrument, a roaring lion, etc.).

Quantization is the process of taking all those samples and reducing the number of them to more manageable quantity. This is accomplished by process similar to rounding or truncating the values of the sampled quantities. Quantization can introduce distortion know as "quantization error", similar to the way that rounding of numbers can introduce a degree of error in a mathematical calculation. Different methods of managing error determine the effect on ultimate sound quality.

This quantization of the samples results in compression, which is the encoding of information using fewer digital "bits" than the original sampling. Compression is often necessary because of the vast quanity of data required to represent a complex signal such as audio. But compression can also introduce trade-offs of its own: in order to reduce file size, data regarded as non-essential for human auditory perception is discarded (also called "lossy compression"). Lossless compression, on the other hand, allows the original signal to be perfectly reconstructed; however the trade-offs include much larger file sizes requiring increased storage capacity, required bandwidth for file transfer, and loading time between devices (known as "latency").

An analog-to-digital converter is usually implemented on a custom integrated circuit (IC). Depending on the manufacturing process of the chip, other nonlinearity errors can occur which ultimately affect the signal-to-noise ratio of the device.

For all of these reasons, in order to optimize ultimate sound quality, system designers prefer to minimize the number of A-to-D and D-to-A conversions in a signal path.

The Traditional B-Chain Approach

The DCP (digital cinema package) contains both digital image and digital sound files. Image files go directly from the IMB embedded in the projector chassis to the projection engine, while the digital sound files leave the confines of the projector on its way to the final analog destinations: the loudspeaker, the room, and ultimately, the ear.

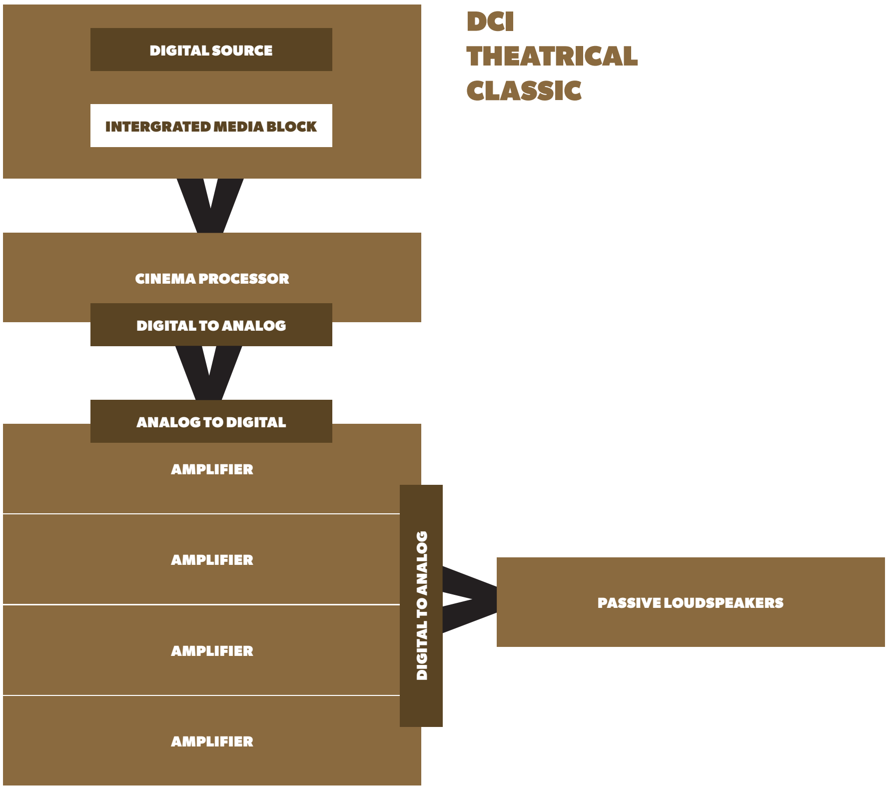
The first stop is the cinema processor. Most, if not all, cinema processors now accept AES3 (or AES/EBU) digital inputs on RJ45 connectors and cabling from the IMB. Coming out of the cinema processor the signal is subject to its first digital-to-analog conversion on its way to the power amplifier (assuming the power amplifier does not allow digital inputs).

While there are many types of power amplifier designs, an increasingly popular topology is called Class D. These designs operate in the digital domain, so any analog signal coming in must be subject to a stage of A-to-D conversion.

As the amplified signal leaves the power amplifier, it is once again converted from digital to analog so that it can be sent down the (sometimes very long runs of) copper loudspeaker cabling, on its way to the loudspeaker.

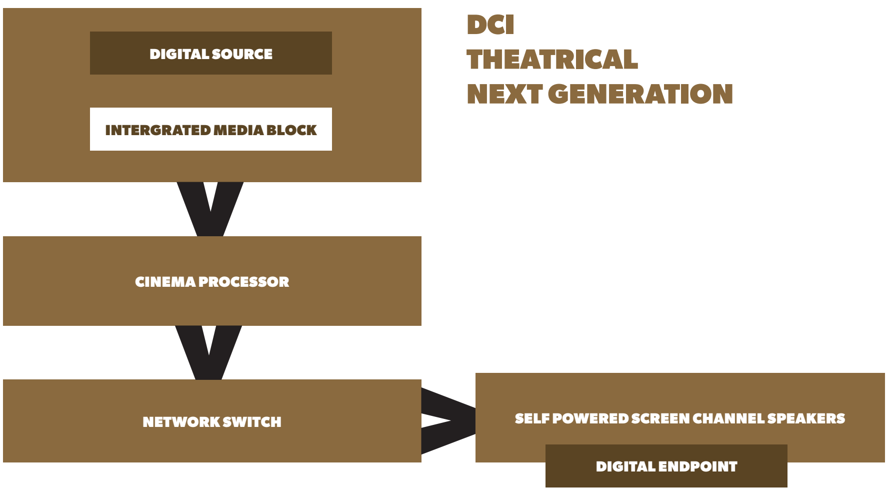
Each of these stages of conversion risks signal loss, increased noise due to improper gain staging, and other unwanted digital artifacts. All of these conditions are types of distortion that affect the sound quality.

3 TRADITIONAL DIGITAL CINEMA B-CHAIN

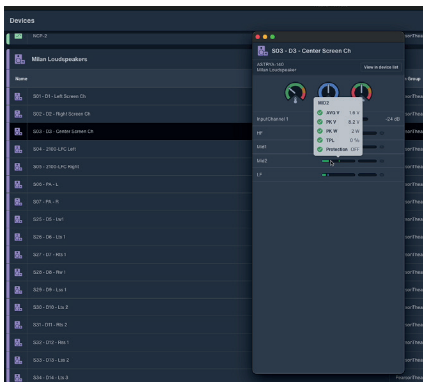


4 ALL-DIGITAL SIGNAL PATH

A better system design would be one that minimizes - or eliminates - A-to-D and D-to-A conversions, thereby preserving to the greatest extent possible the integrity of the audio signal throughout the chain, from the IMB to the loudspeaker. Borrowing from the IT world, this networked approach uses standard audio-over-IP (AoIP) layer 3 protocols to maintain the digital signal path. In this case, the loudspeaker itself is essentially a network endpoint, like your laptop on an office network. The power amplifier has a digital input and is integrated with the loudspeaker, which eliminates long loudspeaker cable runs and enables power matching to the loudspeaker and optimized digital signal processing. The result is a truly end-to-end digital signal path which removes the need for D2A converters, simplifying the complete B-Chain and reducing variables and room for error.



5 TELEMETRY IS TRANSMITTED VIA NETWORK TO MONITOR MANY PARAMETERS OF OPERATIONAL STATUS FOR EACH LOUDSPEAKER COMPONENT



Since it's a network, data can be bi-directional, allowing telemetry to be sent back upstream through the network to a PC (figure 5).

This allows you to monitor operational status, such as the condition of each individual driver, electronics, and overall system health. **CT**

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