Subject: Need help understanding my Museatex BIDAT manual (Ed Meitner)
Posted by akhilesh on Mon, 27 Jun 2005 18:29:18 GMT
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Hi Everyone,I was wondering if someone more into DA processing than me could help me decipher the following, taken from my (fully updated) bidat):----EXCERPT FROM BIDAT MANUAL-------"In January, 1993, the Meitner IDAT (Intelligent Digital Audio Translator) was introduced to the audio market with a price tag hovering around 15,000 dollars. It met with rave reviews around the world. At the same time as IDATs were being produced, Museatex was launching a project that would incorporate the IDAT algorithm and design features into a less expensive product: the Baby Idat, or Bidat. The Bidat, at a fraction of the cost of its predecessor, now outperforms it. At the heart of the Museatex Bidat is a unique and proprietary digital filter algorithm which processes digital audio information using two DSP microprocessors. As the name implies, this algorithm examines the digital audio data and makes an intelligent determination on how best to up sample this data to an eight times over sampled rate (8Fs, 352.8KHz for CD) while maintaining unequaled faithfulness to the sounds originally recorded. Other manufacturers who have designed their own digital filters have mostly used algorithms similar to the textbook topologies found in the commonly available filter IC's. A few have made attempts to optimize their filters by concentrating on transient response accuracy and have done so by sacrificing their frequency response. There are other designs which claim optimization of both time and frequency responses but nonetheless exhibit audio artifacts similar to those produced by the standard mass-market filter IC's; principally a symmetrical ringing at 1/2Fs which occurs before and after each transient and is known as Gibb's Phenomenon The Bidat algorithm analyzes the digital audio samples for relative accelerations and then chooses the type of filter most appropriate for that segment of data. This intelligent process results in a frequency response which is flat to 20KHz for CD playback. More important, it produces a uniquely pure impulse response without Gibb's ringing. For example, a square wave will have clean rectangular transitions and no ringing. The Museatex Bidat can claim transient response demonstrably MORE faithful to the recorded signal than the best analog playback systems for vinyl LP's and open-reel tapes. The undistorted transient response of the Bidat means the algorithm contributes no coloration of its own to the processed audio. This performance eliminates the "digital washout" of sonic subtleties and the harshness heard in listening to a conventional interpolating algorithm. For the first time in audio engineering, the full advantage of Digital Signal Processing has been used in the service of signal fidelity by the implementation of a smart filter which could never have been built with analog circuits.--------END Excerpt------Would appreciate knowledgeable help from people. Usually, I would disregard tech mumbo-jumbo in a hi-fi sales manual, but this is from Ed Meitner, who is anything but that. -akhilesh

Subject: Fiter response - frequency domain and time domain
Posted by Wayne Parham on Mon, 27 Jun 2005 19:53:06 GMT
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Study up on filters, paying attention to amplitude response verses time response. You'll find information about the things mentioned in the excerpt you quoted. Basically, filters that are

optimized for the time domain produce response anomalies and those optimized for the frequency domain produce phase anomalies.A digital filter is usually a software implementation of an analog filter.  One could be written that didn't have an analog, but it would probably do something unnatural and be of little value.  An example would be a digital filter that removed 10 samples every 10th of a second.  Not very useful.  What is usually implemented are digital implementations of standard analog filters.Digital processing also includes algorithms that massage the data in an attempt to get a better representation of the original analog signal.  Upsampling is an algorithm that inserts interpolated data points in between two sampled ones.  This kind of digital processing is a lot like using smoothing and anti-aliasing filters in digital video processors.
Digital Filters

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Subject: Re: Fiter response - frequency domain and time domain
Posted by akhilesh on Tue, 28 Jun 2005 01:49:27 GMT
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THanks, Wayne. So it seems that the bidat uses an algorithm to interpolate that uses the accelaration of prior signal pulses in some way, as opposed to mean  magnitudes of past pulses. I know Norm of audiocrafers guild has done some work on dacs too. No one seems to know how the bidat works (other than Ed Meitner & John Wright!) I know the stuff was proprietary, but does anyone know?I alsoknow,based on my readings that Wadia for example tries to get get the phase anomalies out by optimizing for the time domain, with a rolloff at higher frequencies. For some reason, the bidat (and the IDAT before it) seem to be able to optimize for time and frequency becuase of its interpolation approach. I know it uses FIR filters, and varies the upsampling rate. My best guess is that the upsampling rate variabilty along with the interpolation values predicted on acelration of earlier signals leads somehow to a superior filter. ANyone else care to throw in their comments! I am curious! -akhilesh

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