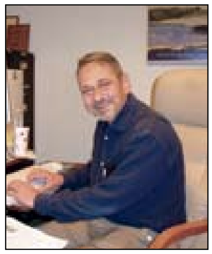
Editors' Notes

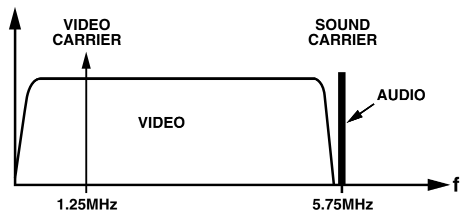
A READER COMMENTS



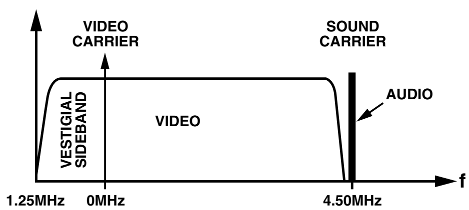
Last fall, we published an article about adding stereo audio to a satellite set-top box. The key idea was to apply a phase-locked loop, with the vestigial 15.734-kHz pilot signal in the composite spectrum as the reference, allowing one channel of the AD71028 BTSC encoder to derive the primary channel's master clock while the other channel provides the MTS stereo-encoded output for satellite set-top boxes and receivers.

We received the following email from Matt Laun, at NASA:

> "I found your design using the AD71028 as a STB (satellite set-top box) stereo synthesizer clever and interesting. However, I can suggest two corrections to the information in the article. One is the RF spectrum of a modulated NTSC video and audio signal. It appears in your article as the following:



> The frequency information may be misleading to a reader not familiar to the standard. More commonly, the audio carrier frequency is measured relative to the channel carrier frequency or the video carrier frequency. The audio subcarrier would be 4.5 MHz. It is more intuitive to view the spectrum as seen below:



> The second correction is minor but may be significant if a reader is trying to understand the design by performing calculations. When describing the need for a precision N-divider, the value N is listed as 780.9838... The precision value is actually 780.9706666... { = 48 kHz × 256/15.734265734 kHz}; [fractionally: = 49201152/63000]. I understand the clever design eliminates the need for a separate precision divider, I just thought a more accurate number would help a reader understand how that number is derived.
>
> Thanks for a relevant, practical, and interesting article."

Lead author, Jeritt Kent, responded:

> "First of all, thank you very much for this kind letter. I am very pleased that you enjoyed the article. Victor and I worked hard to ensure that the paper was clear and easy to understand.
>
> Your two clarifications are very valid. The first one was likely an interpretation by the graphic designer, as the original intended figure would have matched Fig. 2 in this document http://www.sencore.com/custsup/pdf/TT213.pdf
>
> The second observation shows an example where rounded mathematics can generate problematic results, albeit this is not a high volume Intel Pentium processor design nor the Hubble telescope:) Nonetheless, a Google search on <15734.265734 Hz> provides some excellent history on the subject from the University of Victoria, Canada (UV)." (www.ece.uvic.ca/~yli/misc/discussion.htm)

That UV document, titled "44100," was a fascinating bit of history. An abbreviated version follows:

> The original appearance of 44.1 kHz was with the Sony F1 digital audio recorder. It recorded digitized stereo audio as an emulated video signal onto Betamax tape. **You might recall that the horizontal scan rate for NTSC video is 15750 Hz (actually 99.9% of that but we'll get to that later). This comes from 30 frames/sec × 525 lines/frame. Note that 44.1 kHz is exactly 2.8 times 15.75 kHz.** The vertical blanking normally done on video was not used, three stereo samples were recorded on each horizontal line, and every fifth line had a checksum or some kind of redundancy sample for error detection and correction. With 14 samples for every five lines, you get a ratio of 2.8. There was considerable resistance by the AES and the audio community to have 44.1 kHz adopted as any kind of standard. We wanted 48.000 kHz because it was nicely synced to movies and both U.S. and European TV and, being a little higher, gave us a little more transition band for the antialiasing and anti-imaging filters (and we really needed that transition band back before the day of sigma-delta converters), but Sony and Philips had pretty big guns and dictated the standard since they owned the technology.
>
> Why would they not let it be 48 kHz? Rumor has it that the president of Sony wanted *all* of Beethoven's 9th to fit uninterrupted on a single CD. The 12-cm diameter had already been carved into stone, and there was no way for it to fit with a 48-kHz sampling rate. I don't know how long Beethoven's 9th is, but a CD can hold 74 min, 33 sec of music.
>
> **The frame rate (and horizontal scan rate) was actually reduced by 0.1% due to the emergence of color TV. Originally, it was exactly 15750 Hz and the sound subcarrier was at exactly 4.5 MHz.** Color TVs encode R, G, & B into a B&W luminance signal called Y, and I&Q (in-phase and quadrature) chroma signals. Y is transmitted just like a normal B&W signal. I&Q are transmitted using quadrature carrier modulation, and are bumped up to exactly halfway between the 227th and 228th harmonic of the horizontal scan rate. There is not much B&W energy up there at the 227th harmonic, so they could stick the chroma up there without degrading the B&W image too much. In addition, putting the chroma exactly at 227.5 times the horizontal rate (and all of the harmonics of the chroma would be halfway between harmonics of the B&W signal) would cause any interference from the chroma signal to be exactly negated on the neighboring scan line. The same is true if you consider the effect that the B&W signal has on the chroma signal: the B&W signal is at the –227.5th harmonic of the chroma baseband, so its interference is negated with every other scan line. The sound subcarrier is a lot closer to the chroma signal than to the B&W signal, so the color TV guys had to worry about it.
>
> **The sound carrier is at exactly 285.71428571 times the horizontal. The color TV guys wanted it to be an exact integer multiple (so it would be 1/2 harmonic off from the chroma signal). So, they decided to make the sound carrier exactly 286 times the horizontal. Unfortunately, instead of bumping up the sound carrier to 4.5045 MHz (these guys in the '50s thought that this would be too far off for existing FM receivers to lock to), they bumped the horizontal scan rate down to 15734.265734 Hz. This caused the horrible 29.97 Hz drop-frame mess that today's technicians have to deal with and the 44055.944056 Hz sampling rate.**

By the way, 15734.265734 is a truncation of 15750/1.001, or 15734.2657342657..., while 15750 × 0.999 = 15734.25. Now, here is the last word from Matt Laun:

> "Thank you too for the fascinating Google '.doc' about the origins of 15734.265734! I am a recording engineer as well as an electronic hobbyist with a nostalgic love for analog NTSC and BTSC modulations. I never knew that audio 44100 kHz was derived from video (and not even standard NTSC sync at that)! Incredible!"

If only Beethoven had known! A few bars less, and he might have simplified life for designers in the Age of Media.

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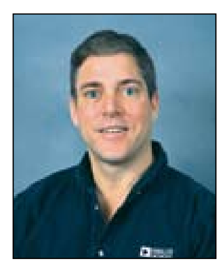
SIMULATED REALITY



In addition to three outstanding feature articles, this issue includes the second installment of Barrie Gilbert's fantasy. In it, Barrie introduces an example showing the power of circuit simulation. In a short time, Niku was able to gain valuable insights regarding the behavior of an ideal oscillator and to debunk the commonly held belief that it would start up given a spike on a supply or bias line.

In this column, Dr. Leif has provided some clues regarding the fourth "Dee" of Analog. We invite our readers to guess what Barrie has in mind, and to provide anecdotes from your experiences with simulators or with analog design in general.

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