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# A Review of the Basic Principles and Methods of Signal Sampling Technology

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**Abstract.** In modern life, communication is an essential human activity. In the process of communication, various information spreads around the world in the form of signals. In order to optimize the communication, the reasonable processing of the signal becomes an important research problem. Signal processing is an important step during signal processing. It is one of two processes that convert an analog signal into a digital equivalent signal. It can be considered that signal sampling transforms the time axis of the signal into a set of discrete time moments, that is, the process of converting a signal in a continuous time domain to a discrete time domain through a series of methods. It is used to convert continuous time signals into digital signals for relevant processing and storage of digital signals in subsequent operations. This paper will summarize the basic principles and methods of signal sampling technology, introduce the sampling theorem, sampling frequency, sampling depth, sampling method and sampling error, and finally discuss the future development of signal adoption technology.

**Keywords:** Signal adoption technology; sampling theorem; sampling frequency; sampling depth; sampling method.

## 1. Introduction

With the advent of the information age, the signal sampling technology has been widely applied in various fields. It plays an important role in communication systems, audio processing, and medical images. Through sampling and subsequent operations, signal processing, transmission and storage can be achieved. Its significance is to reduce the bandwidth and storage space occupied by the signal, and improve the transmission efficiency and signal quality. Further investigation of signal sampling can deepen the understanding of signal processing for further research in the field. This has played a very important role in the development of related industries.

Signal sampling is to discretize the continuous signal in time, that is, to take the instantaneous value point by point on the required sampled analog signal at a certain time interval. The process can be achieved by multiplying the sampling pulse and the simulated signals.

The choice of sampling interval is an important part of signal sampling. Generally speaking, the higher the sampling frequency, the more sampling points obtained, the closer the interval between these points, the more similar the discrete signal; but the high sampling frequency will bring huge amount of data, which brings huge computing workload and storage space to the computer; if the sampling frequency is too low, the sampling points will be too far, making the discrete signal is not enough to reflect the original signal waveform characteristics, resulting in signal confusion.

According to the sampling theorem, when the sampling frequency is greater than twice the maximum composition frequency in the original signal, the signal can be better reduced, while the signal aliasing will occur.

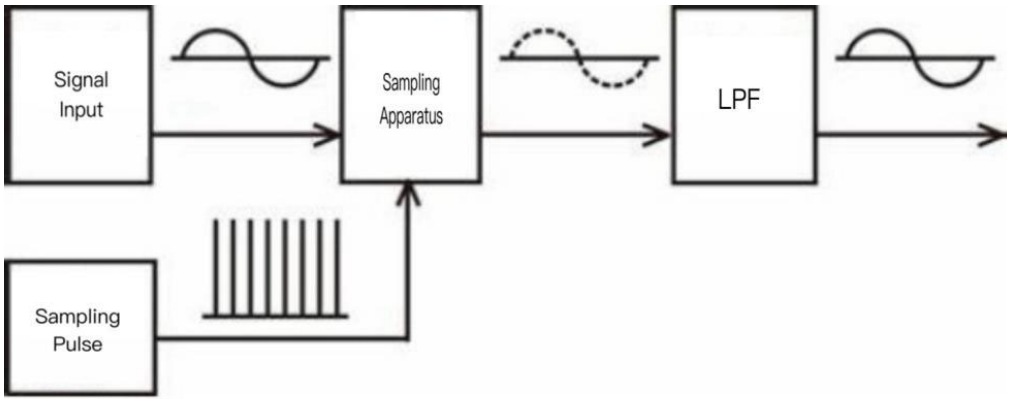


**Figure 1.** Sampling

## 2. Sampling Theorem

The sampling theorem, also known as Shannon sampling theorem or Nyquist sampling theorem, is a basic concept in the field of digital signal processing. It describes how to obtain discrete samples from continuous time signals and then form discrete signals, while also ensuring that there is no distortion when the original signal is reconstructed later. It establishes the relationship between sampling frequency and signal frequency spectrum, which is the basic basis for continuous signal discretization. According to the different sampling domain, the sampling theorem can be divided into time domain sampling theorem and frequency domain sampling theorem. If sampled in the time domain, according to the sampling theorem, the sampling interval should be less than or equal to twice the highest frequency component of the signal. If sampled in the frequency domain, for temporally limited continuous signals, the sampling point frequency interval should be less than or equal to π divided by the maximum value of the signal duration. If the above conditions are satisfied, the discrete samples can fully recover the original signal. Otherwise, the signal of high frequency component will be misinterpreted as low frequency component, resulting to signal distortion.

The application of the sampling theorem is mainly reflected in the conversion of analog signal to digital signal, signal reconstruction, avoiding spectrum aliasing, communication and signal processing, etc.

In the conversion of an analog signal to a digital signal, the sampling theorem sets a sufficient condition for the sampling rate that the sampling frequency must be greater than twice the highest frequency in the signal. In this way, the sampled digital signal can completely retain the information in the original signal.

In signal reconstruction, if the signal is band-limited and the sampling frequency is higher than twice the highest signal frequency, then the original continuous signal can be completely reconstructed from the sampled sample.

When the sampling frequency does not meet the condition (i. e., less than twice the frequency maximum frequency frequency), spectral aliasing will occur, resulting in the failure to correctly recover the original signal. This provides guidance for selecting the sampling frequency and avoiding spectrum aliasing during sampling.

The sampling theorem is widely used in the field of communication and signal processing, which is the theoretical basis of analog signal digitization. It helps engineers design effective sampling and signal processing systems to ensure the accurate transmission and processing of signals.

Band-pass sampling theorem is the specific application of sampling theorem in the sampling of band communication signal (i. e., the signal frequency spectrum in a certain finite frequency band). For band-limited signals, if the spectrum is distributed within a finite frequency band, then a sampling rate below the Nyquist frequency can be sampled without loss of signal information. This is achieved by band-pass filtering the signal before sampling, restricting the signal to a certain frequency band,

and then sampling with an appropriate sampling rate. Its application can significantly reduce the sampling rate and reduce the amount of data, thus saving storage space and transmission bandwidth. However, it should be noted that the sampling rate satisfies the conditions of the band-pass sampling theorem to avoid spectral aliasing and signal distortion.

In recent years, the classical Shannon sampling theorem has been generalized from spectral finite function spaces to more general translation-invariant subspaces, and sampling methods have also been extended from point-by-point values to average sampling and multi-channel sampling, etc[1].

In addition, Zhou Panpan[2] et al. constructed a class of rational function ladder filter based on the existing polynomial ladder filter, and gave a sampling theorem for non-band limit signals.

## 3. Sampling Frequency

The sampling frequency refers to the number of sampling points extracted from the continuous signal and forming the discrete signal per unit time in Hertz (Hz). The reciprocal of the sampling frequency is the sampling period, referring to the time interval between the samples. The continuous signal varies in time (or space), while the sampling process corresponds to measuring the value of the continuous signal in time (or space). Sampling frequency can only be used for sampler with periodic sampling, but there is no rule limit for sampler with aperiodic sampling.

Generally speaking, the sampling frequency reflects the number of signal sample points collected by the computer per unit time. For example, for the waveform recording, the sampling frequency can be the quality standard to describe the waveform. Its importance is that it determines the reduction accuracy of the signal. According to Nyquist theory, the original signal can be recovered from the sampled signal only if the sampling frequency is higher than twice the highest frequency of the signal. The higher the sampling frequency, the more sample data obtained by the computer per unit time, and the more accurate the waveform formed.

The adjustment of sampling frequency has always been a key point in the project. Gai Bingshuai[3] et al. proposed an adaptive sampling frequency adjustment method. They used a linear regression to fit a line to several recent sampling points, then calculated the jitter ratio according to the line, and adjusted the sampling frequency according to the jitter ratio.

## 4. Sampling Depth

Sampling depth represents the precision of sound intensity recording in sampling, also known as quantification precision. It refers to the total number of sample points of the measured signal sampled at one time, which determines the accuracy and dynamic range of the digital signal, and also directly determines the amount of data[4] that can be collected at one sampling time. In sound signal sampling, it determines the resolution of the sound processed by the acquisition card. The larger the value, the higher the resolution, the more real the sound recorded and played.

The common sampling depth is 8 bits, 16 bits and 24 bits, etc. These values represent the size of each sampling point. The larger the value, the wider the range of sound intensity, and the higher the sound quality. A higher sampling depth can increase the amount of data collected, providing a higher signal-to-noise ratio and better dynamic range, but also increase the storage and processing requirements of data.

## 5. Sampling Method

### 5.1 Band-through Sampling

In practice, many of the signals encountered are connected signals, whose bandwidth is often much smaller than the center frequency of the signal. When sampling with band communication signal, the sampling frequency is often higher than two times the cutoff frequency. The sampling frequency can be determined according to the band pass sampling theorem to conduct band pass sampling.

The band-pass sampling theorem refers to that for a certain communication signal, its frequency is limited between the upper cut-off frequency $f_L$ and the lower cut-off frequency $f_H$, and then its bandwidth is $\mathrm{B} = f_L - f_H$. If the minimum sampling frequency $f_S$=$\frac{2f_H}{m}$ (m is a maximum integer not more than $\frac{f_H}{\mathrm{B}}$), then the signal can be completely determined by its sampling value.

As for band-pass sampling, many researchers have explored it. For example, Zhao Deming[5] et al. proposed directional algorithm optimization based on Jetson TX1 band-pass sampling, Fan Jiaheng[6] et al. designed anti-aliasing filter based on band-pass sampling, and Zhang Wenqiong[7] et al. analyzed and studied band-pass sampling frequency domain beamforming.

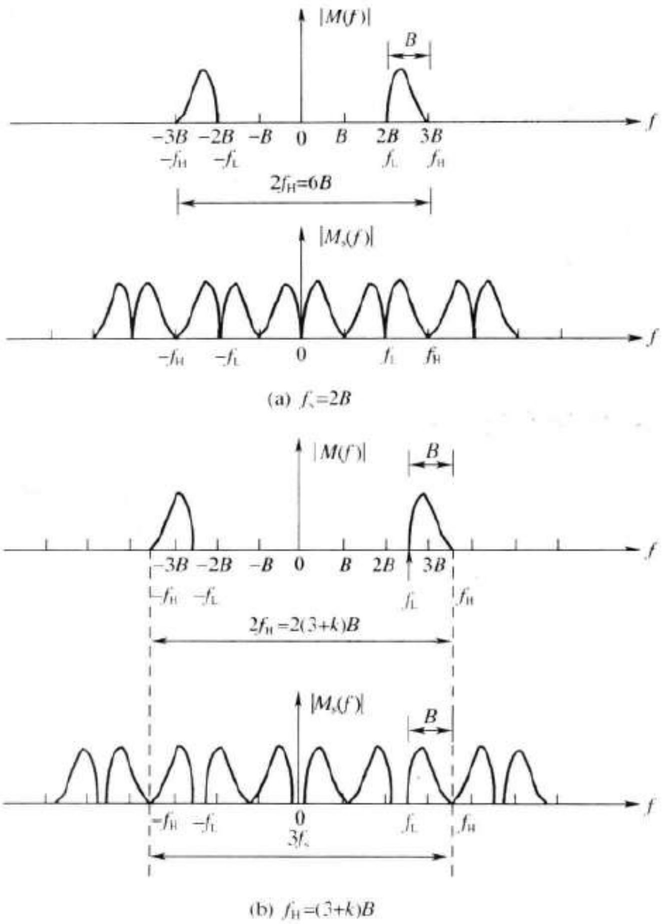


**Figure 2.** Band-through Sampling

### 5.2 Multi-rate Sampling

Multiple rate sampling and its derived rate signal processing is the key to the software radio signal digitization, although band sampling can greatly reduce the radio frequency sampling rate, but from the perspective of the requirement of software radio, its bandwidth should be the wider, the better, so you will have better adaptability to different signals, and simplify the system design. Improving the sampling rate can improve the signal-to-noise ratio of sampling quantification, but at the same time, the subsequent signal processing speed can not keep up, and it is difficult to meet the real-time requirements. So it's necessary to slow down the data flow. The multi-rate signal processing technology provides a theoretical basis for the realization of this deceleration processing. Multi-rate signal processing is essentially a resampling process of discrete sequences after sampling.

Multi-rate sampling can be achieved by integer multiple extraction or integer multiple interpolation methods. The integer D times extraction refers to the formation of a new sequence $xD(n) = x(Dn)$ every (D-1) of the original sampling sequence x(n), and the positive integer D is the extraction factor to improve the signal frequency domain resolution; the integer I times interpolation means the insertion of (I-1) zero values between two adjacent sampling points of the original sampling sequence x(n) to obtain the interpolated sequence $xI(n) = x(n/I)$, n=0, $\pm I, \pm 2I\cdots$ to improve the signal time domain resolution.

In the field of multi-rate sampling, Bu Xiangyuan[8] et al. designed the multi-rate sampling module based on the multiphase decomposition of extraction filters, Liu Yan[9] et al. studied the broadband cooperative spectrum perception of multi-rate sampling in cognitive networks, and Li Xiayu[10] et al. conducted the design of multi-rate sampling control system based on robust stability.

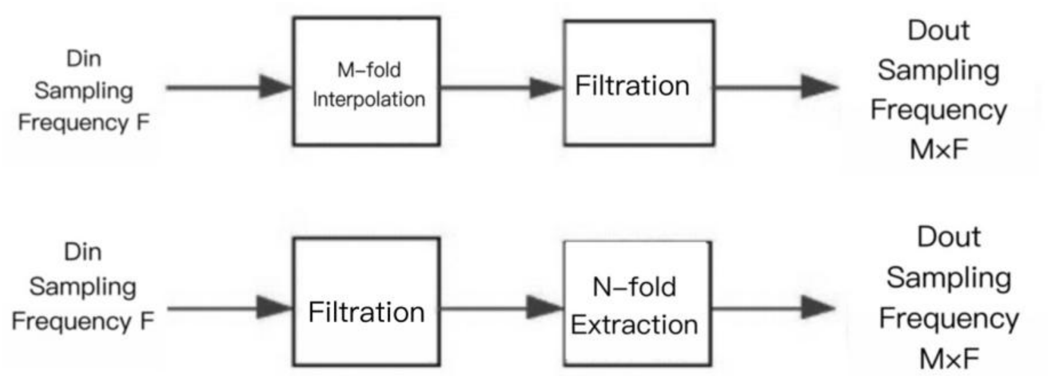


**Figure 3.** Multi-rate Sampling

### 5.3 Compression Perception

Compression sensing is a technique for finding sparse solutions of underdetermined linear systems. It is often used in the field of signal processing to acquire and reconstruct sparse or compressible signals. Compared to Nyquist theory, it takes advantage of the characteristics of sparse signals, obtains discrete samples of signals with random sampling, and uses less measurements to reconstruct the original signal[11] through a nonlinear reconstruction algorithm.

It is now widely used in wireless communication, array signal processing, biosensing and other fields. At the same time, Wang Jia[12] et al. proposed a real-time integrated defense strategy of neural network based on compressed perception, Ren Bing[13] et al. proposed a correlation data filling method based on compressed perception, and Pan Zemin[14] et al. studied the image reconstruction of block compressed perception based on deep neural network.

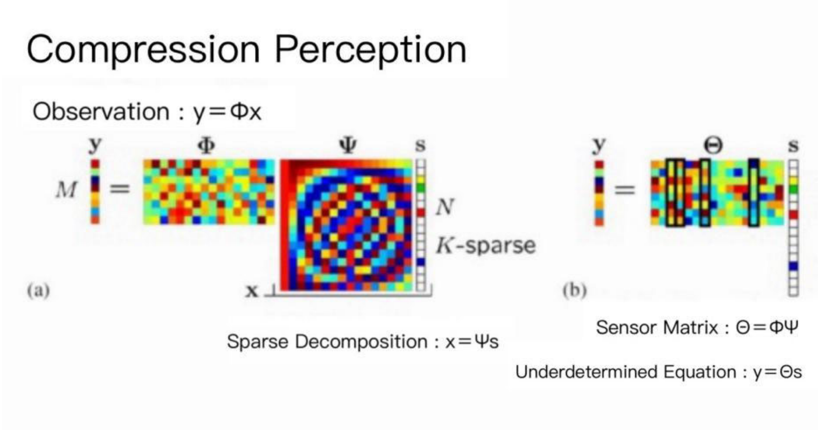


**Figure 4.** Compression Perception

## 6. Sampling Error

Since sampling is a discrete process, a sampling error is introduced. Sampling error includes quantification error and sampling error. The former is caused by the limited sampling depth, which limits the accuracy of the digital signal; the latter is caused by the difference between the sampling time and the true value of the signal, depending on the sampling frequency and the change speed of the signal.

Signal sampling error refers to the difference between the sampling result and the actual signal due to various factors during the signal sampling process. The sampling error may be due to the complexity of non-band-limited random signals, leading to a truncation error during the sampling recovery. In order to sample the recovery more accurately, select the appropriate sampling series to approximate the random signal and analyze its convergence and error.

Other types of error may also be encountered during sampling, such as noise, which refers to the interference signal, i. e., the signal component that we do not want. The sources of noise include potential structural problems, sensor noise, and grounding cycle noise. For example, differences in test objects may lead to different characteristics of the structure itself, introducing irrelevant vibration noise, the type and state of sensors used may also lead to noise, and grounding cycle noise when using "non-isolated" sensors, especially when test links.

In conclusion, signal sampling error is a complex phenomenon involving multiple factors and conditions and needs to be reduced or avoid by appropriate techniques and analytical methods to improve the accuracy and reliability of sampling.

## 7. Summary and Outlook

Signal sampling technology is the process of converting signal from continuous time domain to discrete time domain. It has wide application in the field of digital signal processing and communication, and is of great significance for the development of signal processing and other fields. Based on the basic principles (signal sampling theorem) and some related concepts, combined with the recent research results of others, the basic principles and methods of signal sampling technology are summarized. Around the sampling theorem, the selection principle of sampling frequency is summarized, and several common sampling methods are introduced, and the sampling depth and sampling error are also briefly introduced. It is believed that in the near future, with the in-depth research of sampling principle and the breakthrough progress of related fields, emerging sampling methods will appear in the field of signal sampling. On the premise of without affecting the sampling results, the sampling error will be minimized so as to obtain better sampling results. In turn, the signal sampling technology has a greater promotion effect on other fields.

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