A 16-Bit A-D-A Conversion System for High-Fidelity Audio Research

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Abstract—**An A-D and D-A converter system with exceptionally wide dynamic range and low distortion is discussed. The system utilizes 12-bit floating-point approximation at conversion instead of true 16-bit conversion. A deglitching track and hold is illustrated which avoids heterodyning between signal components and the sample clock. Rate-limiting during the transition from hold to track is discussed.**

Internal to the system are four active low-pass filters which may be connected under program control. Data transfer to a minicomputer is handled by direct memory access and a 64-word data queue.

I. INTRODUCTION

TRADITIONAL 12-bit analog-digital-analog conversion of high-quality audio is becoming inadequate for audio analysis and synthesis research. The need for greater dynamic range and low distortion has led to the development of a 16-bit converter system at Carnegie-Mellon University designed specifically for audio service. The system has a total dynamic range of 90 dB with less than 0.1 percent distortion at large signal amplitudes. Conversion periods up to 20 μs are programmable with an appropriate low-pass filter automatically connected.

II. FLOATING-POINT APPROXIMATION

Figs. 1 and 2 show schematically the operation of the digital-analog converter (DAC) and analog-digital converter (ADC), respectively. The system first prescales the 16-bit digital (or analog) signal to form a floating-point number. Twelve bits beginning with the first significant bit are taken as a floating-point "fraction," while a 3-bit "exponent" is generated to signify the magnitude of the "fraction." Only the 12-bit "fraction" is converted and afterward the analog (or digital) signal is postscaled by the "exponent" to restore proper magnitude. This technique extends the dynamic range of 12-bit conversion by 24 dB without incurring the expense and stability problems of true 16-bit converters. As with conventional designs, track and hold circuits are employed on the DAC to deglitch the converter, and on the ADC to permit successive-approximation conversion.

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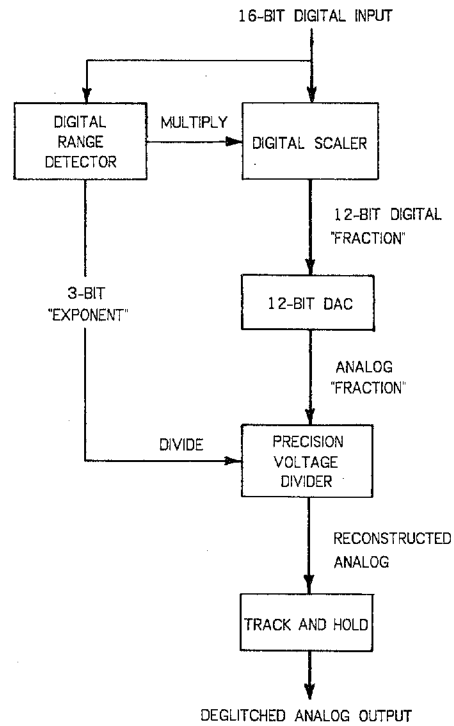

Fig. 1. Floating-point operation of the 16-bit DAC.

III. TRACK AND HOLD

It is not generally recognized that the DAC track and hold can create considerable distortion. During the transition from hold to track, the usual behavior consists of a rate-limited or slewing period followed by quick and exact settling to the new signal level (see Fig. 3). Since the transition is rate-limited, the error of the output is proportional to the square of the transition magnitude, and is not superposition linear in the additive sense. Thus heterodyning effects may occur among signal components, or between the input signal and the sampling clock. For example, two μs is a typical slewing time for a full scale transition. If a maximum amplitude sinewave of 7 kHz is sampled at a rate of 20 kHz, a 1 kHz heterodyne of approximately -35 dB amplitude will be produced. In this case the sinusoid is sampled three times per cycle, and the resulting slewing assymetries repeat every seven cycles.

Changing the track and hold transition behavior to a simple exponential decay eliminates heterodyning. Shown in Fig. 4, the output signal is the convolution of an ideal

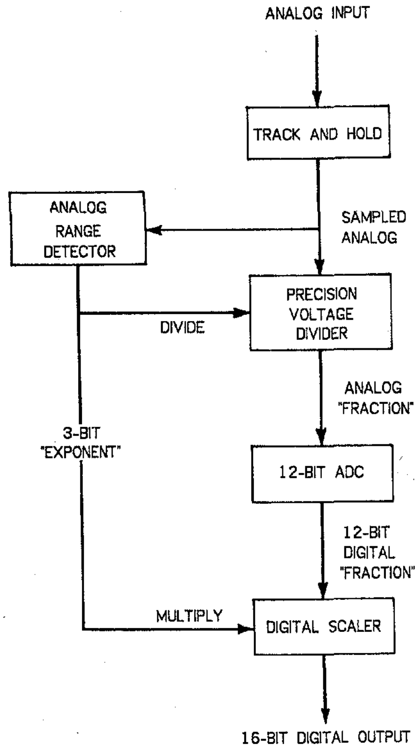

Fig. 2. Floating-point operation of the 16-bit ADC.

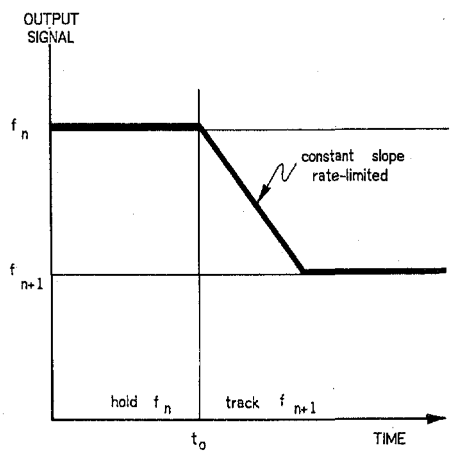

Fig. 3. Rate-limited transition behavior during the step from f_n to f_{n+1}. Deviation from the ideal step is proportional to the square of the step size.

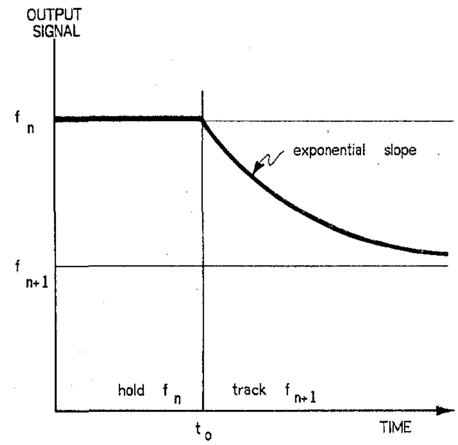

Fig. 4. Exponential transition behavior during the step from f_n to f_{n+1}. Deviation from the step is proportional to the step size.

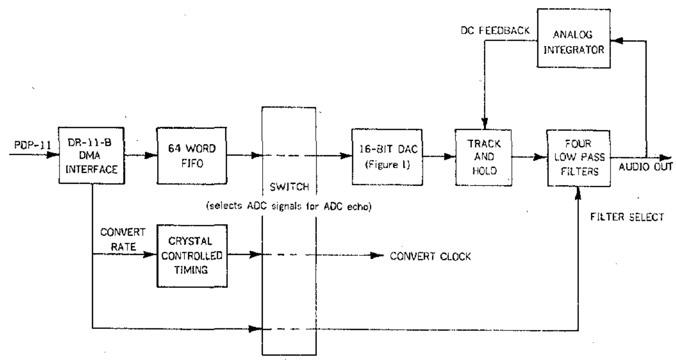

Fig. 5. Complete D-A system.

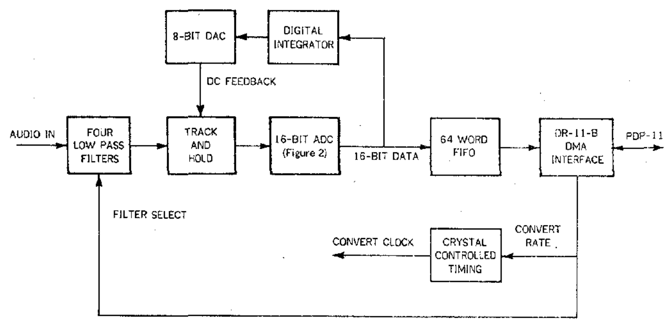

Fig. 6. Complete A-D system.

step and an exponential. The convolution changes the phase and amplitude of signal components, but does not cause heterodyning since superposition linearity is maintained. The track and hold has an exponential time constant of about 0.5 μs and the resulting slight high-frefrequency roll-off is compensated in the low-pass filter.

Since no heterodynes (including dc) are formed in the sampling process, the output can be integrated and fed back around the track and hold to suppress any dc offset. The complete D-A system diagram in Fig. 5 shows that the dc feedback loop includes all amplifiers through the output. The A-D system pictured in Fig. 6 uses a digital integrator and a small DAC to maintain zero digital offset in the data.

IV. LOW-PASS FILTERS

Any audio conversion system clearly must have low-pass filters commensurate with system quality. The frequency-dependent negative resistance (FDNR) active filter configuration permits the design of component tolerant filters with very low distortion and wide dynamic range [1]. To allow flexibility for future requirements, the filters were built as easily modifiable modules: any standard ladder filter of order nine or less can be implemented by changing a few resistors. The present elliptic-function filters have passband to 87 percent of the Nyquist frequency with 0.5 dB measured ripple. The stopband attenuation at the Nyquist frequency and above measures about 70 dB, and signal-to-noise ratio (20 kHz bandwidth) exceeds 95 dB.

Because several conversion rates are required by our user community, four low-pass filters with differing cutoff frequencies are installed in both the A-D and D-A. Keeping the filters inside the converters eliminates many of the noise pick-up problems encountered with very wide dynamic range. The filters include a peaking network to compensate high-frequency roll-off phenomena which are a function of conversion rate [2].

V. SYSTEM FEATURES

Figs. 5 and 6 show the A-D and D-A independently interfaced to a PDP-11 minicomputer. Data, usually divided into large blocks, are transferred by direct memory access (DMA). Processor attention is not required except for interrupt service at the completion of each block transfer. During these interrupts, a 64 word first-in-first-out (FIFO) queue provides several milliseconds of buffering to permit continuous data flow without critical interrupt timing.

A crystal clock divider provides four program selectable conversion rates up to 20 μs. Programing the clock rate simultaneously connects the appropriate low-pass filter from the set of four filters.

For monitoring purposes, the D-A has provision to echo the A-D output independently of the processor. All of the D-A inputs including clock and filter selection automatically switch to echo mode for the duration of A-D operation.

VI. PERFORMANCE TESTS

The D-A system was tested by converting perfect digital sinewaves of varying amplitude and frequency. The fundamental sinewave was removed from the analog output of the converter system with a compensated twin-tee filter and the resulting residue (all noise, harmonic distortion, and heterodynes) is plotted in Fig. 7. For low-amplitude signals, the random noise of the active filter and the DAC quantization noise are about 3 dB above the theoretical minimum quantization noise of the conversion. As the peak sinewave amplitude is increased above 12 bits, conversion noise rises because of the floating-point operation of the converter which truncates low-order bits. In this region total residue is about 0.03 percent, and harmonic distortion becomes noticable only near maximum amplitudes. The increase in residue at 0 dB, 12 kHz input is a heterodyne caused by slight distortion in the active low-pass filter.

The A-D system test was somewhat cumbersome but provides preliminary information until a thorough test can be implemented [3]. Similar to the D-A test, a low distortion sinewave (noise and distortion more than 80 dB down) was converted and the fundamental subtracted (digitally) from the output. The residue was then digitally amplified and reconverted to analog for examination. Fig.

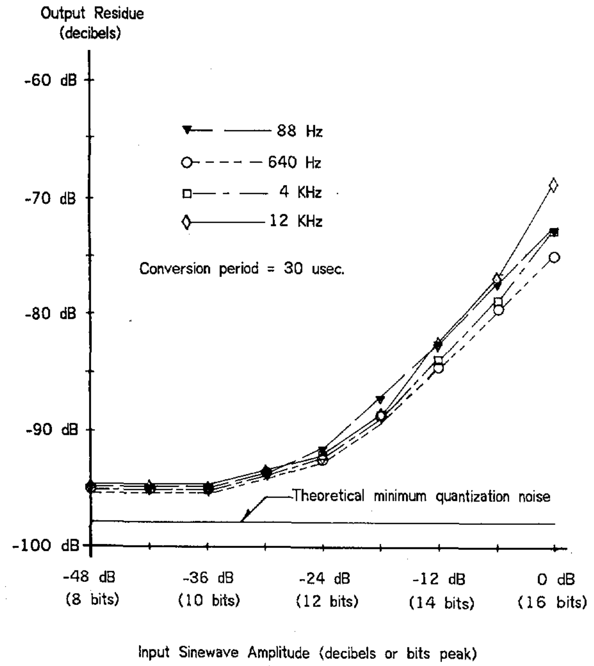

Fig. 7. Performance of the D-A system.

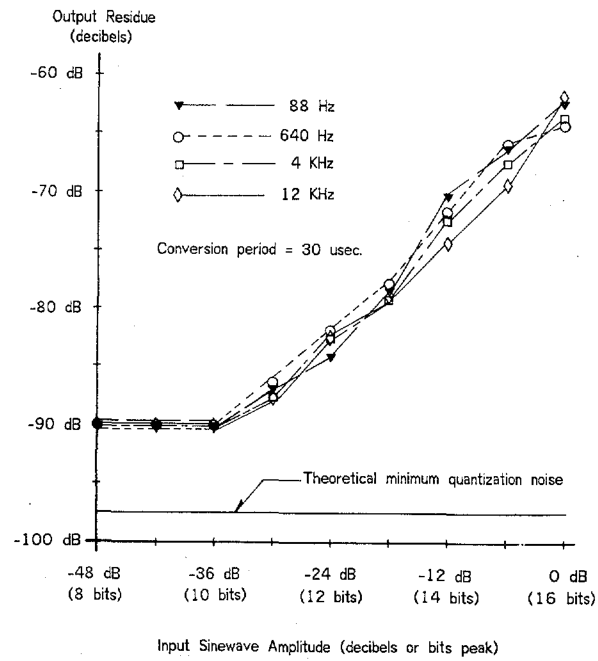

Fig. 8. Performance of the A-D system.

8 shows a noise shelf of about 90 dB: about 8 dB above the theoretical minimum. This higher level is partially attributable to track and hold sampling of high-frequency noise from the filters. In general, quantizing noise at all input amplitudes is higher because of sensitivity of the analog circuits driving the 12-bit ADC. Both of these factors hopefully can be reduced in the near future.

VII. CONCLUSION

A 16-bit A-D-A conversion system has beeen designed specifically for high-fidelity audio service. The system utilizes floating-point approximation at conversion, a nonslewing track and hold circuit, and overall dc feedback. Features include easily modified low-pass filters, a DMA minicomputer interface, and a 64-word data buffer. System performance approaches theoretical limits.

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Book Reviews

Editorial

THIS ISSUE marks the introduction of a book reviews feature in the TRANSACTIONS. Some six months ago, Larry Rabiner asked me to be responsible for this activity. The following months have been spent ordering books, seeking out expert reviewers, allowing for the reviewing process, and waiting for the publication cycle.

The purpose of these reviews is simply to make available the opinions of experts in the particular area covered by the book as to its organization, quality, usefulness as a text (where applicable), and level of difficulty. Beginning with this issue, each following issue will have at least one review until we exhaust the list of available books in the areas of acoustics, speech, and signal processing.

For purposes of completeness, we will review books published within the past three or four years. Although of interest to a smaller percentage of the readership, occasionally books in foreign languages that are believed to contain useful information unavailable in the English language will also be reviewed. An attempt will be made to obtain and publish reviews of new books as soon after publication as possible. I would appreciate hearing from anyone who is presently writing a book or is about to have a book published in one of the areas covered by this TRANSACTIONS and would like to have it reviewed.

We look forward to bringing you, the readership, this new feature, and hope that it provides you with interesting and useful information.

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