THE NEWLY IMPLEMENTED SUPER PCM SYSTEM

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RESUME

Le concept d'utiliser l'échantillon abstrait pour modifier la théorie classique de l'échantillonnage est introduit dans cet article.
La théorie présentée ici, parfois appelée "forme d'onde abstraite [1]", est plus intéressante et plus puissante que la théorie classique de l'échantillonnage. Dans le domaine du système de l'information, la "forme d'onde abstrailte" est l'une des clés pour résoudre le problème de la communication numérique. Beaucoup de problèmes dans la communication numérique peuvent être ainsi facilement résolus ou améliorés. Par exemple, utiliser les principes de "forme d'onde abstraite" pour effectuer la transformation de Fourier est plus simple, plus rapide et plus juste que la méthode DFT/FFT. Le thème de cet article consiste à présenter le résultat fructueux de l'implantation du "Super-Pulse-Code-Modulation(SPCM)[2]. Si le système est utilisé pour la transmission numérique de voix, un système SPCM de 24K bits/s est beaucoup mieux qu'un système ADPCM de 32K bits/s. Pour la reproduction musicale, on n'a besoin prendre que 20K d'échantillons abstraits pour avoir un meilleur résultat que le lecteur de disque compact. Dans cet article, trois sujets sont introduits: (1) L'introduction à SPCM et une comparaison avec le système PCM. (2) La théorie de "forme d'onde abstarite" et le codage. (3) Le "Super PCM system" et ses performances.

SUMMARY

The concept of using abstract samples to modify the conventional sampling theory in a wide sense is introduced in this paper. The theory to be discussed here is sometimes called **"WAVEFORM ABSTRACT THEORY [1]"**, which is a reasonable and powerful replacement of the conventional sampling theory. The **WAVEFORM ABSTRACT** is one of the noble keyes to the information era. Many digital communications problems can thus be solved or improved easily. For example, using the principles of **WAVEFORM ABSTRACT** to do Fourier Transform digitally is much simpler and far better than the so-called DFT/FFT methods. The theme of this paper is to present the fruitful result of the implementation of the Super-Pulse-Code-Modulation (SPCM) [2]. If this system is used for digital voice transmission, a 24K bits/s system of SPCM is much better than the 32k bits/s system of ADPCM. For music reproduction applications, it only takes 20k abstract samples to have nicer performance than a Compact Disk (CD) player. In this paper a description of the following three items will be given.

1. An introduction of SPCM and a comparison with PCM system.
2. The theory of Waveform Abstract (WA) and its coding.
3. The Super PCM system and its performance.

INTRODUCTION

The Super PCM is a nonfiltering PCM system which alleviates all the pains of conventional PCM systems. SPCM can be used to transmit either quality signals through a digital channel or digital data through an analog channel. It gives one more freedom to design modern integrated communications systems. Its quality of regenerated signal is much better than that of all the existed A/D and D/A systems. Of the first importance it is that the implementation of hardwares of SPCM to transmit quality signals through digital systems has been just completed in the winter of the year 1988. The second part to transmit digital data through analog systems, is still in the developing stage. It is expected to be fruitful in the next year hopefully. The fundamental difference between SPCM and PCM is that the former is operated by abstract samples and the latter by direct samples. The ordinary sampler is just a switching modulator which generates a lot of unnecessary harmonics. In order to remove those harmful harmonics of PAM waveforms, it has to use approximated physical filters for

PCM systems. On the other hand the abstract samples can fully characterize a segment of signal's waveform with negligible error. An abstract sample usually has two or three vector components, which are denoted as w_0, w_1 and w_2. Therefore the transmitter is called **ABSTRACTOR.** At the receiving end the binary-coded signal through the channel is restored to be w_0, w_1 and w_2. The summation of the products of w's and regenerator's outputs will be a perfect replica of the original signal. This process is quite similar to the product detection of an AM system. The regeneration process has all the advantages of the product detection without disadvantages of it. The shortcoming of product detection as well as other modulation techniques is the need of a lowpass filter. The operation and performance of PCM/PAM decoding is just like an envelope detection but not as simple as the envelope detection. As mentioned at the beginning the regeneration of SPCM signal does not need any filters. Furthermore the signal's original harmonics are preserved while the PCM system must destroy the higher harmonics first. The hardware to regenerate the signals is called the "**REGENERATOR**".

WAVEFORM ABSTRACT THEORY

The Waveform Abstract Theory was first introduced in the tenth conference of GRETSI four years ago in France [1]. At that time all effort was done to solve the mathematical problems of waveform representation. In the August of 1987 the artical entitled "Generalized Convolution Theory" was presented in the IEEE region ten conference [3]. The important analysis of Waveform Abstract in frequency domain was announced that time. The new method is called Super Discrete Fourier Transform (SDFT). It is a method better than the so-called DFT/FFT digital computation of FT. From the frequency analysis one can prove that the Waveform Abstract is a very good approximation method than all those conventional ones. A compact description of Waveform Abstract was done in IEEE ICCS conference in November 1988 [4]. Since the limitation of the pages of this article, a simplified derivation of WA is as follows.

If a piece of signal segment is convex from t_1 to t_2, when it is normalized, signal f(t) can be represented by the following equation.

$$f(t) \cong f(t_1) + w_1\tau + w_2\tau^2, \quad t_1 \leq t \leq t_2 \tag{1}$$

where

$$\tau = t - t_1$$

The weighting factor w_1 and w_2 has many ways to define as presented in the references. One of the best is that

$$w_1 = -4f(t_1) - 2f(t_2) + 6m_0 \tag{2}$$

$$w_2 = 3f(t_1) + 3f(t_2) - 6m_0 \tag{3}$$

where

$$m_n = \int_h t^n f(t) = \text{nth moment of } f(t)$$

and

$$h = t_2 - t_1 = 1$$

The question is that the time interval ($t_2 - t_1$) is not always unity when a piece of signal is convex. This difficulty is smoothly overcome by normalization method of WA [5]. If there are some designs of electrical circuits which can perform derivatives or zero moments of derivatives, the weighting factors w_1 and w_2 can be approximated by the following rules without proof.

$$w_1 = f(t_2) - f(t_1) + \frac{1}{2}[f'(t_1) - f'(t_2)] \tag{4}$$

$$w_1 \stackrel{\Delta}{=} m(') - \frac{1}{2}m('') \tag{5}$$

The m(') and m('') stand for the zero-moments of the first derivative and second derivative of the function to be processed. The weighting function w_2 is just one half of the zero-moment of the second derivative of f(t). Therefore

$$w_2 = \frac{1}{2}f'(t_2) - \frac{1}{2}f'(t_1) \tag{6}$$

$$w_2 \stackrel{\Delta}{=} \frac{1}{2}m('') \tag{7}$$

There are several ways for the coding of abstract sample vectors. The simplest transmitter can just send pairs of end samples w_0's and area samples m_0's, where

$$w_0 \stackrel{\Delta}{=} f(t_1), \quad t_1 \leq t \leq t_2 \tag{8}$$

Then the receiver must translate w_0's and m_0's into a new vector (w_0, w_1, w_2) for the regeneration of a signal. The mapping rules are depicted in Eq.(2) and Eq.(3) respectively. This translation can also be done in the transmitter. The transmitter can either send a vector of (w_0, w_1, w_2) or a short-hand vector of (w_1, w_2). The latter can save the bit rate of a channel. The simmulated w_0 can be regenerated by the summer itself as shown in Fig. 1. If the signal is only composed of oversampled samples of a analog signal, the transmitter can also send a vector of (w_0, w_1, w_2) as shown in Eq. (4) and Eq. (6), but the conveniet way is to send a vector of [m('), m('')] as shown in Eq. (5) and Eq. (7). By using m(') and m('') the regeneration vector of w's can be calculated at the receiver.

THE SUPER PCM

The heart of Super PCM is the Super PAM (SPAM). This relation is the same for PCM and PAM. What are the major differences between SPAM and PAM? Firstly, the sampling methods are quite different. For quality signal transmission, SPAM only samples the initial value and the area of a waveform segment while PAM shall samples many waveform values in order to have good wave shape. Secondly, most natural signals (such as voice and music) have higher harmonics. If they are direct sampled by a PAM method, the undesirable aliasing effect caused by the harmonics themselves will destory the quality of a reconstructed signal seriously. So the PAM system always has a preceding continuous anti-aliasing filter to prevent aliasing, but this is not necessary for an SPAM system. In addition, the SPAM system can also preserve those beautiful harmonics which are essential to high fidelity music reproduction. Thirdly, a PAM waveform must be shaped by a reconstruct filter to smooth the shape. Though an SPCM system has three PAM-like waveforms, they are not fed through lowpass filters but fed to sub-regenerators separately. Therefore one cannot find a single filter in an SPCM/SPAM system.

The SPAM system has an **ABSTRACTOR** at the transmitting end and a regenerator at the receiving end. The abstractor has a conventional sampler and holder to sample the initial values and an unconventional sampler and holder (area sampler) to sample the area values. The area sample consists of a waveform segment integrator and a conventional sampler and holder to sample the output of the integrator at the end of the being-sampled waveform segment. The initial value is delayed one time unit purposedly in order to compute w_1 and w_2 with the area sample and next initial value. Thus the abstract sample vectors of w's are also delayed one time unit. If the linear phase lag is very important to some systems, this mechanism can serve the purpose of precise linear phase correctly but a PCM system cannot. The computation rules are very simple as those depicted in the above equations. All of the arithmetic operations and the integrations can be performed by operational amplifiers.

At the receiving end the receiver distributes w_0 , w_1 and w_2 to each sub-regenerator. There is no complicate operations here. Only three signal multiplications and one addition are needed. The multiplications can either be performed by three independent multiplication IC's or an operational amplifier circuit. Afterward the three outputs of the sub-regenerators are simply added by a resistive adder. The output of the adder will be the desired reproduction of the input analog signal at the front end of the transmitter. Since the multiplication of w_0 and U(t) is everwhere equal to w_0 except at the turn-on instant, so actually only two multipliers and one adder are needed for the design of this re-enerator. Theoretically there are three sub-regenerators in the structure of a **REGENERATOR** of an SPAM system but practically only two sub-regenerators are required. The sub-regenerator of w_0 is only the power switch of an SPAM/SPCM system. A possible block diagram of SPAM is shown in Fig. 1. In Fig. 2 a piece of Johann Sebastian Bach's music played by a famous pianist, Wolf Harden, was compared with its reproduction after digital transmission of the SPCM system in Chung Yuan Christian University. One can find that there is only 5 microsecond delay and the waveformas are exactly the same. Both the original and the regenerated waveform are recorded by a BBC's digital storage oscilloscope as shown in Fig. 2.

CONCLUSION

On the eve of coming information era, the integration of analog and digital system becomes so important and urgent. Since a PCM system is only effective for low-quality digital transmission, an effective way other than the PCM must be devised in order to keep pace with other improvements. The high-quality digital transmission will be more and more popular for the daily lives and business. The SPCM system is the realized new hope for ISDN. The second generation of SPCM is also under developing stage and has been sponsored by various branches of the government of ROC. The development of the second generation of SPCM includes the researches of a study of shaping functions and a new digital modulation technique called shape modulation in addition to code modulation and pulse modulation for data transmission thru an analog chnnel. Finally, the author would like looking forwad to a **symbolic computer of WA.**

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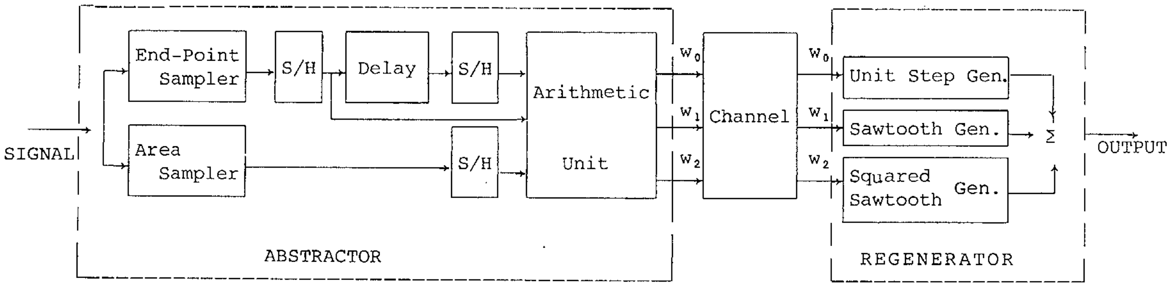

Fig. 1: The SPAM System

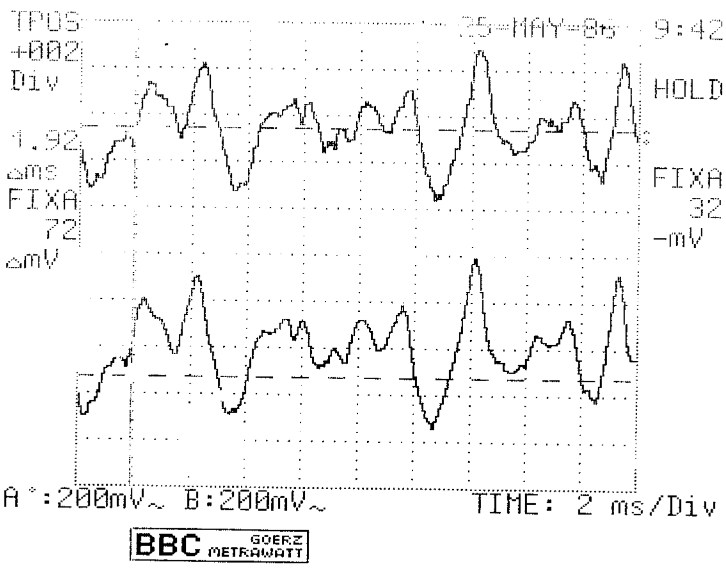

Fig. 2: A Comparison of Music Signal and Its Regenerated Signal by an SPAM System, the Orignal Music is on the Top.