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A Low-Voltage Micropower Digital Class-D Amplifier Modulator for Hearing Aids

Victor Adrian, Joseph S. Chang, and Bah-Hwee Gwee, Senior Member, IEEE

Abstract—We present a micropower digital modulator for class-D amplifiers for power-critical digital hearing aids. The modulator design embodies a proposed Lagrange interpolation (a combined first- and second-order Lagrange) algorithmic pulsewidth modulation (PWM) and a third-order $\Delta \Sigma$ noise shaper. By means of double-Fourier-series analysis, we analyze and determine the harmonic nonlinearities of the proposed algorithmic PWM. At 48-kHz sampling, 96-kHz PWM output, 997-Hz input, and input modulation index $I = 0.9$, the modulator circuit achieves a total harmonic distortion + noise ($THD + N$) of $-74$ dB (0.02%) over an 8-kHz voice bandwidth—a 12-dB improvement over a reported design and yet dissipates only $\sim 50\%$ of the power. The proposed modulator dissipates the lowest power dissipation of all modulators compared, and by means of a proposed figure of merit, the proposed modulator exhibits very competitive performance. The modulator IC is fabricated in a 0.35-$\mu$m digital CMOS process with a core area of 0.46 mm$^2$.

Index Terms—Amplifiers, class D, digital circuits, hearing aids, modulators, pulsewidth modulation (PWM).

I. INTRODUCTION

A DIGITAL hearing instrument (hearing aid) is a low-voltage (1.1–1.4-V) power-critical medical instrument whose power supply is limited by the energy capacity of the small pill-sized battery, which is typically 100 mA $\cdot$ h. Its micropower attribute is largely achieved by employing power minimization techniques, including micropower circuit techniques [2], asynchronous-logic processors [3], power-efficient class-D amplifiers (CDAs) [4]–[6], etc., and using nanoscaled processes for IC fabrication.

From hardware and power perspectives, the digital CDA is particularly advantageous, as the traditional approach comprising a hardware-intensive mixed-signal digital-to-analog converter and a power-inefficient linear amplifier is replaced by the simpler (in terms of hardware) and power-efficient all-digital CDA. The digital CDA, as shown in Fig. 1, comprises a digital modulator and an output stage. The digital signal processor embodied in a hearing instrument usually outputs 16-bit data in linear pulse-code modulation (PCM) to the digital modulator of the CDA. This modulator subsequently converts the 16-bit PCM input into 1-bit output modulated pulses that are buffered by a high-power-efficient output stage with low impedance [7] to drive a subminiature loudspeaker (receiver).

The output modulated pulses are typically either width modulated [pulsewidth modulation (PWM)] or density modulated [pulse-density modulation (PDM)]. The simplest PWM scheme, the uniform sampling method, simply involves generation of the modulated pulses from a digital counter clocked at a rate of $f_{\text{fast, clock}} = 2^N \times f_c$ to resolve $2^N$ distinct pulsewidths per input sampling period, where $N$ is the output resolution and $f_c$ is the input sampling rate or the switching frequency. This simplistic method has two major drawbacks. First, the $f_{\text{fast, clock}}$ can be very high and is impractical; for example, in a system where $N = 16$ and $f_c = 48$ kHz, the required $f_{\text{fast, clock}}$ will be $\sim 3$ GHz. Second, the THD can be excessive; for example, the THD is $\sim 30$ dB ($\sim 3\%$) at modulation index $(M) = 0.9$ full scale (FS), $f_c = 48$ kHz, $f_{\text{in}}$ (input modulating signal frequency) = 997 Hz, and bandwidth ($BW$) = 100 Hz–8 kHz.

The PDM scheme is a special case of the delta–sigma ($\Delta \Sigma$) modulation—specifically, the PDM is a 1-bit $\Delta \Sigma$ modulation. The large in-band quantization noise arising from the quantization of the $N$-bit ($N$ is typically 16) input to 1 bit is usually mitigated by employing either a sampling frequency with a high oversampling ratio or a high-order $\Delta \Sigma$ noise shaper. These usual techniques are incongruous with the power- and area-critical demands of the hearing instrument. This is because a high oversampling requires several multistage interpolation filters, and the high sampling frequency reduces the power efficiency of the output stage [7], [8] and raises the need for a higher rate modulator computational circuit (loop filter). Furthermore, a high-order $\Delta \Sigma$ noise shaper tends to be more prone to instability [9] which may undessimably limit the maximum input signal allowed, hence compromising the dynamic range. The well-established method to circumvent the limitations of the 1-bit $\Delta \Sigma$ modulation (PDM) is the multibit $\Delta \Sigma$ modulation that quantizes the $N$-bit input to multibit (less than $N$) output. The ensuing difficulty is that the multibit output cannot be driven by the (single-bit) CDA output stage. Instead, the usual approach involves conversion of the multibit output into PWM-type pulses (for the CDA output stage to be used)—effectively, the output is modulated using the uniform sampling method. Hence, as expected, the PWM-type output suffers from similar unacceptable harmonic distortion as in the uniform sampling method.

The well-adopted scheme, shown in Fig. 2, that collectively addresses the shortcomings of the uniform sampling method (PWM scheme) and 1-bit $\Delta \Sigma$ modulation (PDM scheme) is the hybrid/combination of an algorithmic PWM and the multibit $\Delta \Sigma$ modulation methods. The algorithmic PWM is an algorithm
that outputs pulsedwidth data from the PCM input and mitigates the harmonic distortion of the uniform sampling method, e.g., the cross-point deriver algorithms [4] (see Section II-A-I later). The multibit $\Delta\Sigma$ modulation, on the other hand, quantizes down the pulsedwidth data into lesser number of bits so that the required $f_{\text{fast, clock}}$ can be reduced. The quantized data are subsequently converted into the final PWM-type pulses for the CDA output stage. This hybrid method was first described in [12], and its realizations have been reported elsewhere in [4] and [13].

In this paper, we present the analyses, design, and implementation of a novel digital modulator (for a CDA) based on the hybrid method for area- and power-critical 8-kHz-voice-bandwidth digital hearing instruments and with relatively low nonlinearity (near high-fidelity standards). This design is for advanced hearing instruments that demand higher fidelity without the usual IC area and power dissipation overheads. Specifically, the nonlinearity specification for a 16-bit PCM input is $THD+N < -70$ dB (0.03%) which far exceeds that specified for typical current hearing instruments whose typical $THD+N$ of the complete path is $-40$ dB or 1% [14]. The other nonlinearities are below the noise floor—specifically, the intermodulation distortion (IMD) between the input and carrier signals, $IMD \ll -80$ dB; this IMD is sometimes referred to as foldback distortion. In this paper, to benchmark the proposed modulator against reported modulators, we propose a new figure of merit (FOM) which is a modification of that originally used for analog-to-digital converters (ADCs). This FOM takes into consideration the bandwidth, $THD+N$, and power dissipation (normalized to $V_{\text{DD}}^2$, hence independent of $V_{\text{DD}}$). On the basis of this proposed FOM, the proposed digital modulator’s performance ranks among the very best and is the lowest power among the reported digital modulators.

The hybrid method embodied in the proposed modulator is the proposed algorithmic PWM, which is denoted as the “combined first- and second-order Lagrange interpolation algorithm” (LAGI) [1], and is an improvement on our earlier linear interpolation (equivalent to the first-order Lagrange interpolation). The objective of the proposed algorithm is to further suppress the harmonic distortion, thereby improving the $THD+N$ of the uniform sampling method with small hardware and power penalties (see Section IV later).

In this paper, by means of double-Fourier-series analysis, we further investigate the harmonic distortion of the proposed LAGI to determine the mechanisms thereof, thereby providing insight into the parameters that influence the distortion. This insight is particularly useful to designers because it delineates the available parameters to vary or tradeoff to meet a given specification. Furthermore, we derive an expression to analytically determine the THD, and this expression is useful to predict the THD without resorting to time-consuming simulations or measurements (as an afterthought). We verify our analyses and proposed CDA modulator design on the basis of simulations and on measurements on a fabricated prototype IC. The design features an overall $THD+N$ of $-74$ dB (0.02%) at $M = 0.9$ FS over an 8-kHz voice bandwidth and dissipates a very modest 31-$\mu$W average power at 1.1 V, which is thus highly applicable to an area- and power-critical advanced hearing instrument with low nonlinearities. On the basis of simulations, this design features a 12-dB THD improvement and yet dissipates only $\sim 50\%$ power of an earlier design [4].

This paper is organized as follows. Section II details designs of the building blocks of the proposed digital modulator. Section III describes the proposed FOM. Section IV presents the simulation and IC measurement results of the proposed modulator and compares the FOM values of recently reported digital modulators in the literature. Finally, conclusions are drawn in Section V.
II. DIGITAL MODULATOR DESIGN

The functional blocks of a typical CDA and of the digital modulator based on the hybrid method were earlier shown in Figs. 1 and 2, respectively. As a preamble to the proposed LAGI, we will succinctly review one of the rudimentary algorithmic PWM methods, a type of cross-point deriver known as linear interpolation. The proposed LAGI is the combined first- and second-order Lagrange interpolation algorithm (LAGI), which is also an algorithmic PWM of the cross-point deriver type. The double-Fourier-series analysis will thereafter be applied to determine the mechanisms of the harmonic nonlinearities and used for deriving an expression to analytically determine THD.

A. Algorithmic PWM

1) Cross-Point Deriver Review: Fig. 3 shows the uniform sampling method along with its analog modulator counterpart (for analog input signal or digital with infinite sampling) known as the natural sampling method, both using single-sided trailing-edge modulation. As shown in the figure, natural sampling is ideal, as an exact cross point, \( S_{NS} \), between the modulating input signal and the carrier signal is obtained. This exact cross point translates to an exact pulsewidth, \( t_{p(NS)} \), and hence has an ideal zero THD (in the baseband of output pulses) [15]. Uniform sampling, on the other hand, results in \( S_{US} \) and \( t_{p(US)} \) that may differ significantly from \( S_{NS} \) and \( t_{p(NS)} \), respectively. The resulting nonlinear output has significantly higher THD in its baseband, e.g., 3% as discussed earlier.

The cross-point deriver algorithmic PWM, such as linear interpolation shown in Fig. 3, improves the linearity of uniform sampling by finding a closer estimation to the ideal intersection point \( S_{NS} \). For completeness, note that there exists [4] a plethora of reported cross-point derivers and other pertinent algorithmic PWM algorithms.

2) Combined First- and Second-Order Lagrange Interpolation Algorithm (LAGI): The proposed LAGI algorithm is shown in Fig. 4, where \( S_2 \) is the current sampled datum at time \( t = T \), and \( S_1 \) and \( S_0 \) are the previous sampled data at time \( t = 0 \) and \( t = -T \), respectively. This method initiates by first approximating the modulating signal arc with a second-order Lagrange interpolation [16] polynomial \( p(t) \) that passes through the three sampled points \( (S_0, S_1, \text{and } S_2) \). Mathematically, the general \( R \)th-order Lagrange interpolation polynomial is of the form

\[
p(t) = \sum_{k=0}^{R} I_k(t) y_k \quad (1)
\]

\[
I_j(t) = \prod_{k=0, k \neq j}^{R} \frac{(t - t_k)}{(t_j - t_k)} \quad (2)
\]

Note that for \( R = 1 \), (1) will result in linear interpolation. The polynomial \( p(t) \) for \( R = 2 \) can be obtained as follows:

\[
p(t) = \frac{t^2(S_0 - 2S_1 + S_2) + t(-S_0 + S_2)T + 2S_1T^2}{2T^2} \quad (3)
\]

for \(-T \leq t \leq T\).

To obtain the pulsewidth, the intersection point of (3) and the carrier ramp \( c(t) = (1/T)t \) for \( 0 \leq t \leq T \) can be solved, but this would involve a square-root computation, which is a complex process. We instead propose to simply substitute \( t = (1/2)T \) into (3) to interpolate a middle point denoted as \( S_M \) between \( S_1 \) and \( S_2 \), and subsequently employ the cross-point deriver linear interpolation to obtain \( S_{1LP} \) and \( S_{2LP} \).

The middle point \( S_M \) can hence be simply determined as

\[
S_M = \frac{S_0 + 6S_1 + 3S_2}{8} = \frac{1}{8}(-S_0 + 4S_1 + 2S_1 + 2S_2 + S_2). \quad (4)
\]

In a hearing instrument where \( f_c \) is typically 48 kHz, the introduction of \( S_M \) effectively increases the input sampling rate and the computational frequency of the algorithmic PWM and the \( \Delta S \) noise shaper to \( f_{2L} = L \times f_c = 2 \times 48 \text{ kHz} = 96 \text{ kHz} \), where \( L \) is the number of interpolations (two in this case). For completeness, note that if a different reference 0 for the time axis is used in Fig. 4, a different expression for (3) may be obtained, but (4) would remain unaffected.
The effective oversampling arising from the interpolation of \( S_M \) reduces the nonlinearities of the output of the algorithmic PWM (and the same effect to the quantization noise at the output of the noise shaper); see the next section for the double-Fourier-series analysis of the LAGI. Physically, the interpolation process renders the distance between successive samples to become smaller, thereby providing samples that have finer quantization in time for the algorithmic PWM. Further improvements to the performance can be obtained by interpolating more data in-between \( S_1 \) and \( S_2 \), but at the expense of increasing (hardware) complexity and higher switching frequency, both of which are incongruous with the desired attributes of a hearing instrument.

The doubling of the input sampling rate \( f_c \) to \( f_{2c} \) also correspondingly increases the carrier frequency to \( f_{2c} \). This is shown in Fig. 4 where the single ramp of the carrier signal \( c(t) \) is now replaced by two ramps \([c_{\text{I}1}(t) \text{ and } c_{\text{I}2}(t)]\) in one \( T \) period. These two ramps intersect with the linear interpolation lines \( l_1(t) \) and \( l_2(t) \) to obtain the two crossover points \( S_{\text{I}1}\) and \( S_{\text{I}2}\). The corresponding pulsewidths of \( S_{\text{I}1}\) and \( S_{\text{I}2}\) can be obtained as follows:

\[
\begin{align*}
t_{\text{p}(\text{I}1)} &= T' \frac{S_1}{1 + S_1 - S_M} \\
t_{\text{p}(\text{I}2)} &= T' \frac{S_M}{1 + S_M - S_2}
\end{align*}
\]  

(5)

(6)

where \( T' = 1/2T = 1/f_{2c} \).

The divider for the aforementioned two division operations can be implemented using the (simplified) parallel divider [17] described in [4], which is a small hardware with modest power requirements.

In a practical implementation, (5) and (6) are used for computing the \( S_1/(1 + S_1 - S_M) \) and \( S_M/(1 + S_M - S_2) \) values which will be converted to the pulsewidths \( t_{\text{p(LI)1}} \) and \( t_{\text{p(LI)2}} \) at the PWM generator stage (after quantization at the \( \Delta N \) noise shaper; refer to Fig. 2). The switching frequency of the resulting PWM-type pulses will be that of the new carrier frequency \( f_{2c} \).

Note that as with any other interpolation process, the interpolated output \( (S_M) \) has some errors. These errors cause spectral-image by-products (in the frequency domain) that are beyond the audio band, and in view of their wasted power dissipation, these by-products should be kept small. To determine the lowest image attenuation (the image closest to the bandwidth of interest), a tone signal at the extreme of the upper band edge, \( f_{\text{BW,U}} \), is input to determine the power of its image component located at \( f_{\text{image}} = f_{2c}/L - f_{\text{BW,U}} \). The resulting image gain \( (G) \) (negative of image attenuation) is

\[
G = 10 \log \left( \frac{\text{Power at } f_{\text{image}}}{\text{Power at } f_{\text{BW,U}}} \right). \tag{7}
\]

In the hearing instrument whose bandwidth is 8 kHz, the test tone signal at \( f_{\text{BW,U}} = 8 \text{ kHz} \) with \( M = 1.0 \text{ FS} \) has a corresponding image at \( f_{\text{image}} = 40 \text{ kHz} \) with \( G = -29 \text{ dB} \). As the image is located well above the upper voice-band band edge, it will be attenuated by the analog low-pass filter after the output stage (Fig. 1). At \( f_{\text{image}} = 40 \text{ kHz} \), the attenuation of a second-order Butterworth low-pass filter with 8-kHz cutoff is 41 dB, yielding a total image attenuation of 70 dB. Put simply, as

the image is substantially attenuated, the power dissipation due to the image by-products is negligible. As a matter of interest, the usual interpolation method of zero insertion (zero stuffing) followed by low-pass filtering using the popular sinc interpolation filter would require a fifth-order sinc filter at twice interpolation to obtain the same \( G = -29 \text{ dB} \) (see (8)–(10), shown at the bottom of the next page). Physically, a direct implementation of \( \sin c^2 \) filter would require four adders at \( f_{2c} = 96 \text{ kHz} \) (equivalent to eight adders at \( f_c = 48 \text{ kHz} \)) which is relatively expensive compared to the simpler implementation of \( S_M \) in (4) that requires only four adders.

3) Double-Fourier-Series Analysis of the LAGI Pulses: In view of the deviation of the \( S_{\text{LI}1} \) and \( S_{\text{LI}2} \) from the exact natural sampling cross points, it is instructive to analyze and determine the harmonic nonlinearities in the LAGI pulses (quantization noise is not considered at this juncture and will be discussed separately in the next section). This analysis is performed by applying the double-Fourier-series analysis method [15] to the LAGI pulses. At the outset, note that, in [1], the double Fourier series of the LAGI pulses was inadvertently approximated to the double Fourier series of the linear interpolation pulses at twice the input sampling rate. In other words, \( S_M \) was assumed to be a true datum (not interpolated) and sampled using the new sampling rate (twice the original). A more precise analytical approach that is adopted here is to allow \( S_M \) to be a function of the other sampled data \( (S_0, S_1, \text{ and } S_2) \) at the original \( f_c \) frequency given in (4). To cater for the two PWM-type pulses with frequency \( f_{2c} \), the entire LAGI process is treated as having a period of \( T = 1/f_c \), but in each period, two PWM pulses are generated. The derivation using this approach is delineated in the Appendix.

The corresponding double Fourier series for the PWM-type pulses generated by the single-sided trailing-edge LAGI of a sinusoidal input modulating signal can be obtained as follows:

\[
\begin{align*}
\int_0^{2\pi} e^{-j2\pi k_0 f} df &= \begin{cases} 0, & n \neq 0 \\ \frac{2\pi}{2\pi N}, & n = 0 \end{cases} \\
K_{0n} &= K_{mn} \big|_{m=0} \\
K_{00} &= K_{mn} \big|_{m=0}. \tag{11} \tag{12} \tag{13}
\end{align*}
\]

\( k \) is the duty cycle ratio of the PWM-type pulses for a zero input modulating signal (ideally \( k = 0.5 \)), and \( V_{\text{ref}} \) is the reference value, e.g., the supply voltage, which determines the amplitude of the PWM-type pulses and can be normalized to unity for convenience of analysis.

Equation (8) is interpreted as follows. The first term \( K_{00} \) is the inconsequential dc component of the pulses. The second term corresponds to the output modulating signal and its harmonics. The third term represents the carrier signal and its harmonics. The fourth term corresponds to the IMD components that arise from the input modulating signal and its harmonics intermodulated with the carrier and the carrier’s harmonics.

Fig. 5 shows the (single-sided) power spectrum of the LAGI pulses from the double Fourier series in (8) for \( f_{\text{in}} = 997 \text{ Hz}, M = 0.9 \text{ FS}, \) with the infinite terms approximated by limiting them to 200, and the spectra filtered by a second-order Butterworth low-pass filter with 8-kHz cutoff frequency (hearing

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The domain linear model is shown in noise shaper is implemented using the error feedback—commensurable with that for a high-fidelity hearing instrument. From a practical perspective, this is much lower than the typical THD (−40 dB or 1%) of hearing instruments.

In Section IV, the THD obtained from power spectral density simulations and measurements of the complete modulator design (including the ΔΣ noise shaper) will be compared against the THD result obtained analytically from the second term in (8). For completeness, note that the other terms (third and fourth terms) in (8) are difficult to be verified from simulations because these terms are masked by the quantization noise floor. Another verification, albeit somewhat crude, is a visual comparison between the waveform obtained from the conversion of the double-Fourier-series expression to the time domain against the PWM-type waveform at the expected frequency $f_{c2}$.

### B. ΔΣ Noise Shaper

The ΔΣ noise shaper is implemented using the error feedback structure [18], [20] whose z-domain linear model is shown in Fig. 6. This structure is chosen for its simplicity, including the following.

1) The quantizer is a simple truncation of the least significant bits of $Y(z)$, and hence, the quantization noise $E(z)$ can be easily obtained.

2) From Fig. 6, the feedback filter $H_f(z)$ is a simple (low-complexity) finite-impulse-response filter expressed as $NTF(z) = 1$, where the noise transfer function $NTF(z) = (1 - z^{-1})^D$ [10] and $D$ is the order of the ΔΣ

\[
\psi_{\text{LAG-1}}(t) = K_{00} + \sum_{n=\pm 1}^{\pm \infty} (K_{0n} e^{j(n\omega_{in} t)}) + \sum_{m=\pm 1}^{\pm \infty} \sum_{n=\pm 1}^{\pm \infty} (K_{mn} e^{j(m\omega_c t + n\omega_{in} t)})
\]

where
\[
\omega_{in} = \text{input modulating signal angular frequency} = 2\pi f_{in},
\]
\[
\omega_c = \text{carrier angular frequency} = 2\pi f_c,
\]
\[
K_{00} = \frac{V_{\text{ref}}}{4\pi^2} \left( \int_{\phi=0}^{\Phi=2\pi} \frac{8\pi (2k + M \cos(\Phi))}{16 + M \cos(\Phi - \frac{2\pi}{p}) + 2M \cos(\Phi) - 3M \cos(\Phi + \frac{2\pi}{p})} \right)
\]
\[
\cdot \left( \int_{\phi=0}^{\Phi=2\pi} \frac{\pi (48k - 2M \cos(\Phi - \frac{2\pi}{p}) + 12M \cos(\Phi) - 2M \cos(\Phi + \frac{2\pi}{p}))}{16 - M \cos(\Phi - \frac{2\pi}{p}) + 6M \cos(\Phi) - 5M \cos(\Phi + \frac{2\pi}{p})} d\Phi \right) - 2\pi^2
\]
\[
K_{mn} = \frac{jV_{\text{ref}}}{4 (m + \frac{n}{p}) \pi^2} \left( \int_{\phi=0}^{\phi=2\pi} e^{-j \phi} \left( e^{-j(m+\frac{n}{p})} \left( \frac{8\pi (2k + M \cos(\Phi))}{16 + M \cos(\Phi - \frac{2\pi}{p}) + 2M \cos(\Phi) - 3M \cos(\Phi + \frac{2\pi}{p})} \right)
\right.
\]
\[
\left. + e^{j(m+\frac{n}{p})} \left( \frac{m(48k - 2M \cos(\Phi - \frac{2\pi}{p}) + 12M \cos(\Phi) - 2M \cos(\Phi + \frac{2\pi}{p}))}{16 - M \cos(\Phi - \frac{2\pi}{p}) + 6M \cos(\Phi) - 5M \cos(\Phi + \frac{2\pi}{p})} \right) \right)
\]
\[
- e^{-j(m+\frac{n}{p})} \right) d\Phi - \int_{\phi=0}^{\phi=2\pi} e^{-j \phi} d\Phi \right) \quad \forall m, n \in \mathbb{Z}
\]
noise shaper. The coefficients for $H_f(z)$ can simply be obtained as powers of two or additions of powers of two.

For this structure, it can be shown that the output signal-to-quantization-noise ratio (SQNR) for an $N$-bit ($N > Q$) sinusoidal input signal (with the same assumptions assumed in [11]) is given by

$$\text{SQNR (dB)} \approx 6.02Q + 1.76 + 20 \log(M) + 10 \log\left(\frac{2D+1}{\pi 2D}\right) + (2D+1)10 \log\left(\frac{f_s}{2f_{BW,U}}\right)$$  \hspace{1cm} (14)$$

where $Q$ is the number of output bits and $f_s$ is the sampling frequency of the $\Delta\Sigma$ noise shaper; in our case, $f_s = f_{c2} = 96$ kHz.

In the proposed design, $Q$ is set to eight (see also Section II-C hereinafter), and $D$ is set to three. Thus, at $M = 0.9$ FS, the approximate SQNR using (14) is 82 dB. By combining this SQNR value with the analytical THD value obtained from the previous section, the overall analytical $THD + N$ of the hybrid modulator system is $-77$ dB which more than satisfies the desired maximum target $THD + N$ of $-70$ dB. On the basis of dc-input time-domain simulations, the $\Delta\Sigma$ noise-shaper implementation is stable up to 0.98 FS.

### C. PWM Pulse Generator

The PWM pulse generator converts the $Q$-bit digital value from the $\Delta\Sigma$ noise shaper into PWM-type pulses by means of a $Q$-bit digital counter. This counter is clocked at a rate of $f_{\text{fast, clock}} = 2^Q \times L \times f_c$, where it counts up per $1/f_{\text{fast, clock}}$ period and sets the pulsewidth when the counter reaches the $Q$-bit value. The output PWM-type pulses have a switching frequency of $f_{c2} = L \times f_c$, where $f_{c2} = 96$ kHz here.

An important consideration in the design is its power dissipation in relation to its number of bits ($Q$), its $f_{\text{fast, clock}}$ frequency, and the SQNR of the $\Delta\Sigma$ noise shaper. On the basis of simulations, Table I tabulates the SQNR and average power dissipation of the PWM pulse generator for several $Q$-bit configurations using a 0.35-$\mu$m CMOS process at 1.1-V supply and $M = 0.9$ FS. As shown in Table I and as expected, the power dissipation approximately doubles for every 1-bit increment of $Q$ or, equivalently, for every doubling in the $f_{\text{fast, clock}}$. As a compromise in the proposed design, $Q$ is selected to be 8 bits, where the average power dissipation of the PWM pulse generator is a modest 5.8 $\mu$W and the required $f_{\text{fast, clock}}$ is $2^8 \times 96$ kHz $\approx 24.6$ MHz. By adopting our frequency doubler [4], the $f_{\text{fast, clock}}$ is halved to 12.3 MHz—this is a modest and practical clock frequency compared to the $f_{\text{fast, clock}}$ requirement of $\sim 6$ GHz for the uniform sampling PWM method.

For hardware efficiency, the counter in the PWM pulse generator also operates as a frequency divider by dual functioning its outputs as clock and control states for the finite-state machine of the computational circuits.

### III. FOM

From a technological perspective, it is usually of interest to benchmark new designs against reported designs. Nevertheless, comparing different designs, including CDA modulators, is often contentious because of the interdependent parameters such as the invariably diverse bandwidths, supply voltages, etc., thereof.

The FOM [21] (and its decibel-scaled form [19]) that is often used for comparing the performance of various ADCs is

$$FOM_{\text{ADC}} = DR_{\text{dB}} + 10\log\left(\frac{BW}{P}\right)$$  \hspace{1cm} (15)$$

where $DR$ is the dynamic range defined as $M(DR_{\text{dB}} = -M_{\text{dB}})$ for which $THD + N = 0$ dB, and $P$ is the power dissipation of the ADC.

We propose a modified FOM of (15) given hereinafter in (16) for comparing CDA modulators to account for the following.

### TABLE I

<table>
<thead>
<tr>
<th>$Q$ (bit)</th>
<th>SQNR (dB)</th>
<th>$f_{\text{fast, clock}}$ (MHz)</th>
<th>Average Power Dissipation (\mu W)</th>
</tr>
</thead>
<tbody>
<tr>
<td>7</td>
<td>76</td>
<td>6.1</td>
<td>2.9</td>
</tr>
<tr>
<td>8</td>
<td>82</td>
<td>12.3</td>
<td>5.8</td>
</tr>
<tr>
<td>9</td>
<td>88</td>
<td>24.6</td>
<td>12.0</td>
</tr>
<tr>
<td>10</td>
<td>94</td>
<td>49.2</td>
<td>26.0</td>
</tr>
</tbody>
</table>

1 Simulated using a 0.35 $\mu$m CMOS process at 1.1 V supply and $M = 0.9$ FS.

### TABLE II

<table>
<thead>
<tr>
<th>Component</th>
<th>Number of Transistors (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Algorithmic PWM and $\Delta\Sigma$ Noise-Shaper</td>
<td>24910 (97 %)</td>
</tr>
<tr>
<td>PWM Pulse Generator</td>
<td>812 (3 %)</td>
</tr>
</tbody>
</table>

Authorized licensed use limited to: Nanyang Technological University. Downloaded on February 26, 2010 at 00:23:32 EST from IEEE Xplore. Restrictions apply.
First, instead of DR that is typically given in ADC specifications, the modified FOM uses (the negative of) $THD + N$ that is typically given for CDA modulators and accounts for the applied input $M$ when measuring the $THD + N$. Note that $[- (THD + N_{dB}) - M_{dB}]$ is equivalent to $DR_{dB}$ if $THD + N$ is proportional to $M$. For example, in a 16-bit Nyquist rate ADC, its DR is ideally 98 dB at $M = -98$-dB FS, the $THD + N = 0$ dB. Conversely, at $M = 0$-dB FS, the $THD + N$ is ideally $-98$ dB. Second, to accommodate different supply voltages that would otherwise skew the FOM, we propose to normalize the power parameter $P$ by dividing it by the squared voltage of the supply ($V_{DD}^2$)

$$FOM_{CDAmod} = \left[ -(THD + N_{dB}) - M_{dB} \right] + 10 \log \left( \frac{BW}{V_{DD}^2} \right).$$ (16)

This $FOM_{CDAmod}$ will be used as the FOM to compare reported and the proposed CDA modulators in the following.

IV. SIMULATION AND MEASUREMENT RESULTS

Fig. 7 shows the microphotograph of a prototype IC embodying the proposed digital modulator and an output stage, using a 0.35-$\mu$m CMOS process. The IC area of the modulator is a relatively small area of 0.46 mm$^2$. Table II tabulates the number of transistors and the respective relative IC areas of the combined algorithmic PWM and the $\Delta\Sigma$ noise shaper and of the PWM pulse generator in the proposed digital modulator design.

The input datum to the modulator is a pregenerated 16-bit unsigned digital sinusoidal waveform of frequency $f_{in} = 997$ Hz (AES standards [22]) and sampled at $f_s = 48$ kHz. The SQNR of this pregenerated digital sinewave is 102 dB for $M = 0.9$ FS and $f_{BW,UL} = 8$ kHz and is sufficiently lower than the analytical $THD + N$ ($THD$ and SQNR) of the proposed modulator. The $THD + N$ measurements are performed using the Bruel & Kjær PULSE Analyzer (Frame Type 3560C and Input/Output Measurement Module Type 3110) in conjunction with the power spectral density analysis using the PULSE Labshop software.

Table III gives the average power dissipation obtained from simulations and from measurements of the proposed digital modulator (1.1 V).

<table>
<thead>
<tr>
<th>Input</th>
<th>Simulation ($\mu$W)</th>
<th>Measurement ($\mu$W)</th>
</tr>
</thead>
<tbody>
<tr>
<td>$M = 0.9$ FS</td>
<td>27</td>
<td>31</td>
</tr>
<tr>
<td>$M = 0$ FS (quiescent)</td>
<td>18</td>
<td>20</td>
</tr>
</tbody>
</table>

Fig. 8(a) and (b) shows the power spectral density of the proposed digital modulator versus $M$ at $f_{in} = 997$ Hz, and $M = 0.9$ FS after low-pass filtering. (a) Simulation results, (b) Measurement results.
simulated SQNR, on the other hand, is 84 dB and largely agrees with the SQNR obtained from the analytical expression in (14) at 82 dB, and the small difference is attributed to the approximations used in the derivations of (14).

The simulated THD+N obtained by summing the harmonic power with the quantization noise power is −78 dB, and on the basis of simulations in Fig. 8(a), this is a 12-dB improvement compared to the first-order Lagrange interpolation design in [4]. The measured THD+N in Fig. 8(b), on the other hand, is −74 dB (0.02%). The 4-dB difference between the simulations and measurements is largely due to the supply noise in the low-frequency region (< 1 kHz). In the perspective of the specifications for an advanced hearing instrument, the measured THD+N still satisfies the maximum target THD+N of −70 dB; note that it is expected that the low-frequency noise is further mitigated in practical battery cell operation in hearing instruments. For completeness, note that the component (the first point) with about 0-dB power spectral density in both figures is an inconsequential dc component.

Fig. 9 shows the THD+N measurement results versus $f_{in} = 997$ Hz. The average measured THD+N for the range of $M = 0.1$–0.9 FS is −70 dB (0.03%). It is worth noting that at high modulation indexes ($M > 0.8$ FS), the harmonic distortions (THDs) dominate the THD+N. This is because the distance between samples is larger, and consequently, the errors in the interpolation and algorithmic PWM processes increase. On the other hand, as the modulation index decreases, the signal magnitude decreases while the noise floor remains unchanged (throughout all of the modulation indices), hence the THD+N increases.

Table III tabulates the average power dissipation obtained from simulations and from measurements of the digital modulator IC for two modulation indexes at 1.1-V supply. The simulated and measured power dissipation results agree well.

Table IV tabulates a comparison on the basis of the proposed FOM$_{CD\text{,Ammol}}$ of recently reported digital modulator designs with various bandwidths and supply voltages. The comparison depicts that the proposed modulator design is very competitive. Note that although the FOM of the reported design [23] appears to be slightly better, the computation of its FOM does not include any digital interpolation circuits which may otherwise increase the power dissipation, hence negatively skewing its FOM slightly. A further worthy attribute of the proposed design is its lowest power dissipation attribute in the comparison, commensurable for a practical advanced digital hearing instrument.

### V. Conclusion

We have presented a digital modulator design embodying a proposed LAGI algorithm for low-voltage micropower operation such as power-critical hearing instruments. We have also derived the double-Fourier-series expression of the LAGI algorithm to analytically determine the harmonic nonlinearities therein. The modulator design and the derived analytical double-Fourier-series expression were verified by means of simulations and measurements on a prototype IC. The modulator featured low nonlinearity with low computational/output frequency requirements and micropower dissipation at 1.1 V. We have also presented a proposed FOM to compare various modulators, and the proposed modulator was shown to be very competitive and features the lowest power dissipation.

### Appendix

#### Three-Dimensional Geometrical Configuration

The double-Fourier-series analysis method [15] is applied to the PWM pulses generated by the single-sided trailing-edge LAGI of a sinusoidal input modulating waveform. This method reconstructs the PWM-type pulses using the 3-D geometrical configuration in [24] and is shown in Fig. 10 for the LAGI pulses. In this illustration, the shaded areas represent walls...
that are parallel and have identical shapes to each other. The walls are flat on one side, while their other sides are shaped according to the input modulating signal. The height of the walls represents $V_{\text{ref}}$.

The $\phi$ axis in Fig. 10 represents the phase shift of the input modulating signal.

$$\phi = 2\pi f_{\text{in}} t = \omega_{\text{in}} t. \quad (A1)$$

On the other hand, the $\theta$ axis represents the phase shift of the original carrier signal ($f_c = 2f_{\phi}$), i.e.,

$$\theta = 2\pi f_c t = \omega_c t. \quad (A2)$$

This is because $S_M$ is not a true datum sampled at $f_{c2}$ frequency but a function of the other sampled datum at $f_c$, given in (4). This $\theta$ axis would impose difficulty in representing the output PWM-type pulses which are at $f_{c2}$. To circumvent this problem, the whole process of LAGI can be treated as having a period of $T = 1/f_c$, whereby, in each period, two PWM pulses are generated (at $f_{c2}$).

In the time domain, the LAGI employs two linear interpolations which result in two PWM-type pulses for every $T$ period. In the corresponding ($\theta, \phi$) domain, these two linear interpolations occur every $2\pi$ length of $\theta$. Since each of them needs to take place on an input modulating signal wall, two identical walls are erected per $2\pi$ of $\theta$. Fig. 10 shows the two linear interpolations that occur in $0 \leq \theta < \pi$ and $\pi \leq \theta < 2\pi$.

For the first wall that is erected on $0 \leq \theta < \pi$, the shape of its sinusoidal side can be defined as

$$\theta = \pi k + \frac{M\pi}{2} \cos(\phi), \quad \text{for } 0 \leq \theta < \pi \quad (A3)$$

while for the second wall on $\pi \leq \theta < 2\pi$, the shape of its sinusoidal side can be defined as

$$\theta = 3\pi k + \frac{M\pi}{2} \cos(\phi), \quad \text{for } \pi \leq \theta < 2\pi \quad (A4)$$

where $k$ is the duty cycle ratio of the PWM-type pulses for a zero input modulating signal (ideally, $k = 0.5$).

The line $AA'$ in Fig. 10 corresponds to the new carrier signal in Fig. 4. The portion of the line within $0 \leq \theta < \pi$ corresponds to the ramp of $c_{\text{in}}(t)$ in Fig. 4, while the portion within $\pi \leq \theta < 2\pi$ corresponds to the new carrier signal $c(t)$ in Fig. 4.
$2\pi$ corresponds to the ramp of $c_{n2}(t)$. The line AA' is expressed as

$$\phi = \frac{1}{p}\theta$$

(A5)

where

$$\frac{\omega_{\text{m}}}{\omega_c} = \frac{2}{p}$$

(A6)

The line BB' in Fig. 10 corresponds to the contour of uniform sampling that samples the input signal periodically in the time domain. Correspondingly, the line BB' also periodically samples the input signal which is now represented by the walls in the $(\theta, \phi)$ domain. The line BB' samples two points ($\theta_1$ and $\theta_M'$) from the two signal walls in every $2\pi$ period of $\theta$ and can be expressed as

$$\phi = \left\lfloor \frac{\theta}{\pi} \right\rfloor \frac{\pi}{p}$$

(A7)

where $\lfloor \theta/\pi \rfloor$ denotes the nearest integer that is less than or equal to $\theta/\pi$.

In Fig. 10, the points $\theta_0$, $\theta_1$, and $\theta_2$, which are located within $0 \leq \theta < \pi$, correspond to the sampled data $S_0$, $S_1$, and $S_2$ in Fig. 4, respectively. These points can be obtained by substituting (A7) into (A3). Similarly, their counterparts located within $\pi \leq \theta < 2\pi$, i.e., $\theta_0'$, $\theta_1'$, and $\theta_2'$, can be obtained by substituting (A7) into (A4).

The point $\theta_M$ in Fig. 10, which is located within $0 \leq \theta < \pi$, corresponds to $S_M$ in Fig. 4 and can be represented by replacing $S_0$, $S_1$, and $S_2$ in (4) with $\theta_0$, $\theta_1$, and $\theta_2$, respectively, as

$$\theta_M = \frac{-\theta_0 + 6\theta_1 + 3\theta_2}{8}.$$  

(A8)

Likewise, its counterpart, $\theta_M'$, which is located within $\pi \leq \theta < 2\pi$, can be represented as

$$\theta_M' = \frac{-\theta_0' + 6\theta_1' + 3\theta_2'}{8}.$$  

(A9)

The PWM-type pulses, shown at the bottom of Fig. 10, can be defined using the pulse amplitude function $v(\theta, \phi)$ as

$$v(\theta, \phi) = \begin{cases} V_{\text{ref}}, & 0 \leq (\theta - |\theta_{\text{ref}}|) \leq \theta_{\text{LI}} \\ V_{\text{ref}}, & \pi \leq (\theta - |\theta_{\text{LI}}|) \leq \theta_{\text{L2}} \\ 0, & \text{otherwise} \end{cases}$$

(A10)

where $|\theta_{\text{ref}}|$ denotes the nearest multiple of $\pi \leq \theta$, $|\theta|_{2\pi}$ denotes the nearest multiple of $2\pi \leq \theta$, $V_{\text{ref}}$ is the reference value, e.g., supply voltage, which will determine the amplitude of the PWM pulses and can be normalized to unity for convenience, and $\theta_{\text{LI}}$ and $\theta_{\text{L2}}$ are the pulselength functions—determined by the input signal and the linear interpolation algorithm.

1) Pulselength Functions: The pulselength functions $\theta_{\text{LI}}$ and $\theta_{\text{L2}}$ are required to solve for $v(\theta, \phi)$. These two functions locate the cross points (in the $(\theta, \phi)$ domain) that correspond to the cross points $S_{\text{LI}}$ and $S_{\text{L2}}$, respectively (in the time domain). Fig. 11 shows a detailed illustration of finding $\theta_{\text{LI}}$ and $\theta_{\text{L2}}$.

The line interpolation line between $(\theta_1, 0)$ and $(\theta_M, \pi/p)$ in Fig. 11 can be obtained as follows:

$$\phi = \frac{\pi}{p} \left( \frac{\theta - \theta_1}{\theta_M - \theta_1} \right).$$  

(A11)

The cross point $\theta_{\text{LI}}$ can be obtained when $\phi$ of (A11) equals $\phi$ of line AA' (A5), i.e.,

$$\frac{1}{p} \theta_{\text{LI}} = \frac{\pi}{p} \left( \frac{\theta_{\text{LI}} - \theta_1}{\theta_M - \theta_1} \right) \quad \Rightarrow \quad \theta_{\text{LI}} = \frac{\pi\theta_1}{\pi - \theta_M + \theta_1}.$$  

(A12)

Using similar steps, $\theta_{\text{L2}}$ can be obtained as

$$\theta_{\text{L2}} = \frac{\pi(2\theta_0' - \theta_2')}{\pi - \theta_2' + \theta_M'}.$$  

(A13)

By substituting $\theta_M$ (A8) into (A12) and $\theta_M'$ (A9) into (A13), $\theta_{\text{LI}}$ and $\theta_{\text{L2}}$ can be written in terms of the sampled points as follows:

$$\theta_{\text{LI}} = \frac{8\pi\theta_1}{8\pi + \theta_0 + 2\theta_1 - 3\theta_2}$$  

(A14)

$$\theta_{\text{L2}} = \frac{\pi(-2\theta_0' + 12\theta_1' - 2\theta_2')}{8\pi - \theta_0' + 6\theta_1' - 5\theta_2'}. $$  

(A15)

The sampled point terms in (A14) and (A15) can be substituted by the input modulating signal wall expressions in (A3) and (A4). However, the cross points or, equivalently, the pulselength functions $\theta_{\text{LI}}$ and $\theta_{\text{L2}}$ still depend on the $\phi$ values that are sampled by the line BB' in (A7). Since this line is discontinuous in the $(\theta, \phi)$ domain, due to the discrete term $\lfloor \theta/\pi \rfloor$, the double Fourier series in the $(\theta, \phi)$ domain cannot be
obtained directly. To circumvent this, (A7) is transformed into a continuous straight line using the following transform:

\[ u = \phi - \left[ \frac{\theta}{\pi} \right] \frac{\pi}{p} + \frac{\theta}{p}, \]  

(A16)

The transformation process is shown from Fig. 12(a) and (b). Note that even after the transformation, the PWM-type pulses, as shown in Fig. 12(c), remain the same. As shown in Fig. 12(b), the line BB' is now a continuous straight line in the (θ, u) domain. This line is given by

\[ u = \phi - \left[ \frac{\theta}{\pi} \right] \frac{\pi}{p} + \frac{\theta}{p}, \]  

(A17)

By substituting (A3) into (A14) and (A4) into (A15), followed by transforming the \( \phi \) terms into the (θ, u) domain, the pulsewidth functions can finally be obtained as (A18) and (A19), shown at the bottom of the page.

2) Double Fourier Series: Consider the parallelogram bounded by \( \theta = 0, \theta = 2\pi, u = (\theta/p), \) and \( u = 2\pi + (\theta/p) \) in Fig. 12(b). The PWM-type pulses (the pulse amplitude function) in (A10), which can now be written as \( v(\theta, u) \), are given by the wall configurations inside this parallelogram. Since the wall configurations are periodic in both \( \theta \) and \( u \) directions, \( v(\theta, u) \) can be expressed by means of a double Fourier series as (its complex form is used here)

\[ v(\theta, u) = \sum_{m=-\infty}^{\infty} \sum_{n=-\infty}^{\infty} K_{mn} e^{j(m\theta+nu)} \]  

(A20)

where \( K_{mn} \) is the complex Fourier coefficient given by

\[ K_{mn} = \frac{1}{4\pi^2} \int_{u=2\pi+(\theta/p)}^{u=2\pi} \int_{\theta=0}^{\theta=2\pi} F(\theta, u) e^{-j(m\theta+nu)} d\theta du, \]  

(A21)

where \( m, n \in \mathbb{Z} \).

The limits of the aforesaid integrals are the edges of the aforementioned parallelogram.

In Fig. 12(b), the height of the walls is equal to the constant \( V_{ref} \) for all values of \( \theta \) between 0 and \( \theta_{L1} \) and between \( \pi \) and \( \theta_{L2} \). It is zero for other values of \( \theta \) up to \( 2\pi \). Consequently, (A21) takes the following form when applied to this particular configuration:

\[ K_{mn} = \frac{1}{4\pi^2} \int_{u=2\pi+(\theta/p)}^{u=2\pi} \int_{\theta=0}^{\theta=2\pi} V_{ref} e^{-j(m\theta+nu)} d\theta du, \]  

(A22)

\[ \forall m, n \in \mathbb{Z}. \]

This integration is difficult to compute because of the term \( u - \theta/p \) in \( \theta_{L1} \) (A18) and \( \theta_{L2} \) (A19). To simplify the integration, the following substitution can be performed:

\[ \Phi = u - \frac{\theta}{p}, \quad 0 \leq \theta < 2\pi, \quad 0 \leq u < 2\pi. \]  

(A23)

\[ \theta_{L1} = \frac{8\pi}{16 + M \cos \left( u - \frac{\theta}{p} \right) + 2M \cos \left( u - \frac{2\pi}{p} \right) - 3M \cos \left( u - \frac{\theta}{p} + \frac{2\pi}{p} \right)}, \quad \text{for } 0 \leq \theta < \pi \]  

(A18)

\[ \theta_{L2} = \frac{\pi}{16 - M \cos \left( u - \frac{\theta}{p} \right) + 6M \cos \left( u - \frac{\theta}{p} \right) - 5M \cos \left( u - \frac{\theta}{p} + \frac{2\pi}{p} \right)}, \quad \text{for } \pi \leq \theta < 2\pi \]  

(A19)
Hence
\[ u = \Phi + \frac{\theta}{p} \text{ to yield } \frac{du}{d\Phi} = 1. \quad (A24) \]

The substitution renders the parallelogram to undergo a linear shearing transform into a square bounded by \( \theta = 0, \theta = 2\pi, \\phi = 0, \) and \( \phi = 2\pi. \) The preceding integration now takes the form of (A25), shown at the bottom of the page. If \( m + (n/p) \neq 0, \) (A26), shown at the bottom of the page, is derived, where
\[
\int_{0}^{2\pi} e^{-jn\phi} d\phi = \begin{cases} 0, & n \neq 0 \cr 2\pi, & n = 0 \end{cases}. \quad (A27)
\]

The evaluation of the integration can be considered for several cases, depending on the integer values of \( m \) and \( n. \) These cases will be discussed in the following.

Case 1: \( m = 0 \) and \( n = 0. \) Since \( m\theta = n\omega_{in}t = 0 \) and \( n\phi = n\omega_{in}t = 0, \) i.e., dc, this case therefore corresponds to \( K_{00} \) of the dc component. \( K_{00} \) can be derived from (A25) as (A28), shown at the bottom of the page.

Case 2: \( m + (n/p) \neq 0, m = 0, \) and \( n \neq 0. \) This case corresponds to \( K_{0n} \) of the input modulating signal and its harmonics. From (A26)
\[
K_{0n} = K_{mn}|_{m=0}. \quad (A29)
\]

Case 3: \( m \neq 0 \) and \( n = 0. \) This case corresponds to \( K_{m0} \) of the carrier signal and its harmonics. From (A26)
\[
K_{m0} = K_{mn}|_{n=0}. \quad (A30)
\]

Case 4: \( m + (n/p) \neq 0, m \neq 0, \) and \( n \neq 0. \) This case corresponds to \( K_{mn} \) of the intermodulation components arising from the input modulating signal and carrier signal harmonics. \( K_{mn} \) is given by (A26).

To obtain the double-Fourier-series expression for the PWM-type pulses in the time domain, \( \theta \) and \( u \) have to be related to time. From (A2), \( \theta = \omega_{in}t, \) and from (A17) and (A6)
\[ u = \frac{\theta}{p} = \omega_{in}t. \quad (A31) \]
Hence, by substituting (A2) and (A31) into (A20)
\[ v(\theta, u) = v(\omega_1 t, \omega_2 n) = v(t). \] (A32)

The coefficients in (A26), (A28), (A29), and (A30) are inserted to the series in (A20), and thereafter, following the substitutions shown in (A32), the double Fourier series for the PWM-inserted to the series in (A20), and henceforth, following the substitutions shown in (A32), the double Fourier series for the PWM-type pulses generated by the single-sided trailing-edge LAG1 of a sinusoidal input modulating waveform can finally be obtained as
\[ v_{\text{LAG1}}(t) = K_{00} + \sum_{n=\pm 1}^{\infty} \left( K_{\text{On}} e^{j(n \omega_1 t)} \right) + \sum_{m=\pm 1}^{\infty} \sum_{n=\pm 1}^{\infty} \left( K_{\text{On}} e^{j(n \omega_2 t + n \omega_2 n \theta)} \right). \] (A33)

REFERENCES


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