An Adaptive Noise Tolerance Method of using Sigma-Delta Modulation

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Abstract

In this paper, we developed a noise tolerance approach by using sigma-delta modulation, which can recover the noisy signals. Beside most of the previously schemes based on solving Single Event Upset (SEU), our approach can cope with the case of multiple faults, even the noise has been continued for a long time. The other property of our method is that no need to modify the original circuit and thus the performance of original circuit itself can be maintained. Meanwhile, an optimal Over Sampling Rate (OSR) has been proposed to obtain a minimum deviation between original and restore signals. A new adaptive gain skill has been introduced to compensate the signal distortion due to different signal frequency, OSR and noise.

In this experiments, we have assumed that a circuit output signal is affected by noises, and the total noise energy changing from 10% to 100% of the original signal has been provided to demonstrate the effectiveness for this research. A fixed gain and adaptive gain methods have been evaluated and the results show that the deviation rate is more improved.

Keyword：Noise, Soft Error, Over Sampling Rate, SEU, DSP, Sigma-Delta

1. Introduction

In the past, the designer does not concern too much about the problem of noise because the noise will not present much hazard for the circuit. However, the density of circuit is becoming higher than before due to the process of VLSI technology has been much improved so far. In other words, many new fault models are found and defined for the more compact and density structure even if the noise is the same. If we do not consider to coping with the new faults or noises, then the system might have much failure probability and thus reduce the reliability dramatically. In addition, the impact of noise might have more than the effects of fault model itself since the fault model is usually predicable or definite. On the contrast, the noise has the property of uncertain and even has the format of random to show up. Thus, how to solve the problems of noise is becoming more and more important issue in the current field of circuit design.

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In general, noise is a random signal with no characteristic of regular period, so usually it can be represented as the form of average power. It has two major sources, one is from the environment or input signal and another is from the circuit itself. There are many kinds of noises, and the most that have been considered in VLSI design are Shot Noise, Thermal Noise, Flicker Noise, Burst Noise and Avalanche Noise [1].

To solve the noise influences, many researches have been presented to increase circuit reliability and noise tolerance. The thermal noise and flicker noise are produced from circuit itself, so they can be reduced in circuit design like decreasing circuit equivalent resistance, increasing transistor’s width and so on. But for the noises not from circuit itself, such as shot noise, it’s not easy to estimate and reduce, thus it must use some additional methods to cut down its influence. The common and effective methods are Triple Modular Redundancy (TMR)[2][3], Algorithmic Noise-Tolerance (ANT)[4], and the method of using modulated signals to enhance the ability of noise tolerance [5][6]. In these methods, they are mainly to solve the circuit that has been interfered by the Soft Error or Single Event Upset (SEU). Nevertheless, when the noise ratio is too high and sustain for a period of time, these methods are ceased to be effective and most of them need to modify the circuit of original circuit. In addition, the original signal strength might be distortion because it is possibly interfered by the various noises, the variation of input signal frequency and the different over sampling rate. Generally, this kind of distortion has much influence on the quality of signal. Thus, a new dynamic recovery method has to be proposed to cope with such an issue.

In this paper, we propose a fixed and adaptive gain noise tolerance approach by using sigma-delta modulation, which can recover the noisy signals dynamically. Besides most of the previously schemes, our approach can solve the case of multiple faults, even the noise has been continued for a long time. Meanwhile, there is no need to change the original circuit itself by using our proposed method, and thus the performance of original circuit itself can be maintained.

In the Section 2, we will discuss the related researches and methods of improving noise tolerance. The Section 3 is about the design and theoretic derivation regarding our proposed work. Then, we will discuss our simulation and experimental results in Section 4. The conclusions are presented in Section 5.

2. RELATED WORKS

2.1. Algorithmic Noise-Tolerance (ANT)

To cope with all kinds of noise interference issues, there are some researchers have proposed several papers to discuss and solve this problem [4][6-10]. The motivation is to reduce power of the traditional methods for noise tolerance [4]. The Triple Modular Redundancy (TMR) [2] adopts three copies of original circuit, and distribute in different locations to disperse the risk of noise influence.

For the different type of estimator, ANT can be mainly divided into Prediction-Based ANT and Reduced-Precision Redundancy (RPR)-Based ANT [7]. A general structure for the ANT circuit is shown in the Fig. 2.1[9]. By applying the same input signal to the original circuit (i.e. M BLOCK) and estimator, the estimator will generate a computing signal as a reference output to compare with the output from the 'M BLOCK'. After that, the reference output will be used to subtract with the output signal from the 'M BLOCK'. Let the subtract result value be T, if |T| is bigger than the pre-computing threshold value, then the signal is influenced by the noise. Otherwise, it is normal and no noise impact. Therefore, it uses a multiplexer to determine unaffected signal ya[n] or estimated signal ye[n] as the output signal.

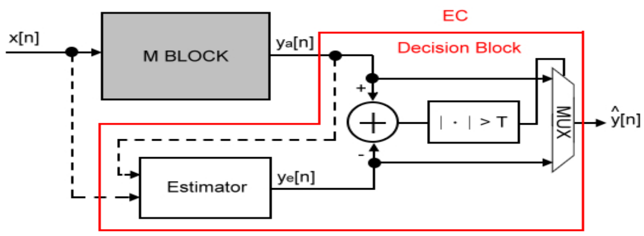


Fig. 2.1 Algorithmic Noise-Tolerance (ANT)

It has to assume the estimator working with no errors in this research. However, in a real circuit, the estimator is not working in this condition, and there still have many noises to be faced in the actual design. Therefore, although ANT can produce more effective signals, but it can't work while considering the estimator is influenced by the noise.

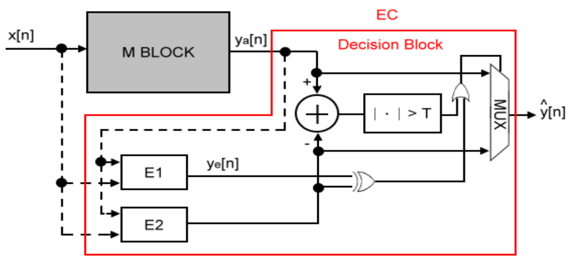


Fig. 2.2 S-ASET ANT

To solving this problem, another method naming Spatial Algorithmic Soft Error Tolerance (S-ASET) has been proposed [9]. This method is used to improve the original ANT to reduce the noise effect for the estimator by duplicating the estimator to increase its fault tolerant capability. It is shown in the Fig. 2.2 and the estimator is duplicated to disperse the risks of noise impacts. In addition to the M block, there are two estimators E1 and E2 in the EC block. In essence, S-ASET is the spatial diversity scheme based on block estimator E1 and E2. Nevertheless, when the noise ratio is too high and sustain a period of time, these methods are still to be ineffective because the E1 and E2 are influenced by the noise.

2.2 Sigma-Delta Modulated Signals

In addition to ANT, another effective method is to use modulated signals. It adopts Sigma-Delta modulated signals to replace the original signals. This method can cope with not only SEU problems but also multiple upset.

In [5][6], they have improved the ability of noise tolerance in DSP to modify the original circuits by using Sigma-Delta modulated signals. To compute the bit stream signals, they provide a set of modified adders and multipliers to have the capability of Sigma-Delta modulation. A new DSP circuit with Sigma-Delta modulated signals has been proposed but keeps the same functions as original unmodified DSP circuit except the high noise tolerant capability [11]. However, these designs will have some influences to degrading the performance. In [12], a noise tolerance with lower hardware overhead had been proposed by using Sigma-Delta modulation skill. This method doesn't ensure to obtain a minimum deviation between original and restore signals.

3. DESIGN AND IMPLEMENTATION

By reviewing the researches in the Section 2, we know that the better way to solve the external noise interference in circuit has three conditions to be considered. Firstly, the original circuit should not be modified as possible. Secondly, the new added noise tolerant circuit should irrelevant with the size of original circuit. Thirdly, the new noise tolerant should cope with the noise occurred for a period of time. Finally, a compensate method to restore the distortion signal by using variable gain while considering different input signal frequency, over sampling rate and noise. To achieve these goals, we proposed a new noise tolerant method that is based on Sigma-Delta modulation skill as shown in the Fig. 3.1. The Self Noise Tolerance Estimator is used to recover the output signal while the output signal is interfered by the noise. The difference between ya[n] and ye[n] is to indicate the degree of noise level. If the difference is larger than the threshold (T), the signal ye[n] is selected as the new output.

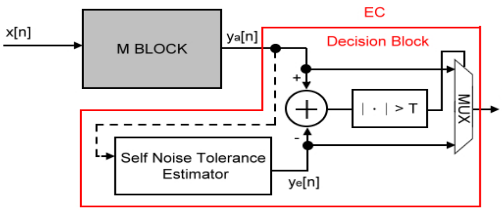


Fig. 3.1 Self Noise Tolerant Esimator

In general, Single Event Upset or Single Event Transient in original system is a great variation in a very short period of time so that its frequency response is in the high frequency spectrum. This noise can be easily reduced by the output filter or the skill of noise shaping. However, this might not be feasible for a continuous noise because the noise energy sustain for a period of time. In other words, the system operation might be stopped during this period due to noise interference. Thus, a new effective idea of reducing noise has to be proposed. The main concept to reduce the interference effects of noise is that to spread the noise into many sub-band to suppress its strength of energy while considering the noise always exists. Based on this concept, the over

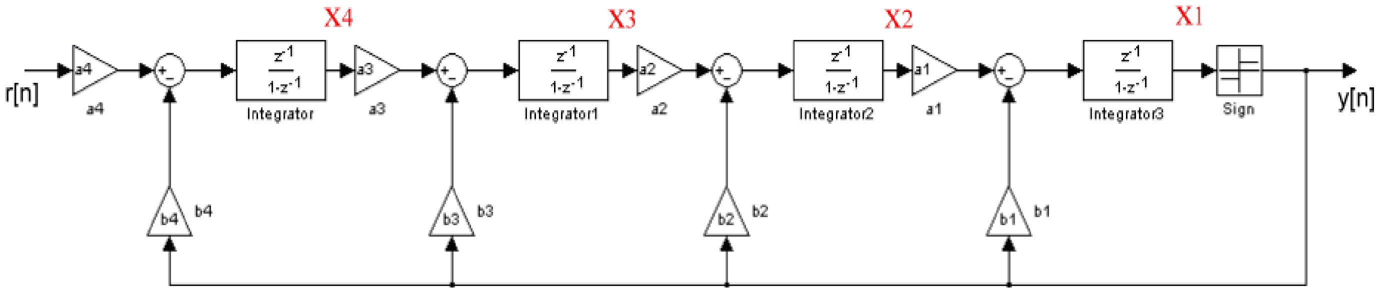


Fig. 3.2 Fourth-Order 1-Bit Sigma-Delta Modulator

sampling skill can be used to sampling the noise and signal at the same time such that the noise is effective only at the sampling point. Hence, the noise is reduced due to only the noise occurred at the sampling point is effective; otherwise, it can be ignore. Thus, a Sigma-Delta modulation skill with high SNR has been applied here to solve such an issue.

In [13], the direct way to enhance SNR is to increase its over sampling rate or its order for the Sigma-Delta modulation. It can be detected that SNR increases slowly while the order is over four. Therefore, we propose a Sigma-Delta modulator with fourth-order, and import the method of using weights to cancel the gain of DAC on the feedback path [14]. The fourth order structure is shown in Fig. 3.2 and its equivalent formula can be obtained as:

$$\mathrm{sgn}[\{[[r(z)(a_4-b_4)(1/(z-1))a_3-b_3](1/(z-1))a_2-b_2](1/(z-1))a_1-b_1\}(1/(z-1))]=y[z]$$

To help us for analyzing easily, we transfer the above formula into a clear and simple form and its simple form can be shown in the Fig. 3.3. Meanwhile, its equivalent Z transform formula is also shown as:

$$X_1(z)=W(z)[F(z)R(z)-Y(z)]$$

, where the F(Z) and W(Z) are summarized as

$$F(z)=\frac{a_1a_2a_3a_4}{N(z)} \quad (1)$$

$$W(z)=\frac{N(z)}{(z-1)^4} \quad (2)$$

$$N(z)=b_1(z-1)^3+b_2a_1(z-1)^2+b_3a_1a_2(z-1)+b_4a_1a_2a_3 \quad (3)$$

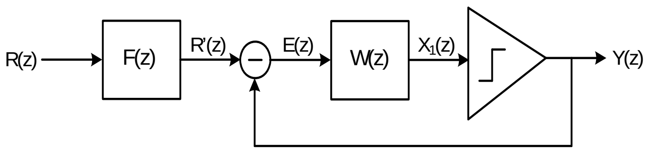


Fig. 3.3 Sigma-Delta Equivalent Model

Here, we can see that F(z) is generated by multiplying the coefficient (i.e. gain) along with the feedback path and is denoted as the Signal Transfer Function (STF), W(z) is the Noise Transfer Function (NTF) in the Sigma-Delta modulator. The variation of the state variable X1 to X4 has to be less than 0 to keep convergence. This implies that the ΔX (i.e. variation of x) is less than 0. For example in the first order of Fig. 3.2, the Sigma-Delta formula is obtained as

X1(k+1)=X1(k)-b1Y(k)+a1X2(k)
,where *Y(k)=sgn[X1(k)]*.

Next, the variation of X1 is obtained as

Δ*X1=X1(k+1)-X1(k)=-b1Y(k)+a1X2(k)*
=-b1sgn[X1]+a1X2(k)

,where the $\Delta X1$ is less than 0 with respect to the change of time. Clearly, the $X1\,\Delta X1$ is less than 0 when the $X1$ is not equal to 0. If $X1$ is large than 0, then the $\Delta X1$ is less than 0. That means the *X1(k+1)<X1(k).* Similarly, if $X1$ is less than 0, then the $\Delta X1$ is large than 0. That is the *X1(k+1)>X1(k).* Both cases show that the change of *X1* is toward to the zero point from the positive axis and negative axis sides. This is satisfy the convergence condition.

Because Δ*X1= -b1sgn[X1(k)]+a1X2(k)* should be less than 0, it induces that the |X2| should be less than $\frac{b_1}{a_1}$ as well. To evaluate the stability in the design of Sigma-Delta modulation, there are two system stability rules to be followed in this design:

1). Let the zero points of W(z) locate inside unit circle.

2).Let system state x_2 satisfy the condition of $|X_2| < \frac{b_1}{a_1}$.

According to the equivalent model and stability rules, there are several steps to design a multi-order Sigma-Delta modulator:

- Design NTF by referring input signal
- Derive W(Loop Filter) from NTF
- Derive F(Pre Filter) from W and state formulas
- Compute the coefficients
- Design output filter to restore the signal

Here, the above five steps are all exercised in the Matlab. First, we assume the input is a 44.1 KHz audio signal, and it needs to be modulated to the sampling rate of 1.5 MHz. Therefore, the over sampling rate can be determined as 34. After that, the NTF can be designed while the condition of over sampling rate has been decided. The NTF can be proposed in Filter Design Toolbox of Matlab and we select the Butterworth type as our filter. To reduce the noise effects, we set the cutoff frequency at 60 KHz, which is much higher than 20 KHz of audio range and the toolbox returned its function as

$$NTF = \frac{0.7193 - 2.8771z^{-1} + 4.3156z^{-2} - 2.8771z^{-3} + 0.7193z^{-4}}{1 - 3.3448z^{-1} + 4.2382z^{-2} - 2.4088z^{-3} + 0.5173z^{-4}}$$

Then, we modify the numerator of NTF in terms of (Z-1)^4. The result will be shown as

$$NTF = \frac{z^4 - 4z^3 + 6z^2 - 4z^1 + 1}{z^4 - 3.3448z^3 + 4.2382z^2 - 2.4088z^1 + 0.5173} \quad (4)$$

$$= \frac{(z-1)^4}{z^4 - 3.3448z^3 + 4.2382z^2 - 2.4088z^1 + 0.5173} \quad (5)$$

In [15], we know that the relation between W(z) and NTF. Thus, the W(z) can be represented as formula 3-6 shows for the Fig. 3.3:

$$W(z) = \frac{1 - NTF}{NTF} = \frac{1 - \frac{(z-1)^4}{z^4 - 3.3448z^3 + 4.2382z^2 - 2.4088z^1 + 0.5173}}{\frac{(z-1)^4}{z^4 - 3.3448z^3 + 4.2382z^2 - 2.4088z^1 + 0.5173}} = \frac{0.6562z^3 - 1.7618z^2 + 1.5912z^1 - 0.4827}{(z-1)^4} \quad (6)$$

According to the formula 3-2, the *N*(z) is equal to $0.6562z^3 - 1.7618z^2 + 1.5912z^1 - 0.4827$.
Similarly, the F(z) can be shown as

$$F(z) = \frac{a_4 a_3 a_2 a_1}{0.6562z^3 - 1.7618z^2 + 1.5912z^1 - 0.4827}$$

from the formula (1).

By solving these equations, we obtain three roots of zero as 0.915+0.152i, 0.855 and 0.915-0.152i. These three roots are all located within unit circle. Thus, the derived W(Z) satisfies the stable condition.

Next, we should find out the system coefficients in the F(z). Here, in the formula (2), we know that the denominator of W(z) is N(z) and the N(z) is $b_1(z-1)^3 + b_2 a_1 (z-1)^2 + b_3 a_1 a_2 (z-1) + b_4 a_1 a_2 a_3$. To find the value of b_1, we transfer the formula (4) to another form in terms of z-1 as formula (5) shows. After that, the W(z) is obtained as formula (6). Thus, the b_1 is equal to 0.6562 after comparing formulas (2), (3) and (6). In general, the system stability depends on the coefficient of a_1, a_2, a_3 and a_4 no matter what the value of b_1, b_2, b_3 and b_4 are. The a_1, a_2, a_3 and a_4 have to satisfy the condition as inequality formula (7) to (9)shows [15], where the || ||ss∞ denotes the maximum stable value. Clearly, the ||x2||ss∞, ||x3||ss∞ and ||x4||ss∞ should all less than or equal to one for stability. Thus, we can find the a_1, a_2, a_3 and a_4 while considering the ||r||ss∞ and ||x1||ss∞ are less than or equal to one. Here, we assume the a_1 = 0.9* b_1. After that, the $a_2a_3a_4$ can be obtained from the formula (7). Next, the a_2 can be got from the formula (8). Finally, we get the a_3 from the formula (9). After getting the $a_2a_3a_4$, a_2 and a_3, the a_4 can be obtained. Because the range regarding a_2, a_3 and a_4 has been obtained, we are able to assume that the a_2 and a_3 is equal to 0.3 and 0.2, respectively, to satisfy the stability condition as formula (7) to (9) shows. After that, the b_2, b_3 and b_4 can be found by comparing the formulas (2),(3) and (6) as Tab.

3-1 shows.

$$\|x_2\|_{ss\infty} \approx \frac{223.5a_1a_2a_3a_4}{a_1}\|r\|_{ss\infty} + \frac{0.4241}{a_1}\|x_1\|_{ss\infty} \le 1 \qquad (7)$$

$$\|x_3\|_{ss\infty} \approx \frac{70.4a_1a_2a_3a_4}{a_1a_2}\|r\|_{ss\infty} + \frac{0.069}{a_1a_2}\|x_1\|_{ss\infty} \le 1 \qquad (8)$$

$$\|x_4\|_{ss\infty} \approx \frac{12.3a_1a_2a_3a_4}{a_1a_2a_3}\|r\|_{ss\infty} + \frac{0.0045}{a_1a_2a_3}\|x_1\|_{ss\infty} \le 1 \qquad (9)$$

As a result, we can get the system coefficients a_1~ a_4 and b_1~ b_4 and take these back into function F(z) to get F.

Tab.3-1 System Coefficients of Fourth-Order SDM

$a_1 = 0.5906$	$a_2 = 0.3$	$a_3 = 0.2$	$a_4 = 0.0299$
$b_1 = 0.6562$	$b_2 = 0.3502$	$b_3 = 0.2043$	$b_4 = 0.0818$

If we take these coefficients into Fig. 3.2, then we can get the overall structure of fourth-order Sigma-Delta modulator. Because the modulated output is a 1-bit signal with over sampling rate and thus the quantization noise can be transferred to high frequency band, we propose a low-pass fourth-order Butterworth filter as its output filter and set its cutoff frequency as 19 KHz. It can be proposed in Filter Design Toolbox. The Toolbox returns its function (i.e. H(z)) as follow:

$$H(z) = \frac{B(z)}{A(z)} = \frac{10^{-4}\left(0.0227 + 0.0907z^{-1} + 0.1361z^{-2} + 0.0907z^{-3} + 0.0227z^{-1}\right)}{1 - 3.7920z^{-1} + 5.3973z^{-2} - 3.4174z^{-3} + 0.8121z^{-4}}$$

Combining all the previous models with all related components, and then we have completed the all system anti-noise architecture as shown in Fig. 3.5.

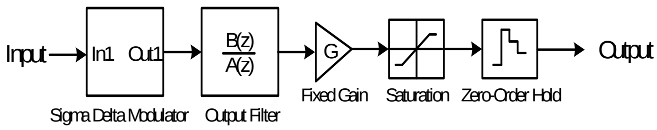


Fig. 3.5 Anti-Noise System Architecture (Self Noise Tolerance Estimator)

In Fig. 3.5, we know that the Sigma-Delta Modulator is used to get ride of the noise in the influenced signal. The output signal from here is a modulated square wave varied between -1 to +1. For a Sigma-Delta modulator, the high frequency noise in the modulator will be shifted to different high frequency band according to different over sampling rate. Thus, the signal itself can be obtained by using a low-pass filter to get rid of the high frequency noise. In general, if the signal frequency is higher, the noise energy is higher. That means the higher frequency signal may induce more noise since the all signals are all encoded as the same digital signals to store. In this research, our input signal is various from 0~20 KHz and they are all encoded as digital signals with 44.1 KHz. A signal with higher frequency will induce more noise than the signal with lower frequency while encoding these signals into the signals with same frequency band.

The output filter is applied here to remove the noise and to reshaping the square wave signal to become a sine wave signal. The amplifier *G* is to enlarge the modulated signal to its original strength. However, if the gain of amplifier is to keep in a constant, the signal strength might have some distortions because different over sampling rate might be required to obtain a minimum distortion signals in corresponding to the signal with different frequency. In here, we also see that the signal strength will be changed with respect to a different over sampling rate while assuming the input signal with a fix frequency. In other words, an optimal over sampling rate must be selected to obtain the minimum distortion. In a real case, the audio input signal is various within a period of frequency band. The over sampling rate need to change to ensure the minimum signal distortion when the input signal frequency is changed. However, the over sampling rate has to be fixed, otherwise the functionality of Sigma-Delta modulator might not be work normally. Thus, we let the over sampling rate be a fix value. Because of this, the only way to make up the distortion is to change the gain.

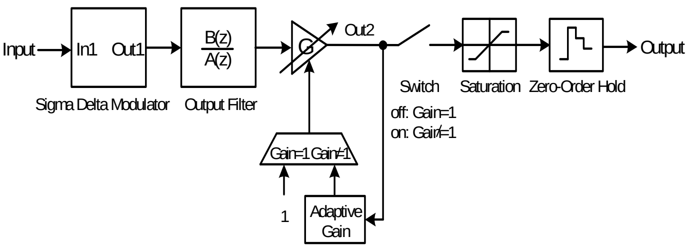


Fig. 3.6 Anti-Noise architecture with an adaptive gain

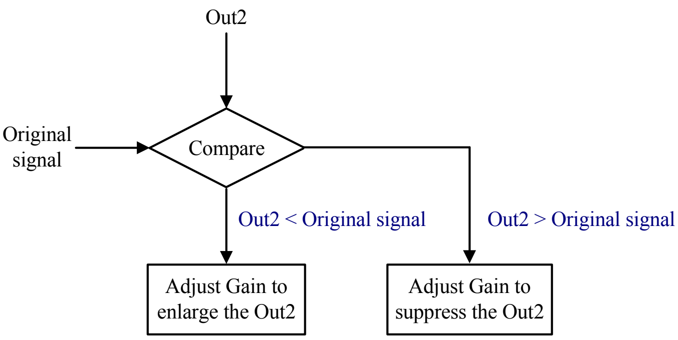


Fig. 3.7 A new self-inverse unit-gain method

In our research, we have proposed another Anti-Noise architecture with an adaptive gain as shown in the Fig 3.6 . In this architecture, the gain is various with a different frequency input signal. To simplify the design of adaptive gain, we have presented a new self-inverse unit-gain method. In this method, we first let the gain to be 1 to obtain a distortion signal, and then we compare the output signal with the original input signal to get the gain to restore the distortion signal. However, the distortion signal may higher or lower than the original input signal. Thus, we propose an adaptive gain with the function as Fig. 3.7 shows. In this figure, if the output signal of amplifier is larger than the original input signal, then we select the lower gain to suppress the signal. Otherwise, if the output signal of amplifier is lower than the original input signal, then we choose the gain to enlarge the signal. For example, we assume the input signal for the Sigma-Delta modulator is a sine wave with 10K Hz. If we fix the amplifier gain with 1 and over sampling rate with 25, the output signal strength from the filter output will decrease from +1 ~ -1 to +0.36 ~ -0.36. In order to compensate this distortions, we have to change the gain to 2.75 such that the filter output signal can be restored to +1 ~ -1. The 2.75 can be used as the gain in this architecture.

Realistically to say, the output signals will not ideal change from -1 to +1 because of noise effects. Hence, the saturation is to limit the signal between -1 and 1. Finally, the Zero-Order Hold block is used to restore the over sampling signal to its original one. This architecture has the advantages of using oversampling skill to deal with the continuous noise without modifying the original circuit itself. This self- noise tolerance estimator can be built as an intellectual property to insert into any node in circuit. The capability of noise tolerant for this system will be demonstrated in the next section.

To give more systematic analysis, we first assume that a signal can be divided into many periods that are from period 1 to n. We suppose that the mi denotes the signal including noise for the period i so that the mi = si + ni, where si and ni represent the signal and noise within the period i, respectively. The duration for period is dependent on the trade-off of overhead. Here, we also use the M denotes the summation for signals including noise from period 1 to n so that the output signal $M = \sum_{i=1}^{n} m_i$. Clearly, the signal mi might have some deviations with the signal si due to noise ni. To restore the signal, we have to select a gain to let the gain(i)*mi = si, where gain(i) is used to indicate the gain adopted in the period i to restore the distortion signal. If we let the period shorter, then the restore effect is much better. However, this kind of design induces more overheads. Hence, we select the period that is equal to a cycle period of the input signal to reduce the overhead. In our research, the period selection is from 1KHz to 19KHz based on the input signal cycle. However, if it is to face signals with non-cycle, the period selection is dependent on the consideration of overhead. In this research, we have applied the period selected from the cycle signal to some nonlinear signals and the results are demonstrated confidence as well.

4. Simulation and Results

In this research, we adopt Matlab as our main design tool and propose the model with simulation by Simulink. We also consider the noise effects in our system to simulate the realistic system behavior. To demonstrate our proposed work can be applied in a real environment, we adopt SD Toolbox in Matlab to modify our structure by considering non-idealities conditions and self noises. The revised structure for our proposed Anti-Noise system is shown in Fig. 4.1. There are white noise and thermal noise have been considered in our proposed work. For the part of Sigma-Delta modulator, the ideal integrator can be replaced by using a non-ideal integrator as shown in Fig.

4.2. Adding the white noise model for each amplifier input and the thermal noise for the input of Sigma-Delta modulator can simulate the non-ideal integrator. We

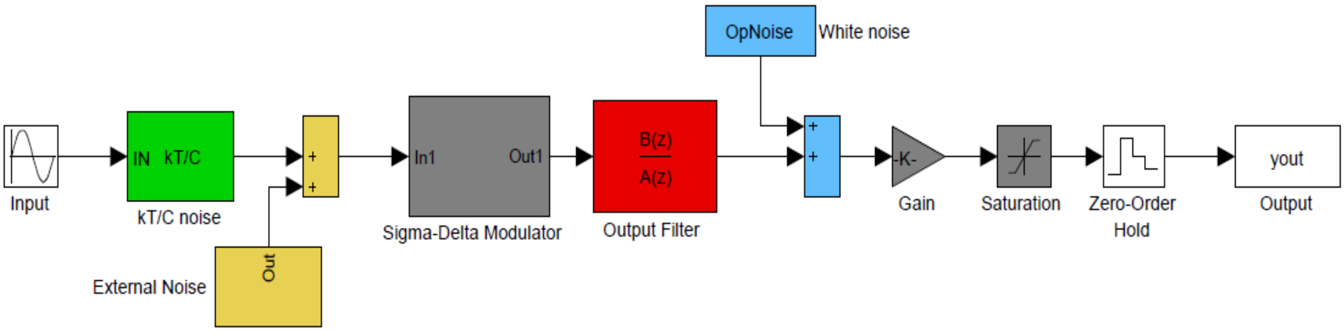


Fig. 4.1 Model with Non-Idealities

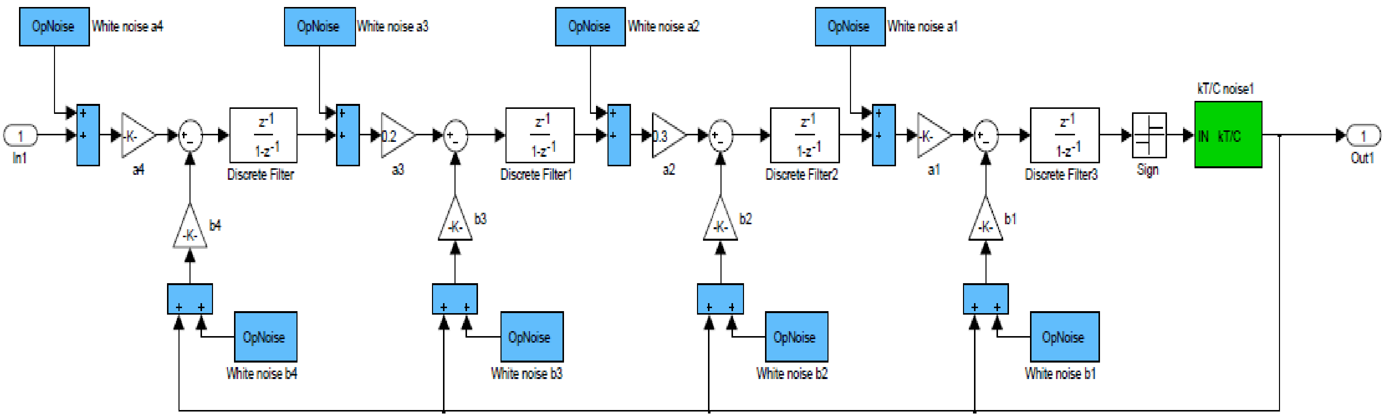


Fig. 4.2 Sigma-Delta Modulator with Non-Idealities

exploit the simulations with this modified model to be close to the realistic environment behavior as possible. After considering the noise from the circuit itself, we also need to concern the noise from the environment. To evaluate the noise effects, we additionally add an external noise source to simulate the environment noise.

To evaluate the experiment, we choose sine wave from 1KHz to 19KHz and two audio files as our input signals. The two audio files are provided from [16]. The welcome.wav is a mono signal with 22KHZ and the Space.wav is a stereo signal with 44KHZ for each of the left and right channels. Meanwhile, we also select the OSR as 26 that is identical to the average OSR for the signal from 1K Hz to 19K Hz to ensure minimum deviation as shown in the Fig. 4.9. To demonstrate the continuous noise tolerant capability in our work, we provide a continuous noise signal with 50% energy scale for the two audio files to evaluate the interference effects of noise. For the fix amplifier gain case, the three different signals are as shown in Fig. 4.3 (a) and (b) for the audio signals “welcome” and “space”.

The signals ‘1’, ‘2’ and ‘3’ denote the original signal, interfered signal and restored signal, respectively. In the red borders of Fig. 4.3, we can see if the period of signal has very few information contained, then the information can be traced by the method of Sigma-Delta is also very few or even almost near to zero. In other words, the anti-noise capability is not good when the signal strength is very low. Thus, there still exist some obvious noises in the restored signal for the row ‘3’. On the contrast, if the period of information has strong potential, then the Sigma-Delta can trace its shape of amplitude more efficient and the restore signal capability can be much more improved.

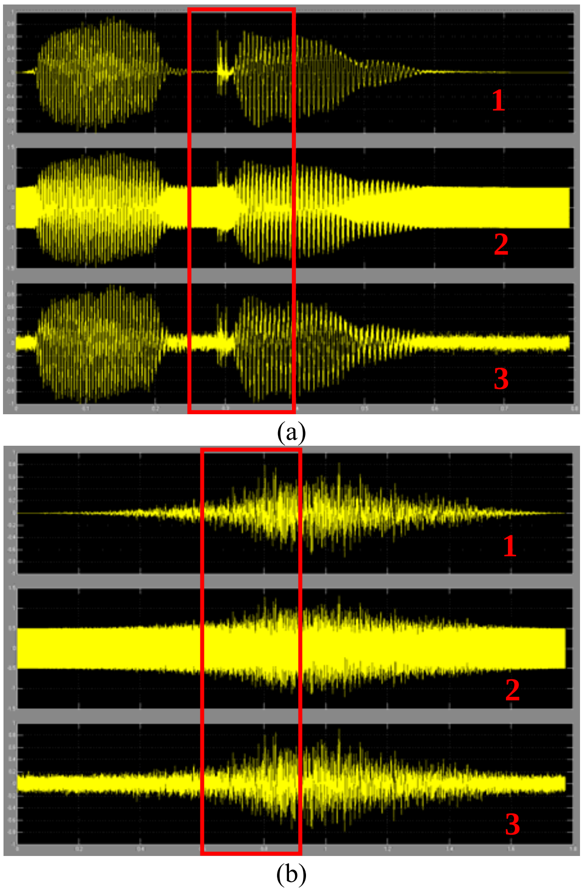


Fig. 4.3 the audio signals "welcome" and "space", (a)Waveform in Each Stage of welcome.wav, (b) Waveform in Each Stage of space.wav.

To evaluate the deviation of noise strength, we exploit the noise strength from 10 % to 100% by the step of 10% for the four test files. Firstly, the maximum deviation can be obtained by selecting the worst case while adding the noise with different strength to the test files. For a fixed gain case, the Fig. 4.4 show the related result for the maximum deviation. In Fig. 4.4, it shows that maximum deviation is proportion to its noise power. In other words, the deviation is become greater while the noise power is increasing. It also shows the maximum deviation ratios in each example are all less than their impacted noise power ratios because of the effectiveness of propose noise tolerant technique. In addition, we see that the deviation in most each case is less than the reference case of 1KHz sine wave since its average signal strength is less than others. Similarly, the results of average deviations have the same condition, as shown in Fig. 4.5. As for the adaptive gain case, the Fig. 4.6 and 4.7 show the results for the maximum deviation and average deviation respectively. The maximum deviation and average are proportional to the noise power as well. By comparing with the Fig. 4.4, 4.5, 4.6 and 4.7, we clearly see that the adaptive gain is better than the case of fixed gain.

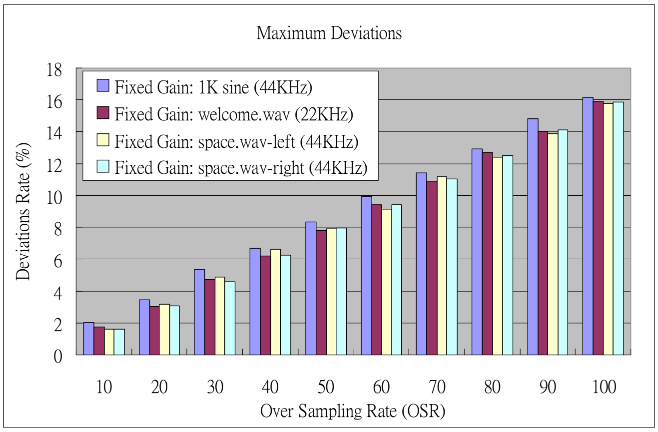


Fig. 4.4 Fixed Gain Case for Maximum Deviations between Restore Signal and Original Signal

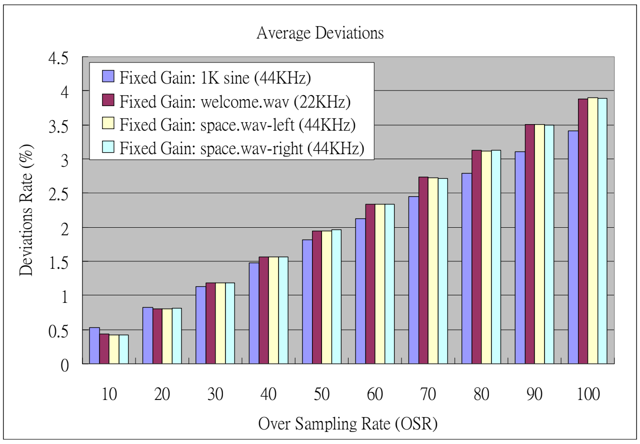


Fig. 4.5 Fixed Gain Case for Average Deviations between Restore Signal and Original Signal

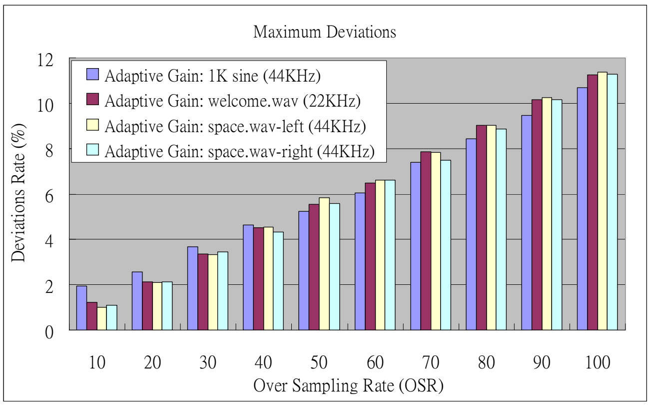


Fig. 4.6 Adaptive Gain Case for Maximum Deviations between Restore Signal and Original Signal

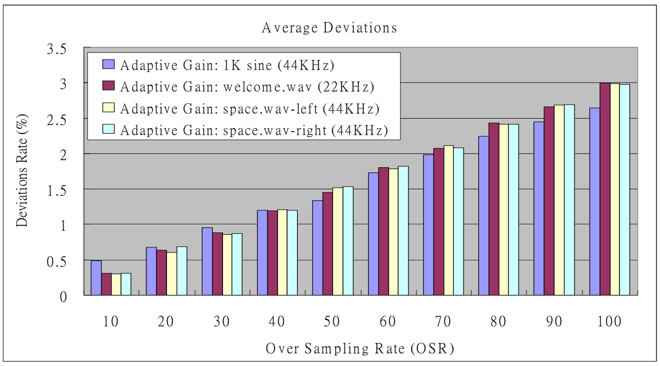


Fig. 4.7 Adaptive Gain Case for Average Deviations between Restore Signal and Original Signal

If we choose OSR (Over Sampling Rate) as 32(blue) and 64(red) in our system, then we can get their frequency response simulation as shown in Fig. 4.8. Here, we know if the over sampling rate is higher, then its frequency response curve will be shifted to high frequency band area. Otherwise, if the over sampling rate is lower, then the response curve will be shifted to the low frequency band area. For this reason, we should enlarge the gap between the high frequency band area and the signal itself band to prevent the overlap between the original signal itself and noise.

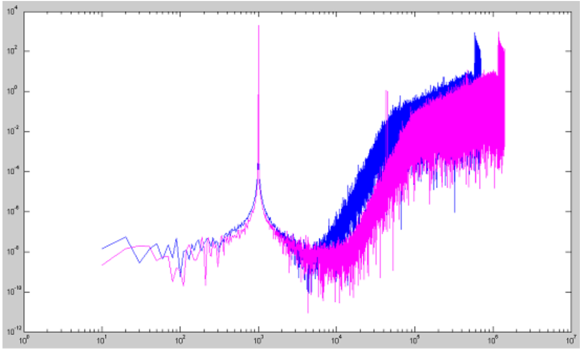


Fig. 4.8 Frequency Response of OSR=32 and

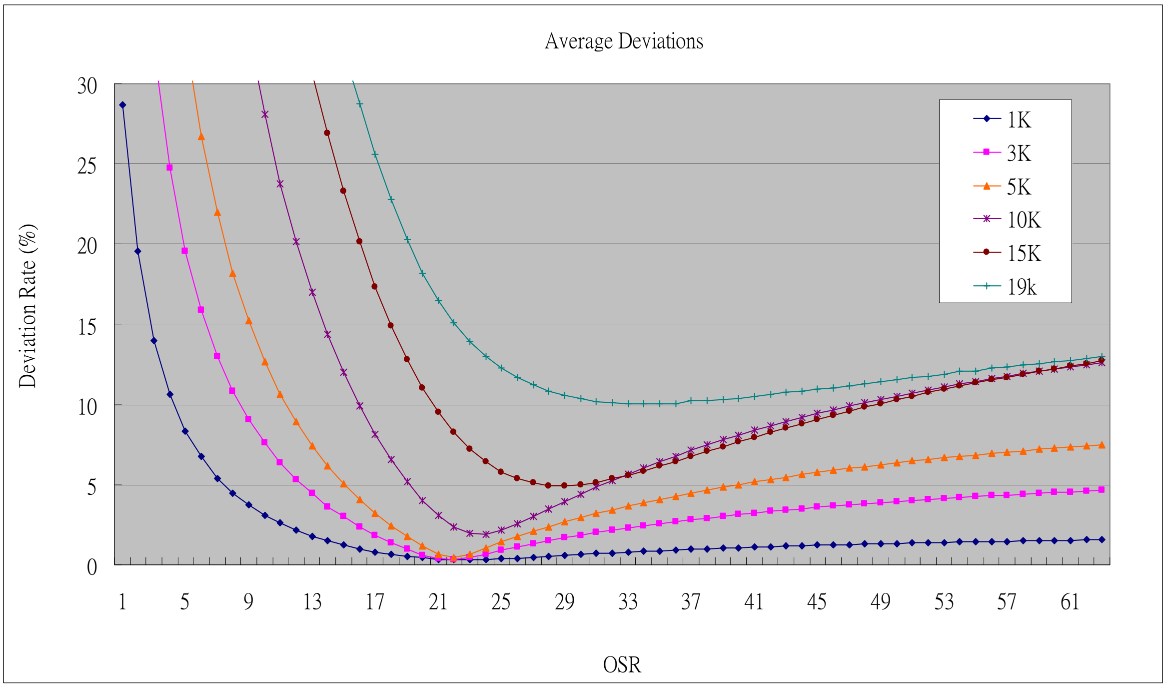
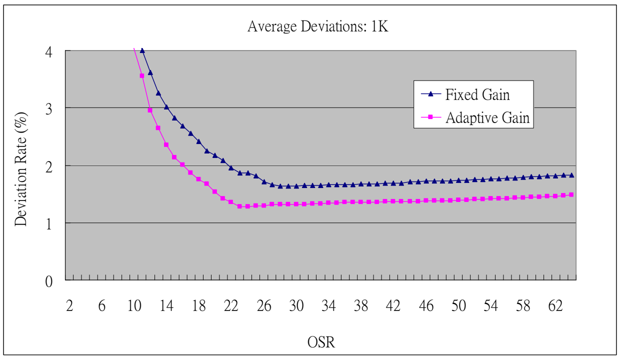
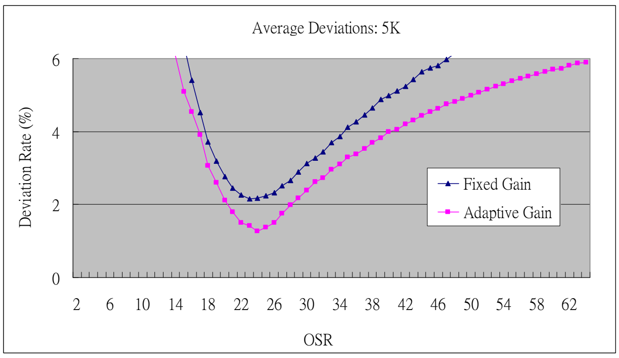


Fig. 4.9 Relationship between OSR and Average Deviations of signal from1 KHz to 19 KHz sine wave

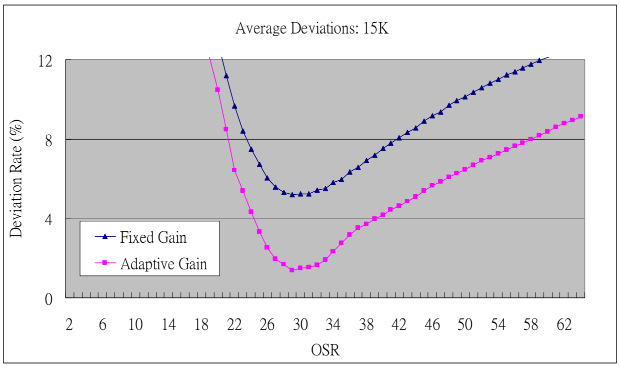
To achieve this goal, we have to find a suitable OSR that is not only to enlarge the gap between the signal itself band and high frequency band area, but also to ensure that a minimum deviation between the original and restore signals. In general, the OSR is not proportional to its signal deviation. In other words, the higher OSR can not ensure that a lower signal deviation is obtained; similarly, the lower OSR will also can not ensure that a higher signal deviation. Thus, there might have an optimal OSR to get the lowest deviation as well as to enlarge the band gap. To exploit this idea, we adopt signal from 1 KHz to 19 KHz sine wave as input signal accordingly and the simulation results between the deviation ratio and over sampling rate can be shown in the Fig. 4.9. We can see its best OSR is 24, 23, 23, 25, 30, and 35 for the 1 KHz , 3KHz, 5KHz, 10 KHz, 15KHz and 19KHz sine wave, respectively. For simplicity, we only illustrate the 1KHz, 5KHz and 15KHz as an example to show the recovery improve rate in the Fig. 4.10 (a), (b) and (c). The results show that deviation rate for the adaptive gain method is lower than the fixed gain method.



(a)



(b)



(c)

Fig. 4.10 Relationship between OSR and Average Deviations, (a)1KHz, (b)5KHZ, (c)15KHZ.

Similarly, let's do the same simulation for different frequency signals from 1 KHz to 22 KHz with the step of 1 KHz, and then we can find out each best OSR as shown in Fig. 4.11. It shows the higher frequency signal needs higher OSR to get its best case of noise tolerance.

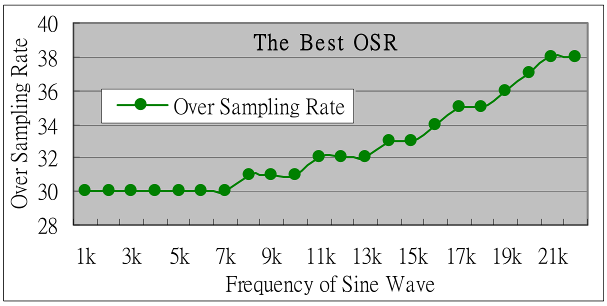


Fig. 4.11 The Best OSR of Different Frequency Sine Wave

5. Conclusions

In this paper, we propose a Sigma-Delta based noise tolerance approach, which has three properties as following list:

- It has no need to modify the original circuit and can upgrade the noise tolerance capability effectively.
- It adopts both properties of over sampling and noise shaping to improve anti-noise capability for the proposed architecture itself.
- It can be used to cope with continuous noise.

- An adaptive gain method has been introduced to recovery the noise distortion.

Our approach combines the noise tolerance methods in previous researches and adopts the concept of noise shaping to deal with the interference causing by sustained noise. The experimental results show our system can indeed against the continuous noise and efficiently restore the interference signal. The effectiveness of the adaptive gain method has been demonstrated. Meanwhile, a best OSR is also discussed while applying different OSR to obtain different deviation. In addition, the relation between OSR and signal frequency are also demonstrated in our research.

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