Z-MOD PRINCIPALS

Z-Mod

TDR Foundation
TRUE DIGITAL RADIO.ORG

#005, US Complex, 120 Mathura Road, New Delhi 110076

Rakesh Aggarwal

Founder of TDR Foundation & Patent Holder

TDR Foundation, Unit #005, US Complex, Mathura Road, New Delhi-76

This document is Confidential, and technology disclosed is protected by Indian Patents 351456, 351458, 360271, 368356 and international patents US 10,979,368 B2, US 2021/0144038 A1 and many more. The document is intended for targeted technology partners and license holders under General Non-Discloser Agreement (NDA). If you disagree with NDA then do not go beyond this page and destroy this document immediately or return back to sender.

Version V19.1

For further information Contact: -

Rakesh Aggarwal
rakesh@comconservices.com
or
info@truedigitalradio.org
TDR Foundation.
Unit #005, US Complex, 120 Mathura Road,
New Delhi, India 110076

About TDR Foundation

TDR foundation is a not-for-profit organisation created specifically for support and promotion of cutting edge Innovative technologies.

Executives of Radio Group of the foundation are experts from the field of broadcasting & communication with commutative experience of more than 100 years in key positions.

Mr O K Sharma

Chairman

Mr. Vinod Mago

Vice President

Mr. Satya Pal

Project Director

Mr. V Ramarao

Advisor

Contents

"Z-Mod" MODULATION TECNIQUE

Preface

21st century humans transfer a multitude of information (which may include data, audio and/or video among others) to reach thousands of miles away using communication technologies. Typically, this information is transmitted by the use of electromagnetic waves. The medium used may be vacuum, air, glass or other substitutes. Currently, the frequency spectrum of the electromagnetic waves used may be from a few Hertz to several Giga-hertz. During the first two decades of the 21st century, use of the available spectrum has reached a point where the capacity to carry more information with existing technology is reaching saturation very fast. Usually, a higher frequency sine wave signal is used as a carrier and a lower frequency signal is used as the modulating signal. It is the modulating signal which carries the desired message/information. Among other reasons, modulation is necessary to allow multiple signals to travel simultaneously through the same medium while avoiding the interference which would be caused by different signals of the same signal frequency from multiple sources travelling through the same medium at the same time.

Background

Use of all kinds of specialised modulation methods and techniques are prevalent currently, each evolved for specific purposes. The most used and basic modulation forms may be broadly classified as AM, PM, FM, OOK and their combinations, using analogue or digital signals. In both the techniques, the baseband information is converted to higher frequency e.g., Radio Frequency signals, but in analogue modulation these RF communication signals are a continuous range of values, changing at every fraction of carrier cycle, whereas in digital modulation these are prearranged discrete states.

The three most prevalent types of analogue modulation are Amplitude Modulation (AM), Frequency Modulation (FM), and Phase Modulation (PM). There are many variants of these and many combinations of these prevalent.

In **amplitude modulation**, the amplitude of the carrier wave is varied in proportion to the message signal, resulting in a kind of multiplication of two instantaneous voltage values and the other factors like frequency and phase are not altered. The modulated signals are shown in the figure below, and its spectrum consists of three components, lower side band, upper side band and carrier frequency components. This type of modulation requires greater band width proportionate to signal bandwidth, more power. Filtering is difficult in this modulation type.

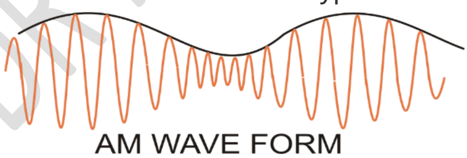
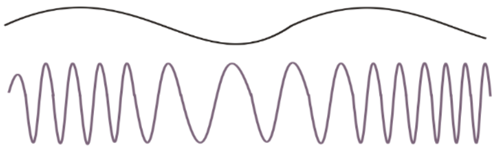


Fig -1

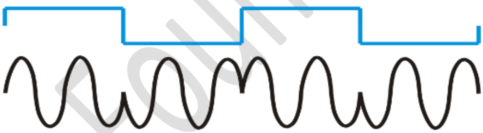
Frequency modulation (FM) changes the frequency of the carrier in proportion to the message or data signal amplitude while maintaining amplitude unaltered. The advantage of FM over AM is greater suppression of noise at the expense of spectrum bandwidth in FM. It is used in applications like radio, radar, telemetry seismic prospecting, and so on. The efficiency and bandwidths depend on modulation index and



Frequency Modulation

maximum modulating frequency.

In **phase modulation**, the carrier phase is varied in accordance with the data signal. In this type of modulation, when the phase is changed it also affects the frequency, so this modulation is also considered as specialised frequency modulation and it is not known so far, how phase can be varied without changing the frequency.



PM WAVE

Analog modulation (AM, FM and PM) are more sensitive to noise. If noise enters into a system, it persists and gets delivered by the receiver. Therefore, this drawback can be overcome by the digital modulation technique.

Digital Modulation

For a better quality and efficient communication, digital modulation techniques are employed. The main advantages of the digital modulation over analogue modulation include moderate RF power requirement, available higher data capacity, and high noise immunity. In digital modulation, a message signal is converted from analogue to digital message, compressed, and then modulated by using a carrier frequency.

In its simplest form, the carrier wave is keyed or switched on and off to create pulses such that the signal is modulated. Like the analogue, here the parameters like amplitude, frequency and phase variation of the carrier wave decides the type of digital modulation. One point to remember is that modulated waves remain analogue sine waves with minor alterations in properties.

(For more info ref: https://www.elprocus.com/different-types-of-modulation-techniques-in-communication-systems/)

Z-Mod Concept

This document describes a new, patented, “**SVURG”** technology named **“Z-Mod”**, which overcomes the limitations common to the presently used forms of analogue and digital modulation processes, challenging, and setting aside the existing belief, that all information is carried by the side bands in any modulated signal. “**Z-Mod” process** uses an innovative process, which eliminates use of side bands and increases spectrum capacity to carry more information both in the digital as well as analogue domains. The operating principles will become clear to readers as the concept is explained in simple terms. One apparent operational requirement for this new technology is that the more the transmission system uses, linearized RF stages, the higher the performance will be.

There are several unique advantages of this “**Z-Mod”** technology, which will unfold in the document, as we progress step by step. The technology is process based and is universal in nature having wide ranging applications in all type of communication systems.

Three prominent advantages of “**Z-Mod”** technology, are accomplished simultaneously and are worth noting are: -

1. Conservation of spectrum usage to just the carrier frequency itself (achieving theoretical limit of Zero bandwidth).
2. Information carrying Capacity increased by many folds.

3. Reduction in radiated power requirement to fraction of a percentage as compared to conventional methods.
4. It is GREEN energy and more efficient technology.

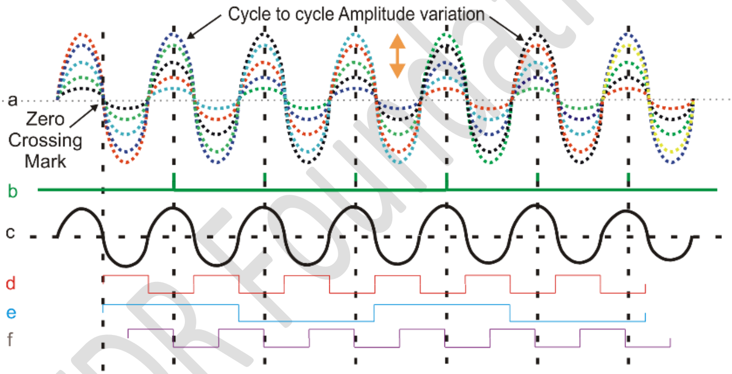
Why Z-Mod?

Modern day communications are becoming starved for bandwidth, despite use of advanced compression techniques, available spectrum seems to be saturating quicker than ever, despite more and more options being made available. This situation is driving technology to higher and higher frequency bands. At the same time the spectrum costs are rising in view of scarcity and cutthroat competition. Growing demand for connected personal devices and diversified digital services, is further exponentially increasing the demand for communication bandwidth capacity. To meet the growing demands, many innovative methods are in practice like, higher and higher compression ratios, banking more on human psycho effects and other feel-good masking factors. With recent work from home compulsions require huge amounts of data capacities, innovative solutions to increase transmission capacity are very much required.

The solution

The solution is now available in the innovative patented, technology named "**Z-Mod"**. This technology provides a new class of modulation process called **"Z Mod"**, which improves spectrum efficiency by removing side bands completely, with unlimited possibilities. The patented "**Z-Mod"** technology provides true inexhaustible bandwidth to carry more signal/data capacity, improve quality of communication with negligible noise and take all future communications to the next generation with abundance of spectrum capacity.

In “**Z-Mod”** systems the modulated carrier frequency remains constant and there is no variation in frequency dependent on modulating signal, but amplitude of each cycle can differ from previous cycle. In a synchronous detector as shown below, we only need sample at peak position of the carrier signal at detector, resulting in data values at 455 kHz sample rate (for 455 kHz IF frequency) and resolution of each sample can be up to 24 bits using existing receiver technology with 455 KHz IF frequency. This is equivalent to 10.92 megabits per second (455K*24), which is phenomenal increase in capacity.



AM Synchronous Detector

In terms of analogue signal Nyquist theorem says sample rate of twice the highest signal frequency is needed to reproduce original wave. In practice, we have seen analogue signal of ⅓ rd. of sample frequency can be reproduced nicely. This calculates to 455/3=151 KHz of analogue signal bandwidth can be reproduced by the above “**Z-Mod”** method.

To achieve this target, “**Z-Mod”** process generates each modulated carrier cycle individually in a cycle-by-cycle process, rather than generating carrier frequency as continuous wave and then modulating it. The main difference in “**Z-Mod”** approach is to interpolate, continuous modulating signals into synchronised stepped static samples at carrier frequency. The process results in transposing information to higher frequency, with one static value of modulating signal, for each carrier cycle at carrier frequency. Next step is to apply this static value at zero crossing point of carrier cycle generation process, to change desired sine property vector for the duration of one entire cycle, ending at another zero-crossing point. As a result, each carrier cycle is generated with pure sine wave properties for each individual complete carrier wave cycle. The next carrier wave cycle can differ in desired sine property, only at the ending zero crossing point of previous cycle and start zero crossing point of new cycle. For example, in case of amplitude modulation the amplitude of new carrier cycle can differ from previous cycle, according to static amplitude difference in samples derived from the modulating signal.

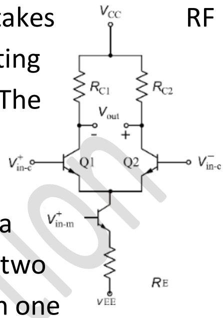
Whereas the invented “**Z-Mod”** method firmly establishes that “only the defined carrier frequency itself with least possible bandwidth, is sufficient to carry large amounts of digital and analogue information without need of any side bands”. Enormous amounts of analogue and digital signals can be sent and received, using just the carrier frequency alone. “**Z-Mod”** modulation process consists of directly generating modulated pure sine wave carrier cycles, in a cycle-by-cycle process, with control of sine property for each individual complete cycle at starting zero crossing point only. Each complete sine wave cycle from 0° to 360°, is individually generated in a variety of ways, resulting in side bands free modulated carrier waves.

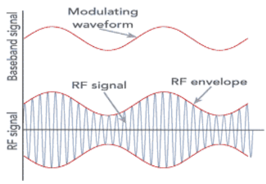
Each pure sine wave cycle generated is representing one static sample value of modulating signal multiplex, transposed at carrier frequency, for the duration of one carrier sine wave cycle period. Thus, every complete carrier sine wave cycle so generated will be a pure sine wave cycle (starting from one zero crossing point and ending at another zero-crossing point) with static pure sine wave properties. But it's one or more sine wave parameter can differ from one complete cycle to another complete cycle while both cycles remain pure sine wave cycles at defined carrier frequency. One or more properties of each carrier sine wave cycle is changed only at zero crossing point at the beginning of a carrier wave cycle, for that complete sine cycle, with static value of modulating signal for duration of that carrier cycle time period. Sine cycle will only change at the start zero crossing point, depends on which type of (FM, AM, PM & other combinations) conventional modulation is under consideration?

“Z-Mod” Mathematical Model

Analysis of standard AM

The amplitude modulation process in its popular known form uses a double balanced modulator which takes RF carrier signal as one input and modulating signal, for example audio as another input. The balanced modulator acts as an amplifier for the carrier signal whereas the modulating signal acts as a gain control. In a simple balanced amplifier circuit shown as two transistors connected in differential par with one transistor acting as constant current drive. The carrier signal is applied to differential pair transistor bases and modulating signal is connected to base of the constant current source transistor base. As the modulating signal changes the current drive through bottom transistor the gain of the RF amplifier changes at V out. In effect the circuit acts as multiplier of the two signals and resultant wave form is amplitude modulated signal as shown below: -





This section establishes how we can eliminate generation of side bands from the modulation process using **“Z-Mod”** process. To understand the concept reader needs to keep these unique conditions in mind: -

- In conventional modulation process Carrier is generated as continuous wave of sine cycles and modulation is applied to these Carrier wave.

  *Where as in **"Z-Mod"** process, we consider and generate only one sine cycle at carrier frequency at a time. The start and end of carrier sine cycle is inseparably bonded to zero crossing point, ensuring pure sine wave property for each 0˚ to 360˚ individual sine wave cycle at the output of modulated carrier wave.*
- In conventional modulation, Carrier is continuous wave of sine cycles as one of the two inputs of modulator.

  *Where as in **"Z-Mod"** process, each individual modulated carrier sine wave cycle is generated standalone independently and may be in continuation to previous or next cycle or discontinued.*
- In conventional modulation the modulating signal which is generally much slower frequency in comparison to carrier, is used in its natural form.

  *Where as in **"Z-Mod"** process, modulating signal is interpolated to carrier frequency and made **static** (constant) for the duration of each carrier sine wave cycle.*
- In conventional modulation the information is contained in side bands generated during modulation process.

  *Where as in **"Z-Mod"** process, there are no side bands generated and total information is contained within carrier frequency*

Modulation equations

Elimination of side bands in ***"Z-Mod"*** can be established by study of various modulation equations, Theory of generation of a **conventional amplitude modulated signal** in following four steps:

1. Carrier signal
2. Modulating signal
3. Overall modulated signal for a single tone
4. Modulation to cover a typical audio signal

These steps will be covered in greater details below and in later separate section, it will be applied to ***"Z-Mod"*** system also to establish the difference.

1. Carrier signal equations

Looking at the theory, carrier is described in terms of a sine wave as follows:

$$C(t) = C \sin(\omega c + \phi) \qquad \text{......(i)}$$

Where:

carrier frequency in Hertz is equal to $\omega c / 2\pi$
C is the carrier amplitude
ϕ is the phase of the signal at the start of the reference time

Both C and ϕ can be omitted to simplify the equation by changing C to "1" and ϕ to "0".

$$C(t) = \sin \omega c \qquad \text{......(ia)}$$

2. Modulating signal equations

The modulating waveform can either be a single tone or multi-tone. single tone can be represented by a sine waveform, or the modulating waveform could be a wide variety of frequencies - these can be represented by a series of sine waveforms added together in a linear fashion.

For simplicity we will start at the equation for a single tone waveform and then expand the concept to cover the more practical case. Single tone waveform will look like this :-

$$m(t) = M \sin(\omega_m + \phi) \qquad \text{....(ii)}$$

Where:

modulating signal frequency in Hertz is equal to ω_m or 2 πFm

M is the carrier amplitude

ϕ is the phase of the signal at the start of the reference time

Both M and ϕ can be omitted to simplify the equation by changing M to "1" and ϕ to "0".

$$m(t) = \sin\omega_m \qquad \text{....(iia)}$$

It is worth noting that normally the modulating signal frequency is well below that of the carrier frequency.

3. Overall modulated signal for a single tone

The equation for the overall modulated signal is obtained by multiplying the carrier and the modulating signal together at time (t): -

$$y(t) = [A + m(t)].c(t) \qquad \text{......(iii)}$$

The constant A is required as it represents the amplitude of the waveform.

Substituting in the individual relationships for the carrier and modulating signal, the overall signal becomes:

$$y(t) = A\,[\sin(\omega_c t).M\sin(\omega_m t+\phi)] \qquad \text{.....(iv)}$$

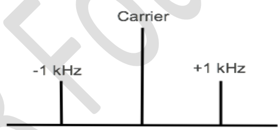
The trigonometric function can be expanded, to generate an equation, that includes all the components of the modulated signal:

$$y(t) = A.\sin(\omega ct)+\frac{AM}{2}[\sin((\omega c+\omega m)t+\varphi)]+\frac{AM}{2}[\sin((\omega c-\omega m)t-\varphi)] \quad \text{....(v)}$$

In this equation, three terms can be seen which represent the carrier, and upper and lower sidebands:

Carrier: **A .sin (ωct)**
Upper sideband: **A . M/2 [sin ((ωc + ωm) t + φ)**
Lower sideband: **A . M/2 [sin ((ωc - ωm) t - φ)**

Note also that the sidebands are separated from the carrier by a frequency equal to that of the modulating signal tone of 1 KHz.



Spectrum (sidebands) from carrier modulated by 1 kHz tone

It can be seen that for a case where there is 100% modulation, i.e., M = 1, and where the carrier is not suppressed, i.e. A = 1, then the sidebands have half the value of the carrier amplitude, i.e., a quarter of the power each. The carrier power remains constant at 100% thus leading to total power in modulated signal becoming 150%.

Equation to cover a typical audio signal

In this case when there is live audio the expression representing audio signal can be considered comprising of multiple sine frequencies from m_1--- to ---m_n. The original signal equation changes to

m(t) = M1 sin(ω1m 1 + φ1) + M2 sin(ω2m 2 + φ2)+ ----Mn sin (ω nm n + φn) ...(vi)

putting these values of **m(t)** in modulated signal equation, it becomes

y(t) = [A + M1 sin(ω1m 1 + φ1)+ M2 sin(ω2m 2 + φ2)+ ---- Mn sin(ωnm n + φn)].sin(ωct) ..(vii)

This is a complex waveform having side bands of all modulating signal frequency components (**M1** to **Mn**) in the modulated output.

Analysis Applied to "Z-Mod" (AM)

For establish the removal of side bands in the output, we will apply **"Z-Mod"** process to the above equations. To start with, the same 2 equations for **C(t)** and **m(t)** are used to establish how **"Z-Mod"** process changes **m(t)** & **y(t)** equation into static values: -

'**C(t)**' remains unchanged, except that processing is applicable for one cycle at a time, according to **"Z-Mod"** ***cycle by cycle*** process and the carrier frequency and amplitudes remain unchanged during each individual carrier cycle.

$$C(t) = A \sin(\omega c + \phi) \text{ ...unchanged} \qquad \text{..... (a)}$$

for **"Z-Mod" process,** it is for an entire cycle and can omit ϕ. It can be written for complete cycle from 0˚ to 360˚

$$C(t)_{0 \to 360} = A \sin(\omega c) \qquad \text{....................(b)}$$

Where $C(t)_{0 \to 360}$ denotes one complete carrier sine wave cycle starting at zero crossing point and ending at zero crossing point, as defined for **"Z-Mod" process**.

Single cycle equations

As the **"Z-Mod"** process is for each individual carrier sine wave cycle, equation (b) is for one complete sine cycle of the carrier wave from 0˚ to 360˚ only. Next carrier cycle will also be governed by the same equation.

Processing of modulating signal **m(t)** in **"Z-Mod"** process requires '**m(t)**' to be interpolated to carrier frequency. This is achieved by sampling **m(t)** at carrier frequency **'Fc'**. Outcome of the sampling process, produces an integer value, static for each one carrier cycle duration, representing its instantaneous/ average amplitude value *transposed at carrier frequency.* Each sample can differ from one another but the change will only be at the carrier frequency.

The sampled modulating signal ‘Msin(ωm + ϕ)’, results in each sample becoming a **static** numerical value at carrier frequency, and can be denoted by **‘MΔ’**, which represent only instantaneous integer value of modulating signal for that individual carrier cycle duration interpolated at carrier frequency and not having any component of modulating signal frequency ‘ωm’.

Original Equation for **m(t)** from previous section

$$m(t) = M\sin(\omega m + \varphi) \qquad ...(c)$$

Denoting sampled static amplitude of **‘sin(ωm + φ)’** by ‘am’ denoted by **‘Δ ‘** it becomes: -

$$m(t) = M^{*}am = M\Delta = M(t)_{0\to 360} \qquad ... (d)$$

Where $M(t)_{0\to 360}$ Is interpolated static sample value of modulating signal for one carrier cycle duration.

It is evident from the above equation that **‘M(t)’** becomes a pure numeric static value for the entire duration of one carrier cycle and does not have “sin(ωm)” component of the modulating signal frequency, which gets transposed at carrier frequency **‘C(t)’**.

The original modulated signal equation

$$y(t) = A\,[\sin(\omega ct).M\sin(\omega_m t+\phi)] \qquad .....(iv)$$

$$y(t) = A\,[\sin(\omega ct).M\Delta] \qquad \text{.....(e)}$$

by replacing 'M sin (ωmt+ ϕ)' with **MΔ, it** can be rewritten for **"Z-Mod"** modulated output as follows: -

$$Y(t)_{0\to 360} = C(t)_{0\to 360}.M(t)_{0\to 360} \qquad \text{....................(f)}$$

Substituting $M(t)_{0\to 360}$ from equation (d): -

$$Y(t)_{0\to 360.} = M\Delta.C(t)_{0\to 360}$$

$$Y(t)_{0\to 360} = M\Delta.A\sin(\omega c) \qquad \text{......(g)}$$

The equation (g) for single cycle of modulated carrier output has no 'ωm' component, and only static value of 'Sin(ωm + ϕ)' interpolated at carrier frequency is present in the modulated output. Which can include average, sampled static value of modulating signal multiplex, for each one carrier cycle period and this value is denoted as **"MΔ"** being incremental static value of amplitude M.

Where **'Δ'** is the sample fraction of the modulating signal amplitude during each individual carrier cycle period under processing and is a static value, dependent only on fraction representing percentage voltage, without any effect of the "ωm" or the frequency component of the modulating signal) for each complete carrier sine wave cycle from 0˚ to 360˚. Thus, putting this static value for **m(t)** in output equation **y(t)** for each individual cycle, becomes: -

$$Y(t)_{0\rightarrow 360} = A\left\{ C(t)_{0\rightarrow 360} * M\Delta \right\} \qquad \text{..........(g)}$$

Rewriting the equation by bringing the amplitude part to gather

$$Y(t)_{0\rightarrow 360} \quad = \quad C(t)_{0\rightarrow 360} = M\Delta * A \qquad \text{....(h)}$$

The above equation (h) shows the modulated carrier output individual cycle is a pure sine wave with amplitude equal to "**A*MΔ**". Thus in **"Z-Mod"** process the carrier frequency function is unchanged by the frequency of modulating signal during any complete carrier sine wave cycle. Multiplication of modulating signal is shifted from continuously changing 'sin(**ω**m)' to a static constant **'ΔM'**, and applied only at zero crossing point of each carrier cycle. Value of "**ΔM**" can change with next sample, at the beginning of next carrier cycle only, ensuring modulated output carrier cycle will be pure sine wave cycle with different amplitude vector. Repetitive application of this **"Z-Mod"** process, in a cycle-by-cycle steps will generate modulated RF output, complying with existing standards, with distinct advantage of ZERO SIDEBANDS in the output. With much larger base band signal bandwidth capacity, in accordance with **Nyquist theorem** where carrier frequency will become the sample rate.

Thus the, **"Z-Mod"** process produces a stream of pure sine wave cycles one after another in a cycle-by-cycle process, with each carrier cycle with different amplitude/frequency/phase. No side bands will be generated if we apply the sampled static amplitude information only at the starting zero crossing point of the carrier sine cycle avoiding any change in sine function during each cycle by generating each complete cycle with pure sine function. In other words, effectively in this process there is no continuous multiplication of the modulating signal with carrier waves. As a result, we have produced only **pure sine wave** carrier cycles of differing amplitude, where amplitude of each carrier sine cycle, is representing modulating signal's average or sampled, static amplitude value, during that carrier cycle period. These sine wave cycles can carry large amount of information without the need of any spectrum bandwidth, by just using the carrier frequency itself.

Nyquist Theorem

Natural question will arise if the modulating signal is static, then where the dynamic information of the modulating signal is carried by carrier. The capacity to carry large amounts of dynamic data can be understood by visualising the variations in amplitude from one carrier cycle to another carrier cycle. The **"Z-Mod"** process of cycles-by-cycle process, where each modulated carrier cycle's peak amplitude embeds one sample value of modulating signal with 24 bits or higher bit depth. This high data sample in each individual carrier cycle, can carry analogue signal bandwidth, which is up to half of carrier frequency as per **Nyquist theorem**.

Shannon's theorem

Unique **"Z-Mod"** process provides extra ordinary bandwidth capacity and satisfies **Shannon's theorem,** $D = B\,\mathrm{Log}_2\left(1+\frac{S}{N}\right)$ by providing extra ordinary high signal to noise ratio, due to, ultra-narrow carrier bandwidth requirement, of less than **1 Hz** at the receiver. This SNR advantage can be understood by applying minimum detectable signal (MDS) principal, suggesting more than 160 dB signal to noise ratio are practically possible and can reach up to thermal noise limit of 174 db. This figure of 174 is known as **Johnson Noise,** based on universally accepted **Boltzmann's constant**.

Johnson Noise

This reduction in spectrum bandwidth to fraction of Hertz, leads to restricting system noise to reach theoretical limits by the **"Z-Mod"** process, satisfying **Shannon's theorem**, which sates, data capacity "D" for specified bandwidth 'B' is directly related to:

$$D = B \, \mathrm{Log}_2\left(1+\frac{S}{N}\right) \qquad \ldots\ldots(i)$$

Where S/N is a linear ratio of signal and noise ratio in numerical values, further explained in detail, based on well-established noise floor calculations, in terms of absolute Minimum Detectable Signal or ('MDS'): -

MDS = 10Log (kTo*1e3) + NF + 10 Log (BW) + SNR (dBm) …(ii)

For a signal bandwidth of 10 KHz and a noise figure of Front end 1.5 dB with signal SNR of 40 db. resulting in -92.5 dBm.

MDS = -174 + 1.5 + 40 + 40 (dBm) = -92.5 dBm …(iii)

MDS (Z-Mod) = -174 + 1.5 + 0 + 10 (dBm) = -162.5 dBm …(iv)

In practice for digital signals, for reliable detection, a signal to noise ratio of 6 dB is sufficient however in practice 10 dB SNR is considered robust and safe resulting in available noise floor of -162.5 dBm.

Converting this 162 dB SNR from log scale to linear number results in a mind-boggling number of 134,217,728 and can be rounded off to more than 134 Mega Bits per second. This is a number derived from '**Shannon's theorem**' and to prove it practically, work is in progress on development of a usable product, with 'ultra narrow band width' front end.

From the above mathematical model supported by well-established theorems it becomes absolutely clear from scientific analysis, that from an amplitude modulated pure sine wave carrier having 1 Hz or less bandwidth, it is possible to carry data of more than 134 megabits per second. We have also established how modulated carrier waves with less than 1 Hz bandwidth can be generated using **"Z-Mod"** process.

Mathematical Analysis of Z-Mod

Zero crossing modulation refers to a modulation technique, where the carrier signal crosses zero amplitude at specific instances resulting in no side bands. This modulation can be analysed in both time domain and the frequency domain.

Let us consider a simple case where we have a continuous wave (CW) carrier signal represented as

C(t) = A Cos (2∏fct)

Where

A is the amplitude of the carrier wave, and fc is the carrier frequency

In zero crossing modulation, we modulate the carrier signal such that it crosses zero amplitude at specific time resulting in no side bands. To achieve this we can multiply the carrier signal by a periodic pulse train with a duty cycle of D and a period of T

$$P(t) = \sum_{n=-\infty}^{+\infty} \mathrm{Rect}\,\frac{(t - nT)}{T}$$

Where $\frac{(t-nT)}{T}$ is equal to " x' and rect(x) is the rectangular function equal to 1 for (x) ≤ 0.5 and O otherwise.

the modulation signal is then given by

S (t) = C (t).P(t) = A Cos (2∏fct)

$$\sum_{n=-\infty}^{+\infty} \mathrm{Rect}\,\frac{(t - nT)}{T}$$

Now let's Analyze this modulation in both the time and frequency domain.

Time domain Analysis :-

In the time domain, we can see that the modulation signal S(t) is a series of pulses, Where each pulse is centered at 'nT' and has the duration of D.T, the carrier signal is multiplied by these pulses, which result in a signal that crosses zero at the edge of each pulse, therefore in the time domain we have zero crossing at the edge of the pulse which is characteristic of zero crossing modulation.

In the time domain the Z-Mod/zero crossing modulation signal S(t) is given by

$$S(t) = A \cos \sum_{n=-\infty}^{+\infty} \mathrm{Rect}\frac{(t - nT)}{T} \; (2\, f_c . t).$$

A is the amplitude of the carrier wave fc is the carrier frequency.

T is the periods of rectangular Pulses.

rect(x) is the rectangular function defined as 1 for |x| ≤ 0.5 and zero otherwise. the sum-over 'n' represents an infinite train of rectangular pulse centered at multiples of T Seconds in the time domain. S(T) looks like a series of pulses that occur at regular interval centered at 'nT', where 'n' is an integer.

The width of each pulse is D.T where D is the Duty cycles.

Frequency Domain Analysis: -

In the frequency domain we can analyze the spectrum of the modulated signal S(t). To do this we can use the Fourier transform of the modulation signal S(t) is given by

$$S\ (f) \quad = F\ \{\ S\ (f)\ \}\ = F\ \{\ A\ Cos\ (2\ fct)\ .\ P\ (t)\}$$

Now let's sub divide the Fourier transform into components:

1. The Fourier transform of the carrier signal A Cos (2 fc.t) is 2 Direct delta functions at ± fc

2. The Fourier transform of the rectangular pulse train P(t) is a sine function centered at 0 Hz with a bandwidth determine by the Duty cycle D

Since we are multiplying the carrier signal by the pulse train in the time domain, the resulting spectrum in the frequency domain will have components ± fc (from the carrier) and a sine shaped envelop, centered at 0Hz from the pulse train. The Key characteristic here is that there is no side bands around the carrier frequency and the spectrum is confined to this frequency range due to zero crossing modulation.

Zero crossing modulation results in a modulation scheme, where the modulated signal crosses zero amplitude at a specific time and this is reflected both in time and frequency domain with no side bands around the carrier frequency.

Using the Fourier transform

$$S\ (f) \quad = F\ \{\ A\ Cos\ (2\ fct).\ P(f)\}$$

Here S (f) represents frequency Domain representation of S (t)

The Fourier transform of the carrier signal A Cos (2 fct) is a pair of delta function located at ± fc Hz.

The Fourier transform of the rectangular pulse train P(t) is a sine function defined as

as

$$P(f) = \mathrm{Sine}\left(\frac{F}{1/T}\right)$$

Where sine (x) = $\frac{\mathrm{Sine}\ (\pi x)}{\pi x}$

When multiply the carrier signal by the pulse train in the time domain it is equivalent to convolving their Fourier transform in the frequency domain. This convolution result in the appearance is of side bands around the carrier frequency but the band width of these side bands is determined by the sine function and is limited by the duty cycle D

Visualization: -

In time domain

To visualize this, consider a time domain plot of the zero crossing modulated signal S(t) and its corresponding frequency domain representation S(f)

It will be seen as a series of pulse, centred at nT where n is an integer.

Each pulse has a duration of D.T

The carrier frequency fc is not visible directly in the time domain plot.

In frequency domain

It will be seen as spike (delta function) at ±fc representing the carrier.

There will be a sine shaped envelope around 'fc' due to convolution with the sine function, the key point is that the modulation scheme result in no additional side bands outside the sine shaped envelop centered around 'fc'.

This visualization demonstrate how zero crossing modulation maintains narrow frequency spectrum with no side bands while is a distinct feature of this 'Z-Mod' modulation technique.

Shannon's Information Capacity Theorem

The Goal of a Communication system Designer is to configure a system that transport a message signal from a source of interest, across a noisy channel to a user at the other end of the channel with the following objective.

The message signal is delivered to the user both efficiently and reliably subject to certain design constraints such as, allowable transmit power, available channel band width and affordable cost of building the system.

In the case of digital communication system, reliability is commonly expressed in term of bit error ratio (BER), or probability of bit error measured at the receiver output. Clearly smaller the BER, the more reliable the communication system.

A question that cross to mind in this context is whether it is possible to design a communication system that operates with zero BER even though the Channel is noisy. In an ideal setting the answer to this question is an emphatic yes. The answer is embedded in one of Shannon's celebrated theorems which is called the information capacity theorem.

Let 'B' denote the channel bandwidth and let SNR denote the received signal to noise ratio the information capacity theorem states that ideally these two parameters are related as.

$$C = B \log(1 + SNR)$$

When 'C' is the information capacity of the channel. The Information Capacity is defined as the maximum rate at which information can be transmitted across the channel without error, it is measured in bits per second (b/s). For a prescribed channel bandwidth 'B' and received SNR, the information capacity theorem tells us that a message signal can be transmitted through the system without error even when the channel is noisy, provided that the actual signaling rate 'R' in bits per second, at which data is transmitted through the channel, is less than the information capacity 'C'.

Unfortunately, Shannon's information capacity theorem does not tell us how to design the system. Nevertheless from a design point of view the theorem is very valuable for the following reason.

1. The information capacity theorem provides upper limit on which rate, data transmission is theoretically attainable for prescribed values of channel bandwidth 'B' and received SNR. On this basis we may use the ratio

$$\eta = \frac{R}{C}$$

as a measure of the efficiency of the digital communication system under study. The Closer η is to unity the more efficient the system is.

2. Equation C=B (1+SNR) B/S provide a basis for the tradeoff between channel bandwidth 'B' and received SNR. For prescribed signaling rate 'R' we may either reduce the SNR or increase the channel bandwidth 'B' hence, motivation for using a wide band of modulated scheme for increased capacity performance.
3. Equation C=B (1+SNR) B/S Provide an idealized framework for comparing the noise performances of one modulation scheme against another.

About Digital Communicatin

When we speak of a digital communication system having a low bit error rate say the implication is that only a small fraction in a long stream of binary symbol is decided in error by the receiver. The issue of the receiver determining whether a binary symbol sent over the noisy channel is decidedly in error or not is of fundamental importance of the design in a digital communication systems. It is therefore appropriate briefly to discuss the basic issue to motivate the study of communications system in detail.

Suppose we have a random binary signal m(t) consisting of symbol 1 and 0, where symbol 1 is represented by a constant level +1 and symbol 0 is represented by a constant level -1, each of which lasts for a duration T. Such a signal may represent the output of a digital computer or the digital version of a speech signal to facilitate the transmission of this signal ever a communication channel. We can employ a simple modulation scheme, known as Phase shift keying. Specifically, the information bearing signal m(t) is multiplied by a sinusoidal carrier wave A Cos (2πfct) where Ae is the carrier amplitude fc is the carrier frequency and time 't' figure shows a block diagram of the transmitted the output of which is defined by

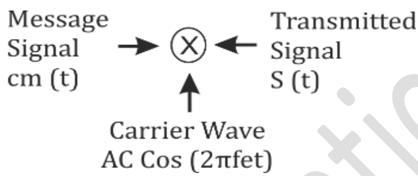
$$S(t) = \begin{bmatrix} AC \cos(2\pi fct) & \text{for symbol 1} & \ldots\ldots\ldots\ldots\ldots 1 \\ - AC \cos(2\pi fct) & \text{for symbol 0} & \ldots\ldots\ldots\ldots\ldots 2 \end{bmatrix}$$

Where the carrier frequency fc is a multiply of 1/T

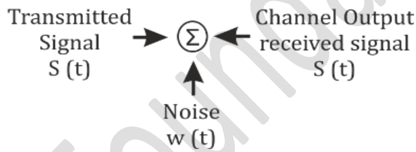
The Channel is assumed to be distortion less but noisy as depicted in below figure the signal received x (t) in this is defined by

$$x(t) = s(t)+w(t) \qquad \text{.........3}$$

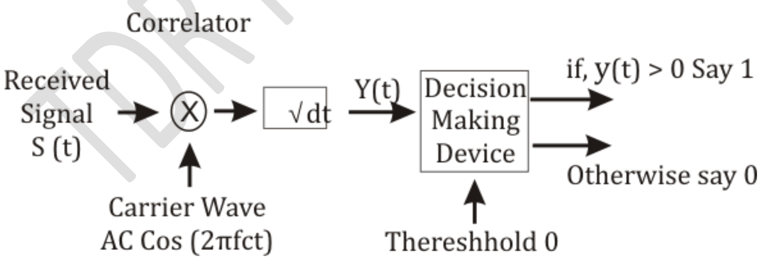
Where w(t) is the additive channel noise.



and



further



The receiver consists of a correlator followed by a decision-making device, as depicted in figure above the correlator multiplies the received signal x(t) by locally generated carrier

Cos (2πfct) and the integrates the product over the symbol interval 0<f<T producing the output.

$$y_T = \int_0^T x\,(t)\,\mathrm{Cos}\,(2\pi fct)\,dt \quad \ldots\ldots 4$$

Substituting Eq (2) and (3) into(4) and invoking the assumption that the carrier frequency 'fc' is a multiple of T/T we obtained .

$$y(t) = \begin{cases} +\dfrac{\underline{AC}}{2} + \text{wt for Sysmbol 1} \\ -\dfrac{\underline{AC}}{2} + \text{wt for Sysmbol 0} \end{cases}$$

Where WT is the contribution of the correlator output due to the channel noise 'w(t)' to reconstruct the original binary signal m(f) the correlator output YT is compared against the threshold of Zero Volts by the operation of which is based on the following rule.

If the correlator outputs Y(t) is greater than zero the receiver output symbol 1 otherwise, it outputs symbol 0. With this background we may now discuss/raise some basic issues. First from Fourier analysis we find that the time bandwidth product of a pulse signal is constant. This means that the bandwidth of a rectangular pulse of duration T is inversely proportional to T. The transmitted signal in figure consists of the product of this rectangular signal and the sinusoid signal (carrier) Ac Cos 2πfct. The multiplication of a Signal by Sinusoid has the effect of shifting the Fourier transform to the right by fc and to the left by an equal amount except for the scaling factor of ½. It follows therefore that the bandwidth of transmitted signal m(t) and therefor the required channel bandwidth is inversely proportional to the reciprocal of the symbol duration T. For the problem at hand the reciprocal of T is also the signaling rate of the system in b/s.

There are however some other issues that require theoretical consideration.

1. What is the justification for the receiver structure of figures.
2. The contribution ‘Wt’ is the value of random variable W produced by sampling a certain realization W(t) of the channel noise t=T in accordance with eq No3 and 4. How we do relate the statistics of the randoms variable W to the statistical characteristics of the channel noise.
3. The receiver depicted in figures makes occasional errors due to the random nature of correlator output that is the receiver decide in favor 0 given that symbol 1 was actually transmitted and vice of versa. What is the probability of decision errors.

Moreover, these are some important practical issues that need attention.

1. Channel bandwidth is highly valuable resource how we choose a modulation scheme that conserve bandwidth in a cost-effective manner.

2. The binary signal m(t) may include redundant symbol introduced into it using channel encoding so to provide protection against channel noise. How we design the channel encodes in the transmitter and the channel decodes in the recover to come very close to Shannon’s information capacity theorem in a physically manner.

The locally generated carrier in the receiver of figure is physically separate from the carrier source used for modulation in the transmitter how we do use synchronize the receivers to the transmitter with respect to both the carrier phase and symbol timing so as to justify the use of Eq ..4 as the basis of decision making in the reconstruction of the originally binary signal.

Universal Application of "Z-Mod" Concept

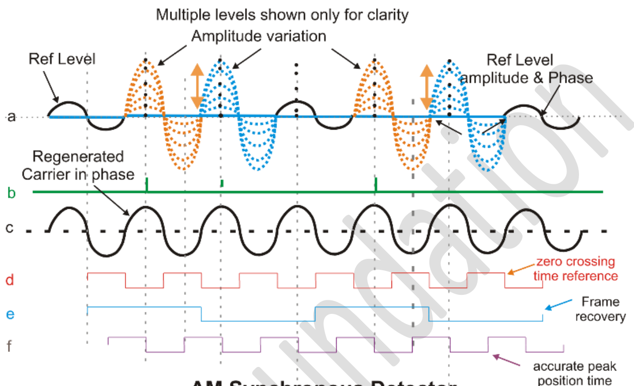
The **"Z-Mod"** process is applicable to all kinds of modulation types known or to be created, using this innovative implementation method. It is simple to apply it to various forms of modulation by simply ensuring the sine parameter changes only at 'ZERO CROSSING' point of the modulated carrier wave sine cycle being generated.

This **"Z-Mod"** process of changing sine cycle property at zero crossing according to **'Δ'** static value (in synchronized incremental steps), repeats itself for generating pure sine wave carrier frequency wave, one cycle after another, with differing amplitude in case of 'AM' and differing pre-defined frequencies 'in steps' for 'FM" and differing starting phase angles for 'PM'. This unique method of using the precise zero-crossing point, to change sine cycle property with modulating signal static values interpolated at carrier frequency, results in removal of side bands completely in the modulated output.

AM Implementation: -

Amplitude modulation is known for its simplicity and is considered delivering lowest quality signals. With **"Z-Mod"** this is no longer the case and we will describe in details a robust and hi quality AM concept capable of delivering large amounts of information.

To establish that large amount of data can be carried by carrier pure sine wave cycles without need of side bands, first we will consider Amplitude Modulation wave form. The carrier sine wave shown in Figure below, is frame-based implementation with 3 cycle frame period. Multiple dotted sine traces in each cycle are to depict that only amplitude



AM Synchronous Detector

changes in each data cycle and timing does not change. Thus, recovering say 1 KHz audio signal from 455 KHz carrier IF signal applied to detector needs only a simple diode detector. Here the point to note is that, from filter efficacy point of view, it will not make any difference whether the IF signal frequency at detector is '455 KHz' or '455 ± 1 KHz' to reproduce 1 KHz baseband signal.

In 'Z-Mod' systems the frequency remains constant, but amplitude of each cycle can differ from previous cycle with precise peak of cycle predictable. In synchronous detector as shown above, we only take sample at the peak position of 455 KHz IF signal, resulting in sample data values at 455 KHZ sample rate and resolution of each sample can be up to 24 bits easily using existing technology. Result is 24 bits of data at 455 KHZ rate, equivalent to 10.92 megabits per second, which is phenomenal increase in capacity. In analogue terms **Nyquist theorem** says twice the highest signal frequency samples are needed to reproduce original wave. Practically we have seen analogue signal of ⅓ rd. of sample frequency can be reproduced very nicely. This calculates to 455/3=151 KHz of analogue signal bandwidth can be reproduced by the above **"Z-Mod"** method.

There is a genuine concern about signal to noise ratio of received signal available at detector input to support 24-bit resolution. For resolving 24-bit resolution we need signal to noise ratio of 2 raise to 24 times, which in dB is 6x24=144 db. The available signal to noise ratio with 10 dB reserve margin from equation (iv) is: -

MDS (1 Hz) = -174 + 1.5 + 1 + 10 (dBm) = -161.5 dBm ......(iv)

This is far-far superior to required 144 dB to resolve 24 bits and in future 28 bits resolutions can be possible.

FM Implementation: -

Same way we can show how the **"Z-Mod"** principals can be applied to Frequency Modulation with the help of "Fig -a" shown below: -

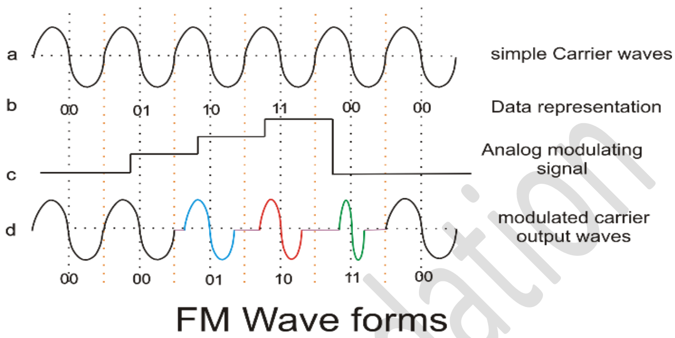


Fig -a

For better visibility of fine differences in timing, we have drawn the wave form with unrealistic frequency change shown which in practice would be few Hz only. Another point in waveform is that we have sampled the analogue wave form at fixed intervals and generated one sample for each level change in modulating signal for easy timing corelation.

For FM implementation in **"Z-Mod"** system, a set of predefined frequencies is decided in advance (can be few Hz.) and each directly generated modulated cycle, frequency Jumps from one frequency to another pre-set frequency. Selection of which step frequency cycle to be generated is decided in accordance with the modulating signal sample. As each modulated carrier cycle is generated, starting from zero crossing point with pure sine function and ends at another zero-crossing point, there is no abrupt change in energy/frequency and in the process no side bands are generated.

Because of Frequency being the selected parameter for modulation change, it will need predefined band of fixed frequencies, occupying a small spectrum with pre-defined carrier frequencies. Ther is no generation of side bands in **“Z-Mod”**, which are proportionate to modulating signal in conventional modulation.

Phase Modulation Implementation: -

Next, we can see how **"Z-Mod"** principals can be applied to Phase Modulation and figure below presents one of its implementations: -

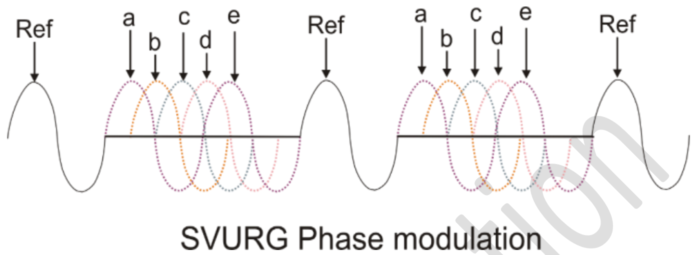


Figure b

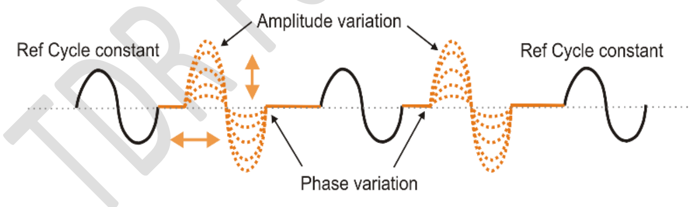
"Figure b" shows a simple phase modulation waveform based on 3 cycles frame, having 1) ref cycle, 2) data cycle and 3) Zero cycle. The data cycles shown from 'a' to 'e' are for illustration only and only any one carrier sine cycle will be present in practice in any given frame. Here presence of patented Zero Voltage Cycle (ZVC) is important to analyse phase difference in terms of time difference at start of carrier sine cycle. In "Z-Mod" no frequency change is taking place in the process. Another unique concept in "Z-Mod" process is Zero Voltage Cycles (ZVC) which are zero energy cycles providing convenient timing adjustment for actual pure sine wave cycles generation to be able to start and end at zero crossing points only wherever needed.

The concept is very simple, where ref cycle is repeated at beginning of every frame, setting precise zero-phase reference for the frame. Remaining two cycle period is shared between Data cycle and Zero Voltage cycle with data cycle varying its start time from ‘a’ to ‘e’ as shown in figure. For position other than ‘a’ to ‘e’ of data cycle the zero-voltage cycle is split in two parts. This split can be any proportion from zero to 100% of the cycle, depending only on the modulating signal amplitude sample for the frame. So, these zero-voltage cycle also contribute indirectly to carrying the information as it is the duration of this zero-voltage cycle which represents the data length/value.

This same concept can also be implemented as On/Off Keying or OOK with a difference that switching is to be done only at the zero crossing point of pure sine wave cycle.

QAM and FDM Implementation: -

In further developments of the same “Z-Mod” principle, it can be applied to various combinations of AM, PM & FM modulation types. To start with consider AM and PM which is the basis of QAM modulation system using example waveform as shown by the wave form below: -



SVURG DOUBLE modulation wave forms

Here the wave form for PM has been modified to apply amplitude variation and phase variation simultaneously to the data cycle, without changing any of the sine function properties of carrier cycle during generation process of each cycle. As required by the **"Z-Mod"** process all the sine wave properties of frame constituent cycles have been retained and 'cycle-by-cycle' step process applied.

In the wave form there are two different sets of signals/data can be carried by data cycle in each frame. One set of data is delivered by the phase variation of the data cycle w. r. to reference cycle. Second set of data can be delivered by the amplitude change applied to the same data cycle in every frame.

This combination and many other will provide unimaginable data capacities in "Z-Mod" systems. To further increase the data capacity of systems FDM with much closer frequencies can be applied as there are no side bands FDM gaps can theoretically be less than 5 Hz apart and depends only on hardware capabilities to resolve and select fine frequencies.

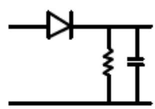
Reception Side Explained

To establish how **"Z-Mod"** principal can recover huge amounts of information from pure sine wave cycles, we start with comparing the various AM envelop detector for detecting conventional and **"Z-Mod"** AM signals for simplicity.

Simple Envelope Detector

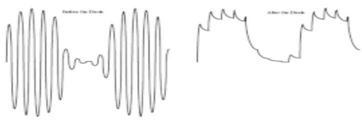
An envelope detector is a half wave rectifier followed by a low pass filter to reproduce demodulated signals optimally. In the case of commercial AM radio receivers, the detector is placed after heterodyne process, at this point IF frequency is 455 kHz while the maximum audio base-band frequency is only 5 kHz. Since the ripple component is nearly 100 times the frequency of the highest baseband signal it does not pass through any subsequent audio amplifiers.

Because it was believed the information is in side-bands and side bands are in proportion to modulating frequency and needing twice the highest frequency of modulating signal. Due to this belief high bandwidth detectors were never imagined for 455 KHz. IF frequency. To explain importance of filter in demodulation of Fc ± Fm in the conventional AM (and only Fc in **"Z-Mod"** AM) basic detector circuit with inadequate Time constant will be used.



The amplitude information of modulated envelope is rectified by a diode, followed by a filter as shown in figure, and in practice Fc/Fm ratio is quite large. Quality of received signal depends on filter as rectified diode charges the filter capacitor to peak value of each IF cycle to average the peak over large no of cycles.

To establish the fact that only less than 10% of sine cycle at peak of each cycle contributes to output, an inadequately filtered waveform shows the



drooping effect on the output. Wave form shown is of an AM signal where the carrier frequency shown is only 10 times the audio signal frequency, which has considerable ripple in output. However, modern SineX/X filters recover modulating signal up to 1/3rd ratio of Fc/Fm.

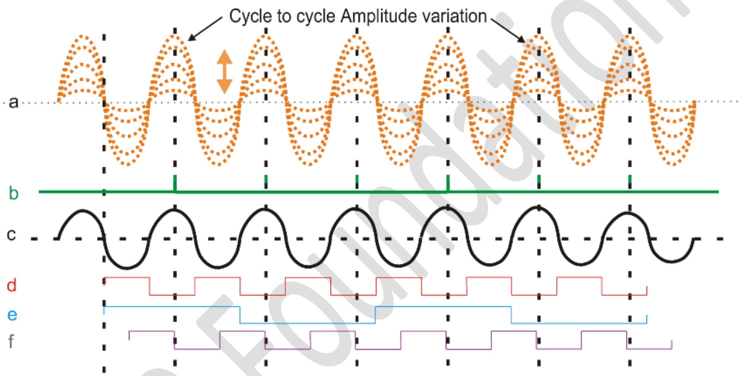
"Z-Mod" Reception Concept (AM)

However, for high bandwidth high quality detection a synchronous detector can provide astonishing results for **"Z-Mod"** as explained below: -

Fact that **"Z-Mod"** AM system is capable of carrying high bandwidth signals, it utilises a precision sampling detector to sample each carrier cycle at peak of each cycle. Using a sample and hold circuit for detecting peaks of each carrier cycle in **"Z-Mod"** provides enormous bandwidth capacity as per **NYQUIST THEORM**. The filter for **"Z-Mod"** detector has to resolve very high signal bandwidth or can be converted to digital byte at its carrier frequency, taking maximum advantage of the **"Z-Mod"** concept.

Going further, as there are no side bands in our modulated signal, each of our carrier cycle peaks are consistent in time, for reliable sampling of peak amplitude at carrier frequency or frame rate is possible. High bandwidth original analogue signal, up to one third of carrier frequency, can be efficiently reconstructed as per Nyquist theorem. This means practically large bandwidth signal up to one third of carrier frequency or more can be carried and detected by **"Z-Mod"** technique.

A sysnchronus sampling detector is shown below:-



AM Synchronous Detector

The AM Detector shown in figure above is a synchronous detector. It can deliver accurate and consistent peak amplitude samples from each cycle of the **"Z-Mod"** modulated AM implementation. "**trace a**" is the received modulated signal and waveform "**trace b**" in green is a sampling pulse synchronised to received signal cycle peak. "**trace c**" is the phase locked oscillator output locked to received carrier frequency, leading to generation of square waves as "**trace d**", and resulting quadrature waveform at "**trace f**". And "**trace e**" is for frame based implementation for ultra-high reliability systems. As all these waves are regenerated phase locked to carrier frequency, the accuracy of detected peak level, as per sample pulse of "**trace b**" can reproduce finest peak amplitude values. These sampled signals can be used to recreate original signal waveforms with appropriate filtering. This type of detector approach can resolve output signals bandwidths satisfying, **Nyquist theorem.**

"Z-Mod" modulation technique is a method to eliminate role of side bands to carry the information and capacity enhancement at the same time which is equally effective in all types of modulation systems known today and it opens new opportunities in all fields of communication.

Phase Modulation 'PM'

Conventional phase modulation resembles frequency modulation in many terms and variation of frequency can not be separated from phase modulation process. As the phase modulation changes phase from one set of phases to another in a defined time period progressively and in the process, frequency will continuously change depending upon the amount of phase change and the time available for the same. On reaching the target phase the frequency may become stable as long the phase change is not started again.

Simple phase modulation like quadrature modulation & others are not very popular options for communications. Whereas QAM is the most popular existing use of phase modulation, where amplitude and phase variations together deliver large amount of data when it is combined with frequency division multiplexing or FDM.

To elaborate further the difference in invented **"Z-Mod"** process, a brief review of theory on the existing modulation process will help establish fine differences in two approaches to the reader. For reason of easy clarity, we will start from the conventional phase modulation example and then elaborate both the process for reference.

Existing Phase Modulation (reference)

PM changes the phase angle of the complex envelope in direct proportion to the message signal.

If *m(t)* is the message signal to be transmitted and c(t) the carrier onto which the signal is modulated is expressed by equations: -

$$c(t) = A_c \sin(\omega_c t + \emptyset_c)$$

then the modulated signal will be

$$c(t) = A_c \sin(\omega_c t + m(t) + \emptyset_c)$$

This shows how m(t) modulates the phase, the greater m(t) is at a point in time, the greater the phase shift of the modulated signal at that point. It can also be viewed as a change of frequency of the carrier signal, and phase modulation can thus be considered a special case of FM, in which the carrier frequency modulation is given by the time derivative of the phase modulation.

The modulation signal could here be

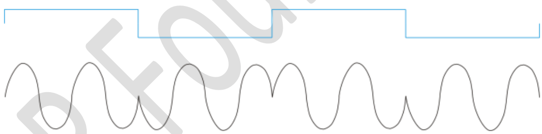
$$m(t) = \cos[(\omega_c t + h\omega_m(t)]$$

The mathematical analysis of the spectral behavior reveals that there are two regions of particular interest:

- For small amplitude signals, PM can be similar to amplitude modulation (AM) and exhibits its unfortunate doubling of baseband bandwidth and poor spectrum efficiency.
- For a single large sinusoidal signal, PM is similar to FM, and its bandwidth is expressed approximately by: -

$$2(h + 1)f_m$$

where $f_M = \omega_m/2\pi$ and h is the modulation index. This is also known as Carson's Rule for PM.



Typical Digital Phase modulation waveform

The above figure shows a typical phase modulation waveform with phase reversing for the digital value change. This wave form is for synchronous digital data with the carrier frequency and is chosen as it shows the change in phase clearly with 180° phase change and is easy to notice. In the above figure it is apparent to reader that phase changes by 180° for data value change between “1” and “0”. This wave form explains the concept of relative phase of the cycles also as apparent the change occurs with reference to previous cycle. For further information reader can refer to many textbooks available on the topic.

ABOUT "Z-Mod" TECHNIQUE PM

Now we move to analyse the **"Z-Mod"** technique and how it works with the help of typical wave forms produced by **"Z-Mod"** PM modulation system, in one of many possible configurations. The chosen setup has been depicted in terms of time and there are two main time periods marked as **"T"** and **"2T"** which are repeated in sequence. Sum of these two periods is the **frame time** marked as **"TF"**, which can also be called sample rate for analogue signals in this implementation. In case of digital signals it acts as a frame rate and the frame can be of any time period and accordingly its time can be higher multiple of **"T"** and accordingly depending on the resolution capacity of the design it can accommodate any finite set of data bits.

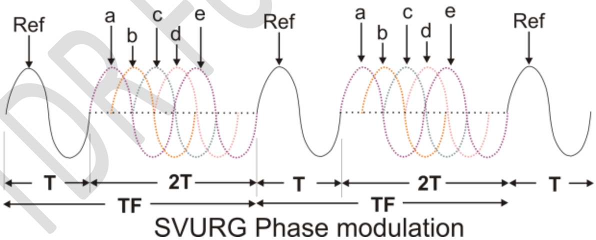
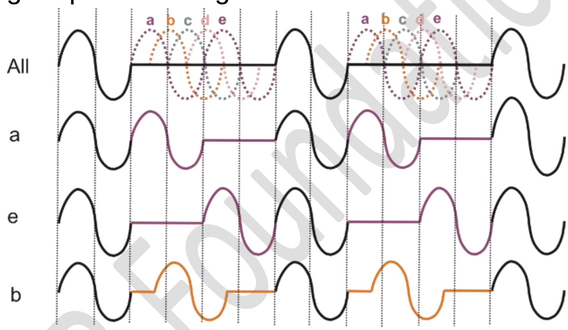


Fig 1

The above wave form depicts a total of 13 individual sine wave cycles shown with combination of solid black lines and dotted coloured lines. The solid line sine wave cycles marked as 'Ref', at fixed time interval "TF", consisting of time period equal to three complete cycle of pure carrier frequency. Ref cycle is shown at three places, generated in a cycle-by-cycle process. The 5 dotted lines sine wave cycles shown from 'a' to 'e' and repeated twice are to show five relative phase positions of data cycle, and in practice, only one sine wave cycle out of these 5 sine cycles, will exist in the dotted line portion marked as 2T. All sine wave carrier cycles are generated starting at perfect zero crossing point and ending at another perfect zero crossing point. They are shown here with cycle starting with positive amplitude direction but can be starting in negative amplitude direction as well. All the 13 cycles shown have same fixed carrier frequency and they all start from zero crossing point, which means they all start from zero energy point and end at zero energy point ensuring there is no abrupt change in energy level or in sine function for each carrier wave cycles. This cycle-by-cycle generation of carrier sine wave cycles with starting and ending at zero crossing point is the key factor to observe, which we explain further in coming paragraphs.

The 13 cycles depicted are governed by three critical time functions. One is reference sine wave cycle mentioned as "Ref" with time period "T" and repeated at 'TF' time period equal to 3 cycle times. second being the data sine wave cycles shown as **"a, b, c, d, e"** scattered in data time period of "2T". In addition, there is third invisible **FRAME RATE** "TF" which is the sum of reference time T + data time 2T = 3T. All the sine cycles are controlled to start from zero crossing point of sine wave and data cycle is placed between two reference cycles also at carrier frequency which is repeated at frame rate. In the time period between two Ref cycles there is one data cycle out of shown 5 dotted line sine wave cycles, which is identical to "Ref" cycle with a difference that they have different staring phase angles, changing by 90° from one to next, in available 0° to 360° available blank data period with respect to "Ref" cycle. In other words they show time difference in starting phase of the cycle. In practice there will be only one data cycle between any two consecutive "Ref" cycles and the 5 sine wave cycles labelled 'a to e' with different colours, are hypothetically selected locations at phase angles of 0°,90°,180°,270° and 360°. They show only five possible positions out of thousands of possible locations with fraction of a degree phase changes.



SVURG phase modulation wave forms

Fig 2

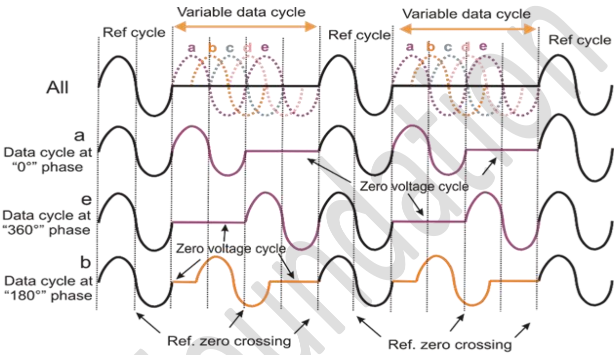
The above figure 2 redraws with actual waveforms cases **"a", "e"**, and **"b"** with top wave form marked **"All"** is similar to fig 1, shown for reference. **"Fig 2-a"** is showing the data cycle beginning is coincident with end of **"Ref"** cycle, in phase with ref cycle and after completion of sine cycle 'a' there is no signal for one complete carrier cycle period (invented zero voltage cycle) and it is then repeated with ref and further data cycles. This representation of 0° phase shift carrying information which is the lowest value or ZERO, while the data cycle shown in **"Fig 2-e"** is with 360° phase shift means data values are maximum. **"Fig 2-b"** shows position of data cycle with respect to reference cycle at 90° phase shift, representing 25% of maximum value of signal.

The consideration for using sample rate to be equal to one third of carrier frequency is selected for establishing concept only, as it provides a convenient way for understanding. And any suitable ratio of sample rate with any finite number of cycles at carrier frequency can be selected according to application needs, where there is only one ref cycle needed per frame. For larger data bytes to be covered in one frame either we need to resolve time period to much finer details or increase the frame time from 3T to 4T …. NT.

Practical Considerations

Phase modulation implementation in **"Z-Mod"** is radically different from conventional PM because spectrum bandwidth used is less than 1 Hz, and there is no possibility of frequency of the carrier to change. As the carrier sine wave cycles are generated in a cycle-by-cycle process and **"Z-Mod"** process uses patented "**zero voltage cycles**" to carry data, particularly in phase modulation. The cycle-by-cycle, direct carrier generation process makes it distinctly independent from conventional phase modulation as there are no side bands generated in **"Z-Mod"** process. Another advantage in **"Z-Mod"** process is its independent enhanced capacity to carry large signals, as it interprets phase difference in terms of time difference. This concept of using zero voltage cycles, based on timing difference, exploits the mature advanced technology, in detecting time resolutions to fractions of a Pico-Second. **"Z-Mod"** process makes phase modulation very simple and efficient to carry very large amounts of data in terms of time differences. As higher and higher time resolution capabilities are being developed, it can reach theoretical limits.

To elaborate the **“Z-Mod”** technology, we will use frame-based concept, as this configuration is the easiest one to understand and is one of many possible configurations.



SVURG phase modulation wave forms

The figure above shows a frame-based phase modulation implementation where frame rate is selected to be equal to 3 carrier frequency cycles combined. Each frame consists of three cycle time period, have a fixed phase, sine wave cycle as reference, another sine wave cycle called data cycle, and third cycle time period is filled by a zero-voltage cycle, which can be split in to two parts also. First cycle shown in solid black line in upper wave form marked ‘All’ is the reference cycle starting only at zero crossing point and ending at another zero-crossing point after completing the complete 0˚ to 360˚ cycle. This cycle does not change in time, amplitude, phase or frequency and is repeated in each frame, at frame rate which is ⅓ of carrier frequency.

Shown next to frame are 5 cycles in coloured dotted lines, in the wave form, representing arbitrarily selected data cycle at 5 out of millions of possible positions in any one frame. These possible million positions become the million data points in every one frame only dependent on hardware capability to resolve time differences. Next three rows in waveform marked ‘a’, ‘e’ & ‘b’ represent data cycle at 0°, 360° and 90° phase shift, marked ‘a’, ‘e’ & ‘b’ respectively. Point to note here is the time gap between reference cycle and data cycle can change in proportion to the modulating signal value and the unique **‘zero-voltage cycle’** can be split in two parts, without effecting the purity of each carrier cycle’s sine properties.

It is essential to understand the role of **‘zero-voltage cycle’** which are unique part of the invented process. **‘zero-voltage cycles’** are best understood as gaps of zero energy time periods between the carrier cycles. These zero energy cycles have no frequency components associated, or have zero frequency. **‘zero-voltage cycle’** only have time period defined dynamically according to system needs. <u>*In other words, addition, multiplication or any other processing does not effect the frequency, phase or amplitude of the associated carrier cycles*</u>. This unique property has been used in the invention to carry data in the **“Z-Mod”** process.

It is worth to review the resultant wave forms, from a Fourier transform perspective which will distribute the energy of pure carrier cycles in to other components not present in the output of modulated carrier cycles. The reason that Fourier transform results are dependent on the number of samples processed to drive the results and can neither recognise cycle by cycle process or analyse the frame-based scheme.

Calculating Data Capacity

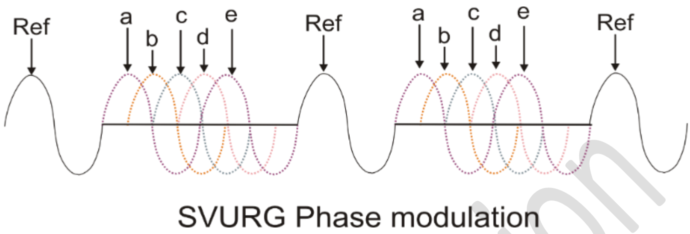


Fig-2

For establishing quantum of data delivered by **"Z-Mod"** method, we will use a practical example with nominal frequency of 1.5 MHz or 1500 KHz and will use the frame rate using 3 cycles. These two numbers are randomly selected for ease of calculation and much higher data capacities can be obtained with other configurations.

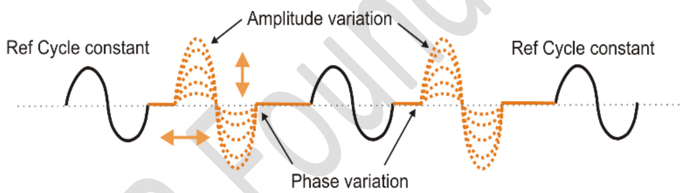
Phase info to Data Calculations

Using above mentioned practical example of 1.5 MHz carrier frequency. And our selected frame rate to be one third of 1500 Khz resulting in 500 KHz makes time period of 2 microseconds per frame. At 1.5 MHz the "**T**" becomes ⅔ rd. of a microseconds which equals to 0.666666 microseconds. This is equal to time period required by one sine wave cycle to complete. Now data cycle can move from position "fig2-a" at 0° starting phase to "Fig-2-e" of 360° phase shift, which can provide a total variation of 666.66 Nano seconds. With current hardware technology, time resolution of sub–Pico seconds can be resolved reliably, at low costs. By using very conservative,10 Pico second resolution, each frame can resolve 66666 steps, which can easily accommodate a 16-bit Byte data needing 65536 steps. As the frame rate is 500 KHz the available 16-bit data byte @500 K sample rate or a total data through put of 8 mbps (16x500K=8000000) using just the pure carrier frequency of 1.5 MHz without consuming any spectrum bandwidth in side bands. This data through put can easily carry 2 or more independent AES-EBU uncompressed digital stereo audio channels at 48KHZ sample rate with scope for additional data for error correction.

This calculation gets multiplied by 10 times, the moment we raise our resolution to 1 Pico seconds with enough data to accommodate some high-quality HD Sports TV channels using minor compression technique. Current technology used in GPS receivers resolves time resolutions to 0.01 Pico seconds raising possibility of another 100 times increase in data capacity by just using the less than 1 Hz bandwidth at only 1500 KHz carrier frequency.

Combining Phase with Amplitude

So far, we have only taken the phase variations of the "data cycle" and saw 8 mbps to 80 mbps data transmission is feasible with present level of technology. In next step if we can add amplitude variations to the data cycle, along with phase variation then we can get another independent set of data, derived from the amplitude changes in the data cycle. We cannot ignore the fact that most phase detectors will lose accuracy when amplitude drops below a reasonable minimum level and precautions to balance the two is essential. Figure below explains the concept in the most simplistic manner.



SVURG DOUBLE modulation wave forms

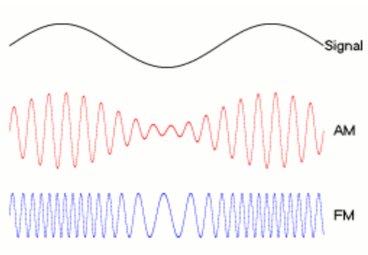
Fig-3

Using above Fig-3 we assume that phase data can be decoded from data cycle without compromising accuracy, with a conservative minimum amplitude ratio at 30% of "Ref cycle" amplitude and conservatively we can accurately resolve amplitude resolution to ≤ 0.01 % accuracy. This available amplitude variation of 70% leaving 30%, can produce around 7000 level steps resolved. For simplicity of calculation, if we use 4048 steps for data, they can provide additional 12 data bits independently per frame, at the sample rate of 500 KHz, with carrier frequency of 1.5 MHz This amounts to carrying another 6 mbps data, using the same 1.5 MHz carrier frequency, without need of any spectrum bandwidth and only using the pure carrier frequency with zero side bands.

This 0.01% amplitude resolution is very conservative achievable even with discrete components and with integrated circuits it can be cost effectively resolved to 0.001% amplitude resolutions or better, for still 10 times more data capacities. Present claims of data capacities are derived from a design based on existing commercial parts and not optimised for dedicated purpose-built semiconductors for the invented method. With more dedicated semiconductor developments these capacities have scope for further increasing by another 100 times.

Frequency Modulation 'FM'

Frequency modulation (**FM**) is the process of encoding information in a carrier wave by varying its instantaneous frequency of the sine wave. In analogue modulation, such as FM, radio broadcasting of audio signal representing voice or music, in difference between instantaneous frequency deviation. The daviation in instant frequency of the carrier and its centre frequency, is proportional to the modulating signal amplitude.



Similarly digital data can be encoded and transmitted via FM by shifting the carrier's frequency among a predefined set of frequencies representing digits – for example one frequency can represent a binary 1 and a second can represent binary 0. This modulation technique is known as frequency-shift keying (FSK).

Frequency modulation technique is widely used for FM radio broadcasting. In radio transmission, an advantage of frequency modulation is that it has a higher signal-to-noise ratio of the audio, and therefore rejects radio frequency interference better than an equal power amplitude modulation (AM) signal. For this reason, most music is broadcast over FM radio.

Frequency modulation and phase modulation are considered the two complementary principal methods of angle modulation, phase modulation is often used as an intermediate step to achieve frequency modulation. These methods contrast with amplitude modulation, in which the amplitude of the carrier wave varies, while the frequency and phase are not changed.

FM Working in details (conventional)

A very useful rule of thumb used by many engineers to determine the bandwidth of an FM signal for radio broadcast and radio communications systems is known as Carson's Rule. This rule states that 98% of the signal power is contained within a bandwidth equal to the deviation frequency, plus the modulation frequency multiplied by two. Carson's Rule can be expressed simply as a formula:

$$BT = 2\left(\Delta f + fm\right) \qquad \ldots\ldots \text{(a)}$$

Where:

Δf = deviation

BT = total bandwidth (for 98% power)

fm = modulating frequency

And the actual frequency at any moment of time can be derived from: -

$$F_t = f_c + 2\left(\Delta f + fm\right) \qquad \ldots\ldots\ldots\text{(b)}$$

Where:

f_c = Carrier frequency with no signal

F_t = instantaneous frequency

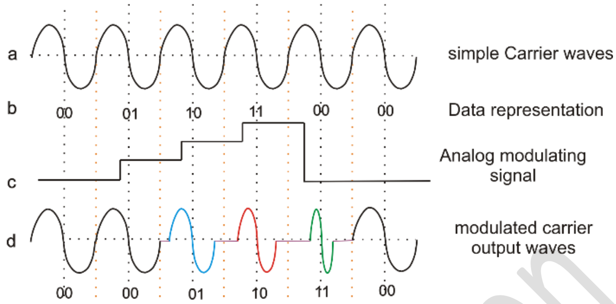
To take the example of a typical broadcast FM signal that has a deviation of ±75kHz and a maximum modulation frequency of 15 kHz, the bandwidth of 98% of the power approximates to 2 x (75 + 15) = 180kHz. To provide conveniently spaced FM channels 200 kHz is allowed for each station (for mono signals). However, for FM broadcasts of today where stereo audio is standard with a signal multiplex of stereo audio with 19 KHz subcarrier requires 39 KHz bandwidth for audio, making minimum bandwidth requirement, 2x(75 + 38) = 216KHz. Whereas all FM stations in a area are spaced 400 KHz or more apart to accommodate additional data etc.

With a fixed tone-modulated FM wave, if the modulation frequency is held constant and the modulation index is increased, the (non-negligible) bandwidth of the FM signal increases but the spacing between spectra remains the same; some spectral components decrease in strength as others increase. If the frequency deviation is held constant and the modulation frequency increased, the spacing between spectra increases.

The rule is also very useful when determining the bandwidth of many two-way radio communications systems. These two way systems use narrow band FM, and it is particularly important that the sidebands do not cause interference to adjacent channels that may be occupied by other users.

Z-Mod Concept FM

"Z-Mod" concept of zero bandwidth communication system having no side bands is an improved Frequency Modulation process, by eliminating multiplication of carrier waves with modulating signal in real time and the modulator module itself. Instead, it uses direct generation of modulated pure sine wave carrier cycle, in a cycle-by-cycle step, starting at 0° and end at 360° phase angle of complete cycle, each start and end at respective zero crossing point.



FM Wave forms

The above figure shows three waveforms the first “a” is pure carrier wave without any modulation. Second “b” shows two-bit data in bit map and “c” shows equivalent analogue representation of the data value with changed level. The wave form “d” shows the resultant modulated sine wave cycles separated by blank spaces or zero voltage cycles. The generated output sine wave cycles are shown with very big difference in frequency for easy visibility of change for understanding and in practice the zero voltage cycles will be very narrow and frequency change may not be visible on waveform view and will depend on designer’s choices according to design needs.

The point to note is that in **“Z-Mod”** the frequency of carrier cycles jumps from unmodulated Carrier frequency to higher frequency which is predefined by design. Between any predefined frequencies other than the base carrier frequency there is a zero-voltage cycle with zero energy is used to fill time gap, between two jumping frequencies. Here as the zero-voltage cycle does not have any frequency of its own to add any frequency component in the output. It provides a convenient way to achieve the target frequency shift without generating side bands. The jump in frequency will totally depend on the kind of resolution the system can resolve practically.

The FM bandwidth equation in **“Z-Mod”** will change completely, as we only have few carrier frequencies predefined within a range, in the output. As the frequency is not deviating from carrier but jumping from one frequency to another frequency, the output does not have any multiplication products of the carrier and signal frequency.

As our process only changes from no signal, base carrier frequency to higher frequency only, the multiplier factor of 2 goes away. So as a first step the equation (1) can be re-written as: -

$$BT = \left(\Delta f + fm \right)$$

Secondly as the change in value of modulating signal is static, each cycle the “(Δf + fm)” get modified to “Δf” only as there is no frequency component of modulating signal is in the output. So, the equation: -

$$\mathbf{BT = 2\ (\ \Delta f + fm\ )}$$

Becomes $$\mathbf{BT = 1\ (\ \Delta f + 0\ ) \quad = \Delta f \qquad \ldots\ldots(c)}$$

Which is the predefined frequency change jump value as per design. Thus, within the design parameters the output carrier frequency will remain pure sine wave within the well-defined stepped frequencies and not generate any intermodulation products appearing in the output as side bands.

And modulated output instantaneous frequency becomes: -

$$F_t = f_{c-1}, = f_{c-2}, = f_{c-3}, \; \text{- -} \; = f_{c-n},$$

Where:

$$f_{c-1}, f_{c-2}, f_{c-3}, \; \text{- -} \; f_{c-n},$$

are predefined carrier frequencies.

Frequencies fc-1 through fc-n are fixed static frequencies for each cycle and not having any effect of modulating signal frequency.

In **"Z-Mod"** FM process the frequency of each pure sine wave carrier cycle so generated from 0° to 360° changing its frequency vector at respective zero crossing, will change (jump) to another predefined carrier frequency from one cycle to another, based on static value of modulating signal for the duration of that particular carrier cycle. Each carrier cycle being generated is either a pure sine wave cycle or has a zero-voltage cycle between them, thus not generating any side bands and using limited spectrum of the predefined carriers.

For example, if predefined carrier frequency range is from 100,000,000 Hz to 100,005,000 Hz and step size is 5 Hz it can easily resolve 1000 steps of analogue signal voltage with 0.1% resolution accuracy along with up to

40,000,000 Hz signal bandwidth as per **Nyquist theorem** or carry almost 10-bit data word within each cycle at 100 MHz sample rate using just **pure sine wave carriers** without any side bands. Here the point to note is that step change in frequency modulated output from one cycle to next could be anything between 0 to 100% within design predefined frequencies, restraining output spectrum to remain within specified band width without any addition of side bands.

If we compare this to conventional FM radio transmission, which uses more than **200 KHz spectrum** to carry one stereo audio with **15 KHz signal band width** at the same 100,000,000 Hz frequency. Whereas in patented **SVURG/Z-Mod** process it needs less than **5 KHz bandwidth** resulting in 40 times less spectrum and carry signal bandwidth more than **40 MHz** making it to carry **1000 times higher bandwidth** signals in a simple straight comparison.

In the patented **"Z-Mod"** process, starting and ending of pure sine wave cycle generation has to be precisely and firmly bonded to zero crossing point of every cycle, at this point the instantaneous energy in cycle is zero. In a cycle-by-cycle process when we change a vector of "next pure sine wave cycle to be generated" at a point when it is at zero voltage point, the energy is not going to change at all, and as a result output will still remain zero, hence output does not have any change. As the cycle starts with new defined sine function value of changed vector, frequency in this case, it will generate the next carrier wave cycle as a pure sine wave cycle till it ends at another zero-crossing point.

It is this cycle-by-cycle controlled process for starting and ending of **“pure sine wave cycle generation”,** for each complete carrier sine wave cycle in combination with zero voltage cycles, directly generate modulated sinusoidal carrier waves with zero side bands. This process completely avoids the separate generation of carrier frequency and multiplication between carrier and modulating signal which produces side bands in the conventional process. The conventional carrier oscillator and modulator module, is also eliminated, as there is no multiplier process in this invention.

RADIATED POWER ADVANTAGE 'SVURG'

So far, we have seen the **"Z-Mod"** advantages in terms of spectrum savings which are big enough to make it desirable. **"Z-Mod"** technology provides another huge advantage in terms of reduced power levels for covering the same area. These features provide valuations which have exceptional commercial advantages, which are to be described in separate document and we will consider simplest understandable perspective in term of power savings. For this we need to understand Minimum Detectable Signal level first which is explained below using published documents which can be referred below:-

https://en.wikipedia.org/wiki/Minimum_detectable_signal

Minimum detectable signal

A **minimum detectable signal** is a value of signal, at the input of a system, whose power produces a signal-to-noise ratio of "**m**" at the output. In practice, "**m**" is usually chosen to be greater than unity. In some documents, the name *sensitivity* is also used for this concept.

General

In general, it can be stated that for any receiver to "see" a signal it must be greater than the noise floor. In practice to detect the signal, it is often required to be at a power level greater than the noise floor by an amount that is dependent on the type of detection used as well as other factors. There are exceptions to this requirement, but coverage of these exceptional cases is outside the scope of this document. This required difference in power levels of the signal and the noise floor is known as the signal to noise ratio (SNR). To establish the minimum detectable signal (MDS) of a receiver we require several factors to be known.

- Required signal-to-noise ratio (SNR) in dB
- Detection bandwidth (BW) in Hz
- Temperature of the receiver system T_0 in kelvins
- Boltzmann's constant $k = 1.38\times10^{-23}$ joules per kelvin
- Receiver noise figure (NF) in dB

To calculate the minimum detectable signal, we first need to establish the noise floor in the receiver by the following equation: -

$$\text{Noise floor}_{\text{dBm}} = 10\log_{10}(k \times T_0 \times 1000) + \text{NF} + 10\log_{10}\text{BW}.$$

Calculating noise floor for 1 Hz bandwidth for a receiver with 1.5 dB noise figure for the receiver we get following

Noise floor (dBm)= 10log($1.38\times10^{-23}\times290\text{x}1\times10^{3}$) +1.5 +10log(1)= -174 dBm

As a real numerical example:

A receiver has a bandwidth of 10 MHz and noise figure of 1.5 dB at the physical temperature of the system is 290˚ kelvins.

Noise floor(dBm)= 10log ($1.38\times10^{-23}\times290$ x 1×10^{3})+1.5+10log(10×10^{6})

= −174 + 1.5 + 70(dBm)

$= -102.5$ (dBm) .... (i)

So, for this receiver to even begin to "see" a signal it would need signal to be greater than −102.5 dBm. Confusion can arise because the level calculated above is also sometimes called the Minimum Discernable Signal (MDS). For the sake of clarity, we will refer to this as the noise floor of the receiver. The next step is to take into account the SNR required for the type of detection we are using. If we need the signal to be 10 times more powerful than the noise floor the required SNR would be 10 db. To calculate the actual minimum detectable signal is simply a case of adding the required SNR in dB to the noise floor. So, for the example above this would mean that the minimum detectable signal is:

MDS (dBm) = -102.5 + 10 = -92.5 (dBm) ....(ii)

MDS (dBm) = 10Log $(1.38\times10^{-23}\times290 * 1000)$ + NF + 10Log (BW) + SNR (dB)

In this equation:

kT_0 is the available noise power in a bandwidth BW = 1 Hz at T_0, expressed in watts. kT_0 x 1000 is the available noise power in a bandwidth BW = 1 Hz at T_0, expressed in milli watts. T_0 is the system temperature in kelvins, and **k** is Boltzmann's constant is (1.38×10^{-23} joules per kelvin = −228 dBW/(K·Hz)).

Putting these values in equation we get Noise floor @ 0˚C

Noise floor (dBm) = 10Log $(1.38\times10^{-23}\times290 * 1000)$ = -174 dbm.

If the system temperature and bandwidth is 290° K and 1 Hz, then the effective noise power available in 1 Hz bandwidth from a source is −174 dBm (174 dB below the one milli watt level taken as a reference).

1 Hz noise floor: calculating the noise power available in a one hertz bandwidth at a temperature of $T = 290°$ K (0°C) defines a figure from which all other values can be obtained (different bandwidths, temperatures). 1 Hz noise floor equates to a noise power of −174 dBm so a 1 kHz bandwidth would generate $-174 + 10 \log_{10}(1\ \text{kHz}) = -144$dBm of noise power (the noise is thermal noise, Johnson noise).

MDS (dBm) = 10Log(kTo*1e3) + NF + 10Log (BW) + SNR (dBm) …..(iii)

The equation above indicates several ways in which the minimum detectable signal of a receiver can be improved. If one assumes that the bandwidth and SNR are fixed however by the application, then one way of improving MDS is by lowering the receiver's physical temperature. This lowers the NF of the receiver by reducing the internal thermally produced noise. These types of receivers are referred to as Cryogenic Receivers.

MDS Calculations for Z-Mod

Using these equations "Z-Mod" technique theoretically needs less than 1 Hz bandwidth (or fraction of it if possible). Let us presume current technology cannot resolve 1 Hz bandwidth of filters reliably, so we take practical approach with 20 times this at 20 Hz bandwidth of filters at the point of detection. For 20 Hz bandwidth of received signal the MDS will be 13 dB higher from -174 dBm and become -161 dBm. For all practical existing communication systems generally

8 to 10 dB carrier to noise is considered adequate for digital signals, let us be liberal again and base our calculations with additional 10 dB SNR making it 20 db. With this margin our MDS becomes -141 dBm, which is 58.5 dB better than earlier calculation of 82.5 dBm (for a 100MHz bandwidth) referred at ………..(ii).

Now let us calculate for practical case of analog AM broadcast service at carrier frequency at 1.5 MHz, which in existing practice needs minimum audio SNR of 40 dB and carries a signal of 5 KHz bandwidth with a channel bandwidth of 10 KHz.

Putting these parameters in equation (iii): -

MDS (old) = 10Log (kTo*1e3) + NF + 10 Log (BW) + SNR (dBm)

MDS (old) = -174 + 1.5 + 40 + 40 (dBm) = -92.5 dBm ……(iv)

And compare conventional system results with **"Z-Mod"** technique Equation (iv) with **"Z-Mod"** system becomes: -

MDS (Z-Mod) = -174 + 1.5 + 13 + 40 (dBm) = -119.5 dBm …(iv)

This calculation shows a direct advantage of 26.5 dB over conventional systems without taking in consideration other advantages in comparison. This 26.5 dB difference, in MDS, can be interpreted in terms of radiated power requirement, reduced to almost 0.2% to maintain same SNR of 40 dB. Or instead of needing 500 Watts of radiated energy we will only need 1 watt radiated energy in **"Z-Mod"** system. This is a huge advantage in term of required RF power reduction making it a green technology. **"Z-Mod"** process offers many more advantages in addition, not factored in our calculations.

We should not overlook other advantages of **"Z-Mod"** technique, which is the fact that our RF received signal passes through a narrow pass band of 20 Hz. So, the detector does not see any other signal other than the carrier and detected signal cannot have noise density bandwidth above 20 Hz (or so), defined by accuracy of the filters. But **"Z-Mod"** system can easily carry baseband analog signal bandwidth in the range of Fc/3 or 500/3 = 166.6 KHz, or more as per Nyquist theorem. This means we can do away with the requirement of 40 dB SNR of received base band signal to 20 db. When we put this value in equation ……(iv)

MDS (SVURG) = -174 + 1.5 + 13 + 20 (dBm) = -139.5 dBm …(iv)

Or the carrier to noise ratio advantage with conventional modulation becomes 46.5 dB.

This when interpreted in required power for same coverage area, our radiated power requirement reduces to **10 mw** in comparison to conventional RF power of **500 Watt** (a reduction of total 46 dB from conventional approach). This correlation is based on direct scientific and mathematical calculations in a worst-case scenario with margins for further improvements and further practical trials is expected to exceed these calculations by another 10 to 20 dB more.

Implied Advantages in Z-Mod

So far, we have evaluated and compared the direct advantages of **"Z-Mod"** technology in terms of communication channel bandwidth reduction, of the carrier, reduction in effective radiated power requirements and tremendous increase in signal bandwidth all three available concurrently at the same time. Now we will be looking at the implied advantages of the **"Z-Mod"** technology in term of signal quality improvements of received signal. To explain and understand these, we have to go a bit farther from the detector output point in the signal chain.

Noise Floor in Received Baseband

At the detector input, **"Z-Mod"** technology requires less than 1Hz carrier bandwidth (currently not possible with existing components) and with future developments in semiconductors for narrow band signal handling, it will be possible to detect signals from carrier waves which are almost noise free having a noise floor in the range of -150 dBm or better. This ultra-narrow bandwidth of 1 Hz at front end ensures that only carrier signal and unwanted noise only within 1 Hz of carrier frequency, is presented to detector. Output at detector can only recover signal from carrier frequency components and probable noise within bandwidth of 1 Hz around the received carrier frequency. Filter following the detector is optimized to filter out the carrier (or IF) frequency components. Thus, detected signal will only have noise which was either present in the signal when it was modulated (not a consideration) or ultra-low bandwidth one Hz noise. Other than this detected output signal will only have thermal noise, also known as **"Johnson noise".** It amounts to reducing noise bandwidth to 1 Hz, which can be easily ignored.

Johnson–Nyquist noise (*thermal noise, Johnson noise, or Nyquist noise*) is the electronic noise generated by the thermal agitation of the charge carriers (usually the electrons) inside an electrical conductor at equilibrium, which happens regardless of any applied voltage. Thermal noise is present in all electrical circuits, and in case of sensitive electronic equipment such as radio receivers, can drown-out weak signals, and can be the limiting factor on sensitivity of an electrical measuring instrument. As the name implies thermal noise increases with temperature. Some sensitive electronic equipment such as radio telescope receivers are designed to operate at close to cryogenic temperatures to reduce thermal noise in their circuits. The known generic, statistical physical derivation of this noise is known as 'the fluctuation-dissipation theorem', where use of generalized susceptibility or generalized impedance can characterize the medium.

Thermal noise in an ideal resistor is approximately white noise, meaning that the power spectral density is nearly constant throughout the frequency spectrum. When limited to a finite bandwidth, thermal noise has a nearly Gaussian amplitude distribution.

For 20 KHz audio signal bandwidth, this thermal noise works out to -131 dBm, based on the well-established Johnson noise calculations of -174 dBm, noise floor for the 1 Hz channel bandwidth. This -131 dBm, noise floor available, for received base band signal, is unimaginable with existing modulation techniques and open new possibilities as per **Shannon theorem's last part**. Which corelates the higher SNR with higher capacity.

$$D = B \operatorname{Log}_2\left(1+\frac{S}{N}\right)$$

Hence in terms of quality improvements the communication system can become almost noise free in comparison to present day realities. This improvement in signal quality is another indirect advantage opening unlimited possibilities in **base band multiplex** configurations. This can lead to many more innovative applications as the technology evolves and systems become more accepted with better application specific hardware is developed.

Bandwidth in Received Baseband

Signal bandwidth capacity in **"Z-Mod"** concept approaches the theoretical limits. As per **"Z-Mod"** we can presume each cycle carrying one sample and as per Nyquist theorem it can reproduce signals up to half the sample frequency. The capacity may depend on many other factors and type of modulation being used. So, if we take the same simple example of AM at 1.5 MHz carrier frequency which we have calculated to carry analog signal bandwidth of at least 200 KHz in the most basic implementation with 455 KHz IF based receiver, ensuring backward compatible with legacy receivers. Even if we use the entire 200 KHz available base band spectrum for some super hi-quality large bandwidth application the noise floor in detected output will be -121 dBm for 200 KHz bandwidth and -131 dBm for audio bandwidth of 20 KHz as calculated in previous section. This superior noise free performance can set new quality standards as well as options for improving reliability.

Presently an AM radio receiver provides a SNR of around 50-60 dB with 5 KHz of signal bandwidth.

Digital Multiplex in Baseband

Digital baseband multiplex is the future of all communication systems, and **“Z-Mod”** concept provides some extra ordinary capabilities. Now we reconsider the above AM example of 1.5 MHz carrier frequency which we have calculated to carry analog signal bandwidth of 200 KHz of analog bandwidth with -121 dBm of noise floor.

This available high quality, analogue signal bandwidth can be very useful in a digital multiplex situation, for carrying extremely high data capacity. This digital multiplex becomes more important considering the available high signal to noise ratio for the complete multiplex. To put this in hypothetical perspective we can this 200 KHz bandwidth as available spectrum and corelate it to crest factor of the multiplex. In practice, for combining 2 subcarriers of same amplitude, the peak amplitude of combined coherent signal (worst case) will increase by 6 db. When we use 4 subcarriers it will increase by 12 dB and so on …for 1000 sub carriers, it will increase by 60 db. This 60 dB requirement to accommodate 1000 carriers, in comparison to available noise floor of -121 dBm uses less than half of available SNR. This correlated with crest factor calculations confirms that hypothetically the multiplex could have hundreds of subcarriers at lower peak levels. Now when we correlate this reduction in individual amplitude of subcarriers in baseband multiplex, we will still have more than 60 dB signal to noise ratio for each subcarrier available for decoding the multiplex. Thus, if we are able to transpose these subcarriers with **"Z-Mod"** principal compliant base band multiplex, the average subcarrier frequency can be presumed 100 KHz (average of 200 Khz). The data capacity based on one bit sample (could be Byte) per cycle could be 100 Kbps minimum. Further if we keep our subcarriers at practical 100 Hz apart, there will be 2000 subcarriers, yielding 200 Mbps data capacity using 1 Hz bandwidth of the entire 10KHz AM channel. This hypothetical example is to project the potential capabilities using presently available hardware technology and all the limitations of backward compatibility with legacy receivers and only using partial capabilities.

When we do not want to maintain the backward legacy receiver compatibility, we can easily have more than 100 such main carriers in one MW AM radio transmission channel. The point to emphasize is the fact according to **Shannon's Theorem** availability of Very high Signal to Noise ratio can provide enormous data capacities.

To make communications **ultra-robust** and take best advantage of such superb noise performance provided by **"Z-Mod"** systems, digital data can be encoded, using Bi-phase mark code (BMC) like **AES/EBU** audio format, which allows the data to be run at any asynchronous rate and receiver can easily decode data by decoding the Bi-phase mark code (BMC). This way we will be able to carry sensitive commerce data with fail safe provisions.

Going further we need to calculate SNR of individual subcarriers after detection of each RF carrier with 1 Hz bandwidth, we can have about 1600 subcarriers at 100 Hz apart, For allowing each of 1600 subcarrier to have equal amplitude, each will have 1/1600th of amplitude, this means each carrier will be 1 mv if the peak amplitude of detected signal is 1.6 volts. Or about -65 dBm with reference to a 0 dBm amplitude of multiplex signal, having 200 KHz bandwidth. This provides available SNR of 56 dB for each subcarrier, as we have Noise floor of -121 dBm calculated above. Whereas for digital signal to be decoded properly within a multiplex, without errors, a SNR of 10 – 12 dB is considered more than adequate. This leaves us still margin of more than 40 dB for either increasing the number of carriers with more bandwidth or increase the detector bandwidth with optimization for applications.

Reduction in Health Hazards

Considering these technical improvements **"Z-Mod"** technology has another advantage of reducing human exposer to RF waves which is known to be a health hazard, for exposer above small quantity. It is believed that the current levels of RF radiation exposer by Cell phones on prolonged use, crosses the border line of safety. Most other devices whether walkie talky or even legacy chord less phones or Wi-Fi gadgets, expose humans to beyond safe levels of RF radiations. With **"Z-Mod"** technology which requires 1,000 times less RF radiation for covering similar distances, we can make the world a much safer place with weaker electromagnetic waves radiated and without sacrificing quality, connectivity and reliability of communication systems.

New Challenges

This path breaking technology creates new technical challenges due to fast processing of information in a cycle-by-cycle process. To realise full potential of this superfast processing of information to the extent of one third of carrier frequency and projected data capacities reaching theoretical limits, it will need all round improvements in electronics. This could be components, design, or analytical instruments.

How ever the validation of the **"Z-Mod"** with present technology and test and measuring system is possible with distinctly displayed higher capabilities with higher bandwidth of the systems.

Technological challenges

First challenge is to formulate new standards to apply within existing standards to accommodate "Z-Mod" method to provide improvement in existing systems, with backward compatibility. Even with present hardware capabilities **"Z-Mod**" process can offer phenomenal advantages allowing progressive transition to next phase, targeting full potential. How ever for utilising full potential of **"Z-Mod**" process, many technical challenges in hardware needs to be overcome. For-most is the need to have ultra narrow band filters suitable for **"Z-Mod**" fast cycle-by-cycle changes. Challenging improvements required in filter design is to preserving high dynamic data with in ultra narrow pass band of few Hz. allowing modulated signal, having swift changes in cycle-by-cycle operation.

It will need improvements in active as well as passive electronic components to achieve temperature stability to keep frequency drifts to new manageable limits, reaching theoretical limits due to narrow bandwidth, of few Hz.

Test Setup challenges

Many new technical challenges in respect of measurement, to meet contradicting requirement, needing frequency selection of 1 Hz or less and still be able to retain step changes from one cycle to another cycle. Filters needed for reception and for that matter even for transmission, will need filter designs with contradicting requirements of high selectivity at low 'Q'.

Conclusion

These claims and calculations provide enough inputs for development of future communication systems with far superior, safe and far-reaching applications. This patented technology is changing the fundamental belief of communications theory, that the information is carried in the side bands and establish that side bands are not needed to carry large amounts of information within the carrier frequency itself theoretically with zero bandwidth. This is certainly a new class of modulation process named "Z-Mod" which can be applied to all types of known modulation processes to remove side bands and provide, benefits beyond imagination.

At present the pilot project is in advanced stage, with proof of concept already tested and demonstrated. Now the next phase with FPGA based hardware design, development is in process.

Simultaneously the Fast Fourier Transform analysis results confirm the concept. Certification work for the same is in progress from reputable organizations.