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Practical Approach of Producing Delta Modulation and Demodulation

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Abstract: *Engineers working in the field of Speech Coding have been actively searching for methods to reduce the bandwidth consumed by quantized speech signals. Human ear has a build in ability to become more or less sensitive to audible signals. We have a difficult time hearing someone whispering at a rock concert. Some speech coding algorithms exploit this phenomenon by using a greater number of progressively smaller quantization levels for low amplitude signals and fewer, coarser quantization levels for large amplitude signals. This is known as non-uniform quantization. A slightly more complex approach takes advantage of strong correlation between adjacent speech samples, quantizing the amplitude difference (delta) between two samples as opposed to the entire sample amplitude, requires fewer quantization levels for the same signal quality and consequently, reduces the required bandwidth. Algorithms employing this technique are classified under the broad category of differential quantization or differential PCM (DPCM).*
Applying adaptive techniques to a DM quantizer allows for continuous step size adjustment.

I. Introduction

A modulation technique for encoding an analog-to-digital signal conversion. Delta modulation samples the analog signal at precise points in time, converting the signal into a series of segments, each of which is compared to the signal to determine if the amplitude changes. If there is a change in relative amplitude from one sample to the next, a series of bits is sent to indicate the extent of the increase or decrease in signal amplitude. If there is no change in amplitude, no bits are transmitted. We can make "educated guess" of what the next sample value depending on the current sample value. Though there is error, it is much less than peak to peak signal range. This concept is used in **Predictive Coded Modulation**, where instead of sending the signal, it transmits just the prediction error. Delta Modulation employs Predictive Coded Modulation.

Delta Modulation (DM or Δ-modulation) is an analog-to-digital and digital-to -analog signal conversion technique used for transmission of voice information where quality is not of primary importance. DM is the simplest form of difference pulse code modulation (DPCM) where the difference between successive samples is encoded into n-bit data stream. Its main features are:

- The analog signal is approximated with a series of segments
- Each segment of the approximated signal is compared to the original analog wave to determine the increase or decrease in relative amplitude
- the decision process for establishing the state of successive bits is compared by this comparison
- Only the change of information is sent, that is, only an increase or decrease of signal amplitude from the previous ample is sent whereas no change condition causes the modulated signal to remain at the same 0 or 1 state of the previous sample

In *delta modulation*, the analog signal is quantized by one-bit ADC (a comparator). The comparator output is converted back to an analog signal with a i-bit DAC, and subtracted from the input after passing through an integrator. The shape of the analog signal is transmitted as follows: a "1" indicates that a positive excursion has occurred since the last sample, and a "0" indicates a negative excursion has occurred since the last sample.

In 1962, Inose, Ysuda and Murakami elaborated on the single-bit oversampling noise shaping architecture proposed by Cutller in 1954. Their experimental circuits used solid state devices to implement first and second-order ∑-Δ modulators. The 1962 paper was followed by a second paper in 1963 which gave excellent theoretical discussions on oversampling and noise-shaping. These two papers were also the first to use the name *delta-sigma* to describe the architecture. The name *delta-sigma* stuck until the 1970s when AT&T engineers began using the name *sigma-delta.* Since that time both names have been used; however *sigma-delta* may be the more correct of the two. It is interesting to note that all the work described thus far was related to transmitting an oversampled digitized signal directly rather than the implementation of a Nyquist ADC.

II. Circuit Operation

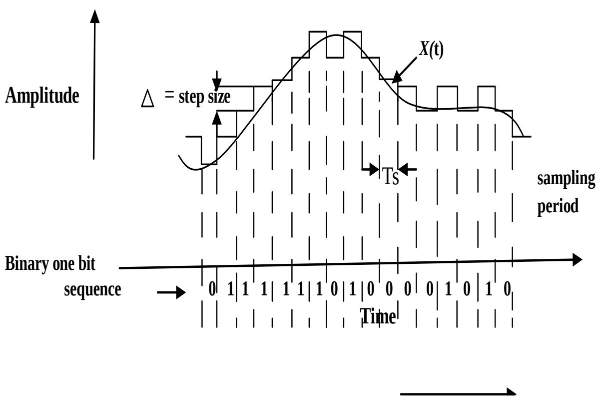
- **Reasons to use Delta Modulation**

PCM transmits all the bits which are used to code a sample. Hence, signalling rate and transmission channel bandwidth are quite large in PCM. To overcome this problem, **delta modulation** is used.

- **Working Principle**

Delta Modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample value and this result the amplitude is increased or decreased is transmitted. Input signal x(t) is approximated to step Δ signal by the delta modulator. This step size is kept fixed. The difference between the input signal x(t) and staircase approximated signal is confined to two levels, i.e. $+\Delta$ and $-\Delta$. Now, if the differences are positive or negative, then approximated signal is increased by one step, i.e. '$+-\Delta$'. Now, if the difference is positive, the approximated signal is increased by one step, i.e. '$+\Delta$'. If the difference is negative, then approximated signal is reduced by '$-\Delta$'.

When the step is reduced, '0' is transmitted and if the step is increased, '1' is transmitted. Hence, for each sample, only one binary bit is transmitted. Figure 1 shows the analog signal x(t) and its staircase approximated signal by the delta modulator.



DM waveform

- **Mathematical Expression**

Thus the principle of delta modulation can be explained with the help of few equations as follows:

The error between the sampled value of x(t) and last approximated sample is given as,

$e(nTs) = x(nTs) - x\hat{}(nTs)$ …… (i)

where,

$e(nTs)$ = error at present sample

$x(nTs)$ = sampled signal of x(t)

$x\hat{}(nTs)$ = last sample approximation of the staircase waveform.

If we assume u(nTs) as the present sample approximation of staircase output, then,

$U[(n-1)Ts] = x\hat{}(nTs)$ ……… (ii)

= last sample approximation of the staircase waveform.

Let us define a quantity b(nTs) in such a way that,

$b(nTs) = \Delta \operatorname{sgn} [e(nTs)]$ …… (iii)

This means that depending on the sign of error e(nTs), the sign of step size is decided. In other words we can writte,

$b(nTs) = \{+\Delta$ if $x(nTs) \geq x\hat{}(nTs)$

$= \{-\Delta$ if $x(nTs) < x\hat{}(nTs)$

………. (iv)

Also,

if $b(nTs) = +\Delta$

then a binary '1' is transmitted

And

if $b(nTs) = -\Delta$

then a binary '0' is transmitted

Here, Ts = sampling interval'

- **Transmitter Part**

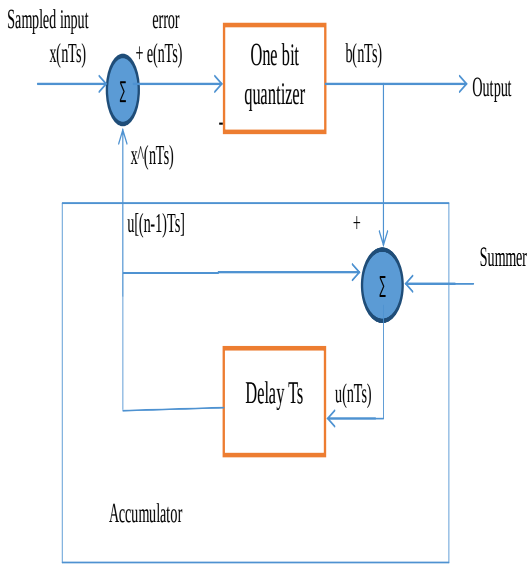


Figure 2shows the transmitter (i.e. the generation of Delta Modulated signal).

The summer in the accumulator adds quantizer output (±Δ) with the previous sample approximation. this gives the present sample approximation. That is,

u(nTs) = u(nTs – Ts) + [±Δ]

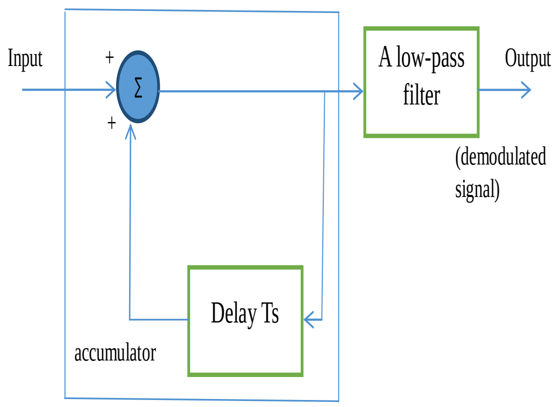
or u(nTs) = u[(n-1)Ts] + b(nTs)

……. (v)

The previous sample approximation u[(n-1)Ts] is restored by delaying one sample period Ts. The sampled input signal x(nTs) and staircase approximated signal x^(nTs) are subtracted to get error signal e(nTs). Thus depending on the sign of e(nTs), one bit quantizer generates an output of +Δ or -Δ. If the step size is +Δ, then binary '1' is transmitted and if it is –Δ then binary '0' is transmitted.

- **Receiver Part**

At the receiver end, shown in Figure 3



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the accumulator and the low pass filer (LPF) are used. The accumulator generates the staircase approximated signal output and is delayed by one sample period Ts. It is then added to the input signal. If the input is binary '1' then it adds +Δ step to the previous output (which is delayed). If the input is binary '0' then one step –Δ is subtracted from the delayed signal. Also the low pass filter smoothens the staircase signal to reconstruct the original message signal x(t).

Thus, we can reconstruct the original message signal x(t) by using delta modulation receiver circuit.

- **Transfer Characteristics**

The transfer characteristics of a delta modulated system follows a signum function, as it quantizes only two levels and also one-bit at a time.

The two sources of noise in delta modulation are 'slope overload', when steps are too small to track the original waveform, and 'granularity', when steps are too large.

- **Output Signal Power**

In delta modulation there is no restriction on the amplitude of the signal waveform, because the number of levels is not fixed. On the other hand, there is a limitation on the slope of the signal waveform which must be observed if slope overload is to be avoided.

- **Bit-Rate**

Delta modulation bit rate (r) = number ofbits transmitted/second
= number of samples/second X number of bits/sample
$=f_s \text{ X } 1 = f_s$

- **Transmitter Part**

1. Transmitter Clock Circuit
2. Voltage Comparator Circuit
3. Integrator Circuit
4. Unipolar to Bipolar Converter Circuit –transmitter part

- **Receiver Part**
- Receiver Clock circuit
- Unipolar to Bipolar Converter- receiver part
- Integrator Circuit
- Low Pass Filter Circuit

A. Transmitter Part

I. Transmitter Clock Circuit

- CLOCK Generator section providing clocks of adjustable frequencies of 32 KHz, 128 KHz and 256 KHz (to be used as a carrier of given sampling rates).
- Here the oscillator provides us with an output without any input. The output from the inventors (74HC04) goes to the binary counter (74HC4020) which is then fed to the multiplexer (74HC151).
- The outputs from the oscillator clock circuit are then connected to the D-flip-flop of Unipolar to Bipolar circuit of transmitter part.

Transmitter Clock

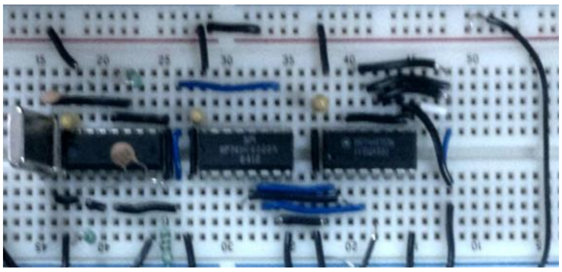


Figure 4: Transmitter Clock

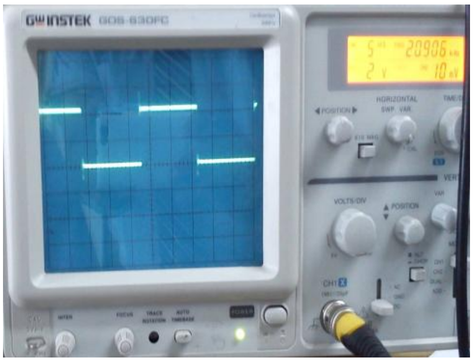


Figure 5: Output Clock

II. Voltage Comparator

A comparator is a device that compares two voltages or currents and switches its output to indicate which is larger. The required IC number is LM311. It is a comparator IC.

- **Circuit Symbol**

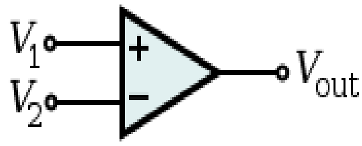


Figure 6

- **Circuit Diagram**

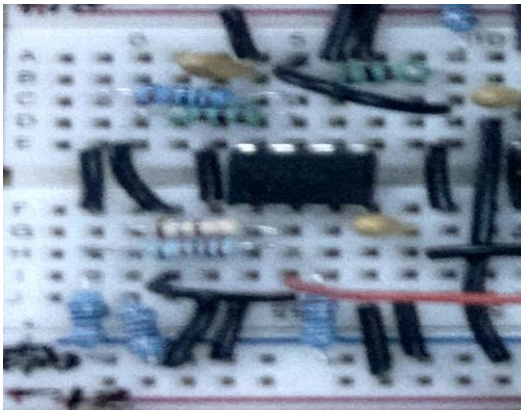
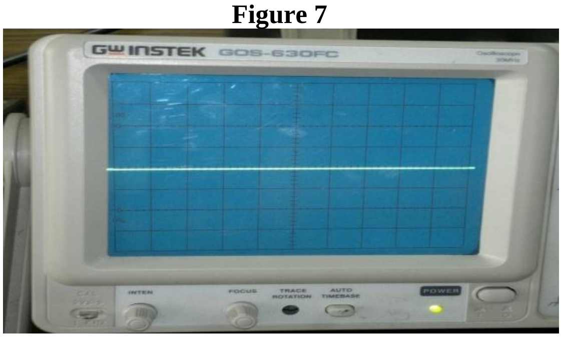


Figure 7



Output of voltage Comparator

III. Integrator Circuit

- The delta modulated signal is back in a feedback loop via an integrator to a summer. The integrator output is a saw tooth-like waveform. It is shown overlaid upon the message, of which it is an approximation.
- The saw tooth waveform is subtracted from the message, also connected to the summer, and the difference – an error signal – is the signal appearing at the summer output.
- An amplifier is shown in the feedback loop. This controls the loop gain. In practice it may be a separate amplifier, part of the integrator, or within the summer.
- Here we adjust the transmitter's LEVEL CHANGER preset until the output of INTEGRATOR is a triangle wave at the integrator's output should be 0.5V (approx.), this amplitude is known as the **integrator STEP SIZE** □.
- The output from the Transmitter's BISTABLE Circuit will now be a stream of alternate '1' and '0'; this is also the output of the delta modulator itself. The Delta Modulator is now said to be 'balanced' for correct option.

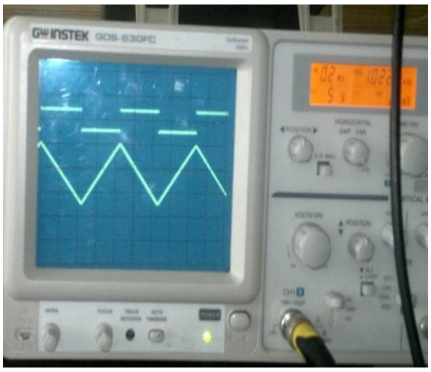


Figure 8 (Output Waveform of Integrator)

III. Unipolar to Bipolar Converter Circuit

- Unipolar to Bipolar circuit is required to ease the operation of the integrator circuit. This circuit is used so that the integrator circuit does not face any trouble while it is sending its output to the voltage comparator.
- The job of the Unipolar to bipolar converter block is to accept the data and to change its level before it is sent to the integrator.
- The input/output table for this block is as follows:

Input	Output
0	+5V
1	-5V

Output waveform of Unipolar to Bipolar Circuit

B. Receiver Part

I. Receiver Clock Circuit

- CLOCK Generator section providing clocks of adjustable frequencies of 32 KHz, 128 KHz and 256 KHz (to be used as a carrier of given sampling rates).
- Here the oscillator provides us with an output without any input. The output from the inventors (74HC04) goes to the binary counter (74HC4020) which is then fed to the multiplexer (74HC151).
- The receiver clock output then connected to the D-flip-flopof Unipolar to Bipolar circuit of transmitter part.

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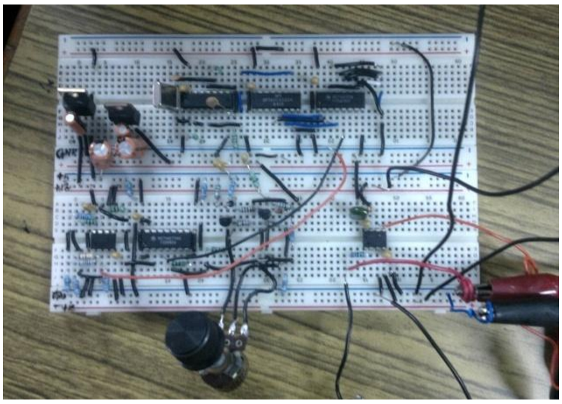
IV. Low Pass Filter

A **low-pass filter** is anelectronic filter that passes low frequency signals and attenuates (reduces the amplitude of) signals with frequencies higher than the cut off frequency. The actual amount of attenuation for each frequency varies from filter to filter. It is sometimes called a **high-cut filter**, or **treble cut filter** when used in audio applications.

In receiver circuit we use the low pass filter to smoothen the output waveform and reduce the noise. **Circuit Diagram of Delta Demodulation and Demodulation Process**



IV. Results And Discussions

After the paper work we have finally been able to understand the working principle of the different parts of delta modulation circuit. We at first made the individual circuits of the oscillator clock, voltage comparator, integrator, unipolar to bipolar converter and low-pass filter. Then we checked their outputs before connecting them in the final circuit. Then we received a waveform in the CRO. Though it was not perfect but still we finally were successful to create a delta modulation and demodulation circuit.

Here are some calculations that we have got while checking individual circuits:

Transmitter Clock:
Total Block = 4T= 16.8
Time = (16.8/4) = 4.2 us
Frequency = (1/T) = (1/4.2) = 238 KHz
Voltage Comparator:
Fixed voltage = 5V
Input voltage = 5V
Output voltage = while comparing above two voltages it gave 0V
Low-pass Filter:
Frequency (F1) = (1/2*pi*R*C)
Here, R = 47.5 Ω
C = 1 nF
Therefore, F1 = (1/2*pi*47.5*10^{-6})
= 3.35 KHz

V. Merits And Demerits

Merits

The delta modulation has certain advantages over PCM as below:

- Since, delta modulation transmits only one bit for one sample, therefore the signalling rate and transmission channel bandwidth is quite small for delta modulation compared to PCM.
- The transmitter and the receiver implementation are very much simple for delta modulation. There is no analog to digital converter required in delta modulation.

Demerits

Delta modulation has two major drawbacks that are:

- *Slope overload distortion*
- *Granular noise*

  Now, let us discuss the two drawbacks in detail.

a. Slope Overload Distortion

This distortion arises because of large dynamic range of input signal.
As it can be observed from figure 14, the rate of rise of input signal x(t) is so high that the staircase signal cannot approximate it, the step size ‘Δ’ becomes too small for staircase signal u(t) to follow the step segment of x(t). The error or noise is known as slope overload distortion. To reduce this error, the step size must be increased when slope of signal x(t) is high. since the step size of delta modulator remains fixed, its maximum or minimum slopes occur along straight lines. Therefore, this modulator is known as **Linear Delta Modulator (LDM)**.

b. Granular Noise

Granular noise occurs when step size is too large compared to small variations in the input signal. This means that for very small variations in the input signal, the staircase signal is changed by large amount because of large step size. Figure 14 shows that when the input signal is almost flat, the staircase signal u(t) keeps on oscillating by ±Δ around the signal. The error between the input and approximated signal is called **granular noise**. The solution to this problem is to make step size small.

- ***Adaptive Delta Modulation***

To overcome the quantization error due to slope overload distortion and granular noise, the step size (Δ) is made adaptive to variations in input signal x(t). Particularly in the step segment of the x(t) , the step size is increased. Also, if the input is varying slowly, the step size is reduced. Then this method is known as *Adaptive Delta Modulation (ADM).* The adaptive delta modulators can take continuous changes in the step size or discrete changes in the step size.

- ***Advantages of Adaptive Delta Modulation***

  Adaptive delta modulation has the following advantages over delta modulation:
- The signal to noise ratio becomes better than ordinary delta modulation because of the reduction in slope overload distortion and idle noise.
- Because of the variable step size, the dynamic range of ADM is wider than simple DM.
- Utilization of bandwidth is better than delta modulation.

VI. Applications

- It is used for digital voice storage
- It is used in voice transmission of active vibration control and is used in channel vocoder
- It is used for radio control of model aircraft, boats, and cars.
- The Star/Delta starter is probably the most commonly used reduced voltage starter.

VII. Conclusion

Thus we can see that we can be able to generate Delta Modulation and Demodulation technique of an analog signal. We know in Delta Modulation it transmits one bit per sample but it has the drawbacks (slope overload distortion and granular noise). We can use Adaptive Delta Modulation Signal to overcome these problems. Here, in practical, we get the output of delta modulation but some noise present there, though it has been overcome using appropriate parameters of right values.

Hence, overall we can see that it is working and its application can be seen in practical fields.
Limitation: Different parts of our circuit have performed well and we have got the correct waveform and results for each part. But the input-output independence of different parts should be synchronized to get a perfect output waveform of whole circuit. But it would take more time, that’s why we could not do this part.

References

[1] J. C. Bellamy, Digital Telephony, Wiley and Sons, New York, 1982.
[2] J. A. Greefkes and K. Riemens, “Code Modulation with Digitally Controlled Companding for Speech Transmission”, Philips Tech. Rev., pp.335-353, 1970.
[3] A. Gersho, “Principles of Quantization”, IEEE Transactions on Circuits and Systems, pp. 427-436, July 1978.
[4] N. S. Jayant and P. Noll, Digitall Coding of Wavforms: Principles and Applications to Speech and Video, Prentice-Hall, Englewood Cliffs, N. J., 1984.
[5] A. B. Jerri, “The Shannon Sampling theorem- Its Various Extensions and applications: A Tutorial Review, “Proceedings of the IEEE, pp.1565-1596, November 1977.
[6] R. A. McDonald, “Signal-to-noise and Idle Channel Performance of Different Pulse Code Modulation Systems-Particular Applications to Voice Signals”, Bell System Technical Journal, pp.1123-1155, Sept. 1966.
[7] P. Noll, “A Comparative Study of Various Quantization Schemes for Speech Encoding”, Bell System Technical Journal, pp. 1597-1614, November 1975.
[8] L. R. Rabiner and R. W. Schafer, Digital Processing of Speech Signals, Prentice Hall, Englewood Cliffs, N.J., 1978.
[9] M. Schwartz, Information, Transmission, Modulation, and Noise, McGraw Hill, New York, 1980.
[10] R. Steel, Delta Modulation Systems, Pentech Press, London, England, 1975.