

ABSTRACT

The use of under-sampling is quite new to the telemetry marketplace, including not only the topics of rates and planned aliasing, but also how to evaluate the effects of under-sampled IF recording.

ABOUT THE AUTHOR:

Tim Gatton has been involved in real time telemetry since the late 1970's; first with EG&G Special Projects in southern Nevada, and currently with Wideband Systems Inc. Mr. Gatton teaches Intermediate Techniques for the annual International Telemetry Conference (ITC) each year, but also teaches a three-day multi-disciplinary course on Telemetry Systems Engineering for TTI.

THE GOAL OF THIS PAPER:

It is not the goal of this paper to review all of the high-order math proofs that others have accomplished and published on the Internet. Instead, this paper is designed to provide the fundamental understanding of IF recording using under-sampling techniques that Messer's Nyquist and Shannon outlined in their works at Bell Labs, and then comparing the various PCM recording data rates.

To learn more about how to test IF recording using an undersampled approach, please see the appropriate WSI paper.

INTRODUCTION:

Over the last fifty years, we have undergone incredible changes in the data recording arena. From post-detection, post-decision recording with reel to reel magnetic tape, to using the ubiquitous Chapter-10 digital recorders with many gigabits of bandwidth to record pre-detected data. This paper examines this latest technology implementation ... using the Nyquist-Shannon's Theorem to under-sample high-frequency data and compare the results.



Fig 2
Harry Nyquist

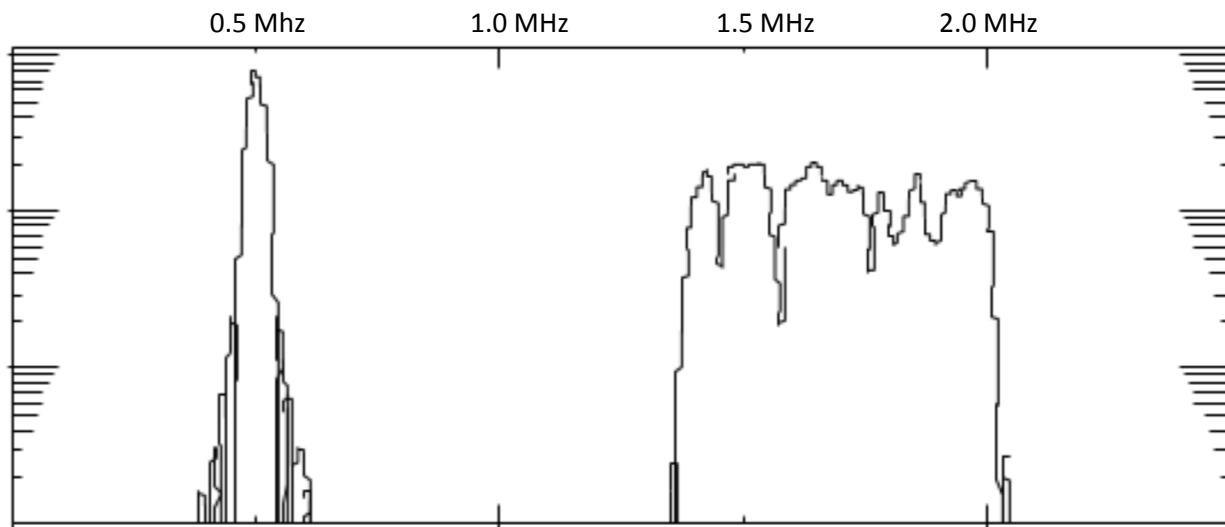


Fig 1
Claude Shannon

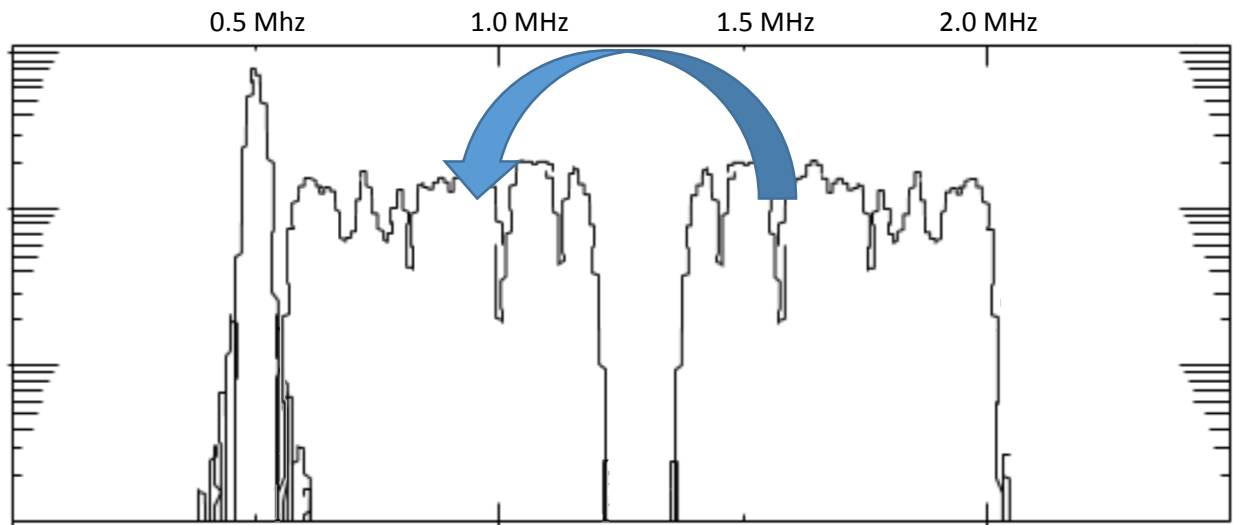
NYQUIST AND SPECTRAL FOLDING:

In order to appreciate under-sampling techniques, you first need to understand the concept of spectral folding. The following example will help with that.

First, assume that you have a signal with the following spectral plot:



The Nyquist-Shannon Theorem says that if you sampled this signal at 5 MSPS, all frequencies below $\frac{1}{2}$ of the sample rate ($F_s/2$) can be reproduced. Furthermore, the theorem continues with the concept that anytime you sample a signal and the $F_s/2$ is less than the highest frequency in the signal, that energy above $F_s/2$ corrupts (or aliases) the signal. This occurs due to spectral folding ... as can be seen below when you sample this signal at 2.5 MSPS (and the Nyquist, or $F_s/2$, is 1.25 MHz):

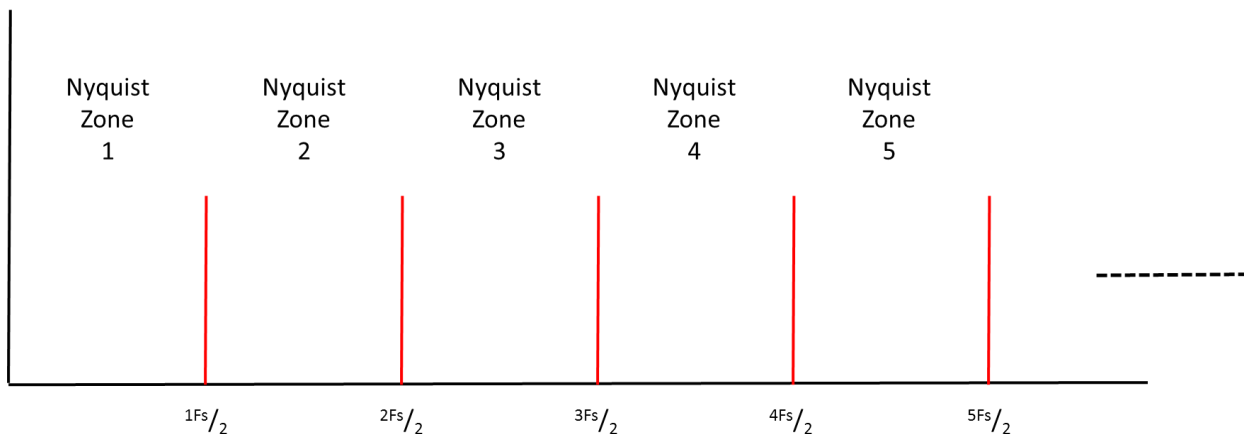


This spectral folding can continue again and again as the sample rate drops lower and lower. This progression of “Nyquist Zones” can be used to your advantage because:

1. Spectral folding, or aliasing, is not always destructive. If you keep your spectrum within a single Nyquist zone, when the signal is re-constructed, aliases (or echos) of the original signal remain (albeit at a lower amplitude).
2. If you can stay as closely centered to each Nyquist zone as possible, the signal bandwidth can be maximized.

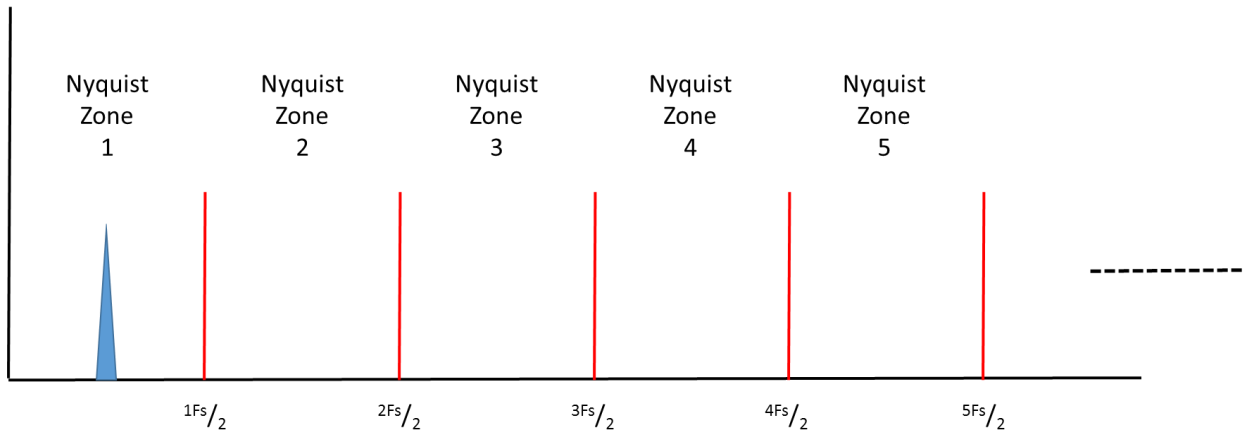
NYQUIST ZONES:

Nyquist zones are best visualized using a chart like this one:

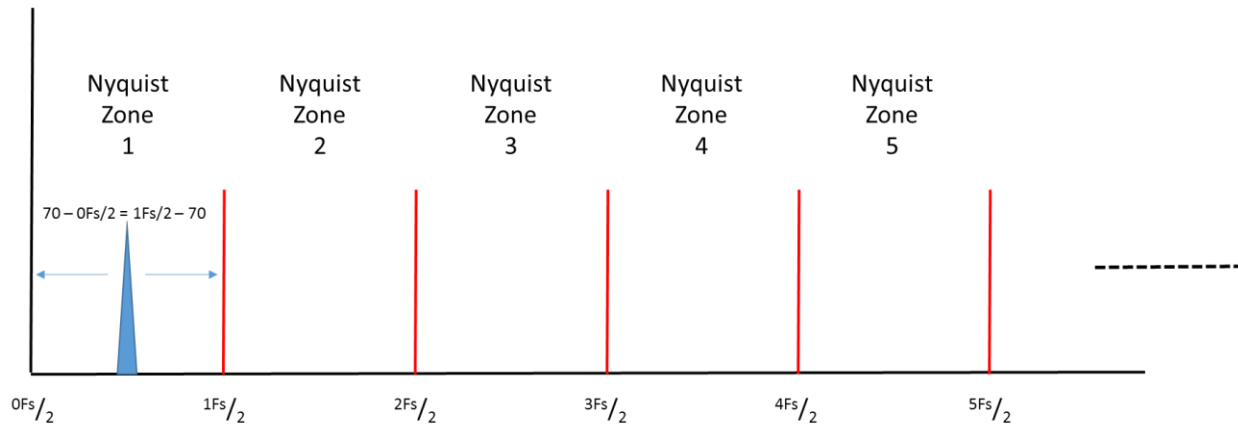


The way to use this spectrum allocation is to decide which Nyquist zone you want your signal to occupy.

As an example of addressing a 70 MHz IF signal, let us first assume we wish to use Nyquist Zone 1 ... the traditional oversampling approach. To do that, we place the signal of interest in Zone 1 as follows:



The formula to compute the optimal sampling rates to use (optimal in this case means centered in the spectrum) is $70 - 0F_s/2 = 1F_s/2 - 70$ as shown in the following:

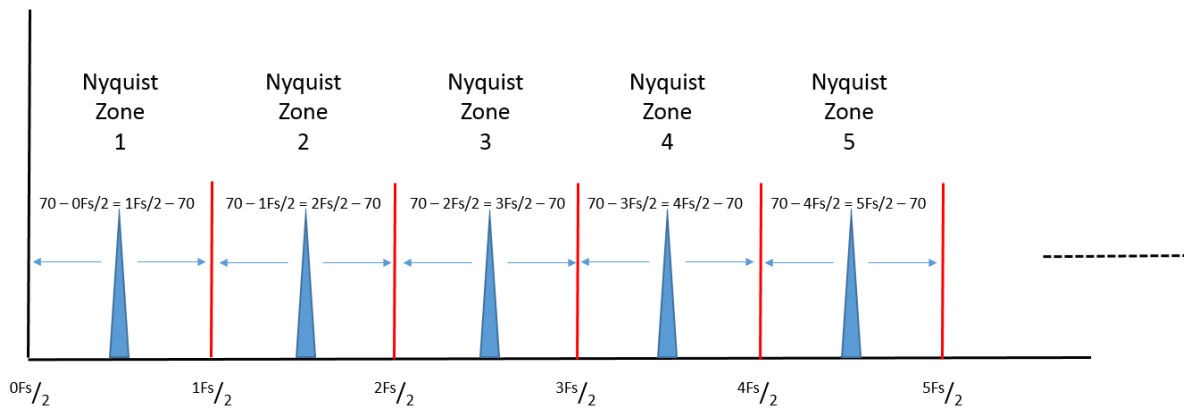


The formula resolves as follows:

- $70 - 0F_s/2 = 1F_s/2 - 70$ (since $0F_s/2 = \text{DC, or } 0$)
 $70 = 1F_s/2 - 70$ (add 70 to each side)
 $140 = 1F_s/2$ (multiply each side by 2 now)
 $280 = F_s$

Thus, sampling at 280 MSPS makes the signal (70 MHz) sit at the center of the supported spectrum (140 MHz). While the theoretical bandwidth is $F_s/2$ (or 140 MHz), the reality of analog filter implementation typically limits you to 120 MHz.

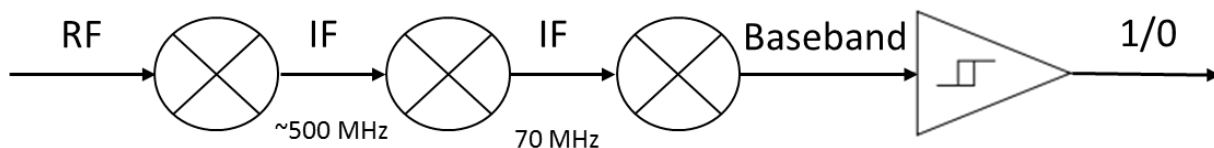
This same approach is used for all of the other zones as outlined in the following section.



Zone 1	Zone 2	Zone 3	Zone 4	Zone 5	
$70 - 0F_s/2 = 1F_s/2 - 70$	$70 - 1F_s/2 = 2F_s/2 - 70$	$70 - 2F_s/2 = 3F_s/2 - 70$	$70 - 3F_s/2 = 4F_s/2 - 70$	$70 - 4F_s/2 = 5F_s/2 - 70$	Sample Rates
$70 + 70 = 0F_s/2 + 1F_s/2$	$70 + 70 = 1F_s/2 + 2F_s/2$	$70 + 70 = 2F_s/2 + 3F_s/2$	$70 + 70 = 3F_s/2 + 4F_s/2$	$70 + 70 = 4F_s/2 + 5F_s/2$	
$140 = 1F_s/2$	$140 = 3F_s/2$	$140 = 5F_s/2$	$140 = 7F_s/2$	$140 = 9F_s/2$	
	$280 = 3F_s$	$280 = 5F_s$	$280 = 7F_s$	$280 = 9F_s$	
$280 = F_s$	$93.33 = F_s$	$56 = F_s$	$40 = F_s$	$31.11 = F_s$	
$F_s/2$ or 140 MHz	$F_s/2$ or 46 MHz	$F_s/2$ or 28 MHz	$F_s/2$ or 20 MHz	$F_s/2$ or 15 MHz	Theoretical Bandwidth
120 MHz	40 MHz	20 MHz	10 to 15 MHz	5 to 10 MHz	Practical Bandwidth

BIT RATE CONSIDERATIONS:

For many years, the telemetry community has most commonly only recorded the final output of 1’s and 0’s, allowing the settings of the receiver and bit synchronizer to decide what is a zero and what is a one. With the advent of high speed digital recorders, we now have the flexibility and bandwidth to record at the IF, Baseband, or the final 1’s or 0’s output.

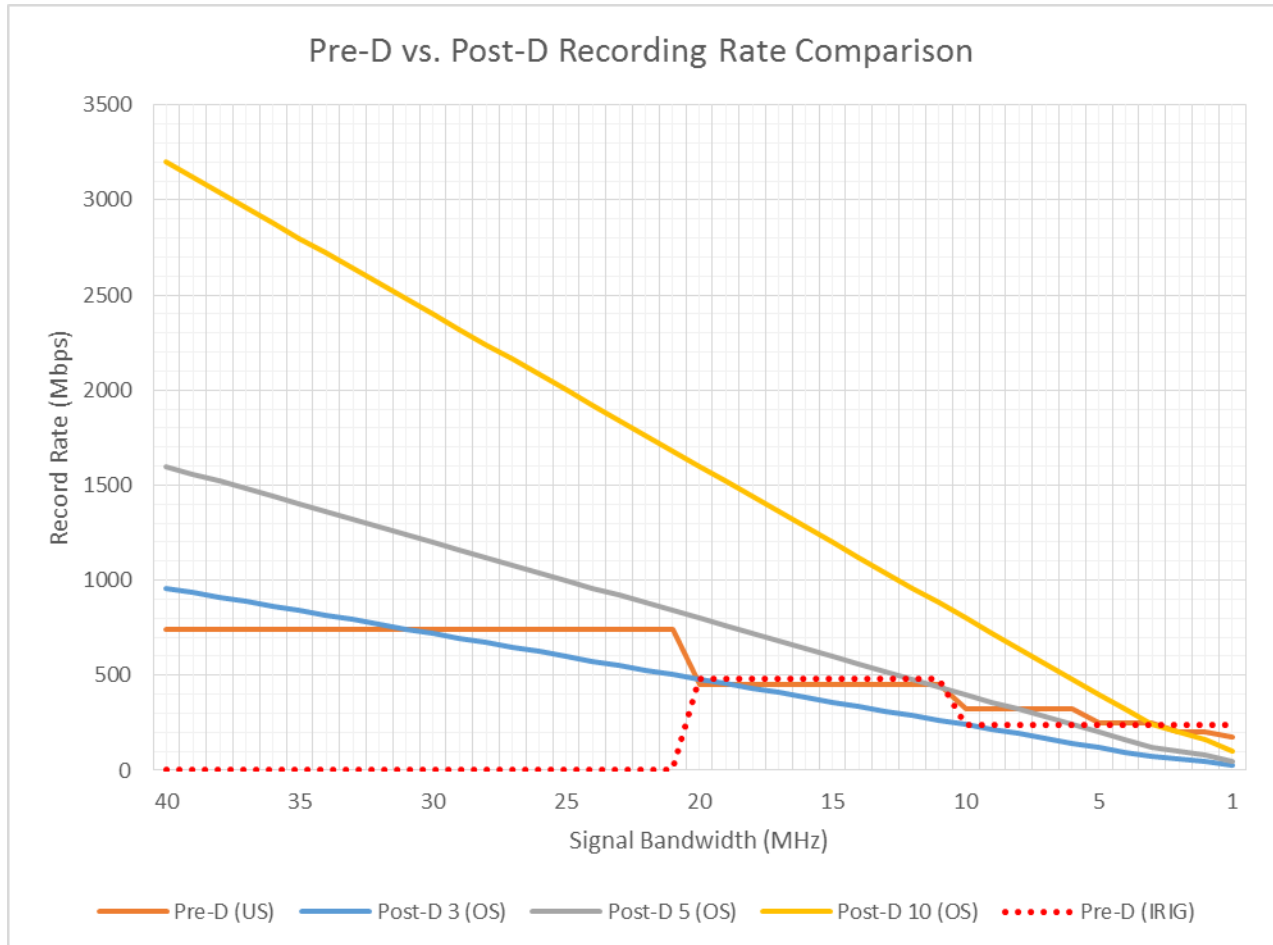


Certainly, recording only the final output of ones and zeros provides the best utilization of “file size” efficiency, but if the user wants to be able to adjust his receiver and/or bit synchronizer settings, this can only be accomplished by recording further “up-stream” in the flow of data.

Most users of recording technology have assumed for years that only the typical “over-sampled” approach was applicable, and that the raw baseband output was the only methodology available to record.

The following plot shows a) the bit rates associated with recording the baseband signal at 3, 5, and 10 samples per bit, b) the bit rates associated with under-sampled approach in recording the 70 MHz Pre-D signal, and c) the bit rates associated with recording the IRIG standard 10 MHz and 20 MHz pre-d signals

(note that recording the final 1's and 0's is not plotted as it is acknowledged to be the most effective in minimizing bit rates and file sizing).



Of interest is that recording in the classic Pre-D Cf signals of 10 and 20 MHz is no more efficient than that of recording the 70 MHz IF with an under-sampled approach. If you note the orange plot of a 70MHz IF (under-sampled), you'll see that recording bandwidth is less than 5 or 10 samples/bit plots of baseband data, and supports rates that traditional IRIG-based pre-D recording could not implement (via its 3rd IF).

Naturally, nothing is more efficient than recording the ONEs and ZEROS after the bit sync has made its decision, but if you want to get better results by adjusting bit sync and decod settings, only IF recording can provide that capability.