SAMPLING FREQUENCY SELECTION SCHEME FOR A MULTIPLE SIGNAL RECEIVER USING UNDERSAMPLING

Yoshio Kunisawa (KDDI R&D Laboratories, yokosuka, kanagawa, JAPAN; kuni@kddilabs.jp)

ABSTRACT

A multi-mode terminal is required in order to offer users multiple communication services for ubiquitous communication and Software Defined Radio (SDR) technology has attracted considerable attention as a means of realizing a multi-mode terminal. In such a terminal, a multi-band receiver is required, since each communication service uses an individually specific frequency and bandwidth. It is also desirable to be able to utilize multiple communication services simultaneously at multi-mode terminals in order to offer various applications. Undersampling is a potential method allowing multi-band and multiple signals to be simultaneously received by selecting a suitable sampling frequency. An undersampling technique that can lower the sampling frequency for receiving two signals, compared with the conventional method has been proposed [1]. This paper presents a novel sampling frequency selection scheme, which improves the BER (Bit Error Rate) characteristics further at a multiple signal receiver using undersampling.

1. INTRODUCTION

Multiple communication services are desired at mobile terminals for various applications. SDR technology has been studied for realizing such terminals [2], [3]. Signal processing for wireless communication is usually executed by hardware such as an ASIC (Application Specific IC) or a custom IC because a lot of processing power is required to execute signal processing. In recent years, processing devices such as DSP, FPGA and other reconfigurable processors that have the flexibility to vary signal processing, have become high performance and low power consumption. For these reasons, such flexible device are used in SDR-based terminals as the signal processor. To extend the digital processing area and make the terminal applicable to many communication systems, it is desirable to put the ADC (Analog-to-Digital Converter) nearer the antenna part of receiver block diagrams. In conventional baseband sampling receivers, some analog devices such as oscillators, mixers or filters restrict the flexibility of the receiver, because the parameters or values are fixed to receiving a specific system, and cannot be changed in such devices. To make a receiver more flexible, it is desirable to sample and digitize the received signal at the RF frequency. The RF sampling scheme expands the region of the digital signal processing, and makes the receiver more programmable. However, a high speed ADC is required for the RF sampling receiver because the ADC needs to sample the high frequency signal and convert it to a digital signal. As a potential solution, an undersampling scheme, in which the frequency conversion and sampling are performed simultaneously using a sampling frequency less than the Nyquist frequency of the sampled signal, is considered [4]-[6]. In Reference [4], the computational method for the sampling frequency of RF undersampling was proposed to sample and receive the signals of multiple communication systems simultaneously using single ADC. The paper presents the conditions needed to avoid the influences of interference from the aliasing signal of the other systems. Using the proposed schemes, individual channels from multiple communication systems can be received by one ADC. However, the sampling frequency becomes higher when receiving the signal of a communication system that occupies a wide bandwidth. In Reference [1], a sampling frequency selection scheme is proposed to solve the aforementioned problem. The scheme offers a sampling frequency selection method according to the bandwidth of a transmitting channel, and is not heavily dependent on system bandwidth. As a result, the sampling frequency becomes lower.

In this paper, a novel sampling frequency selection scheme, which improves the BER (Bit Error Rate) characteristics further at a multiple signal receiver using undersampling, is proposed and evaluated by computer simulation.

2. UNDERSAMPLING [3]-[5]

Undersampling is a sampling technique that utilizes the aliasing signal caused by using a sampling frequency lower than the Nyquist frequency. Using undersampling, the signal is simultaneously sampled and converted to low frequency. Figure 1 shows an example of a block diagram

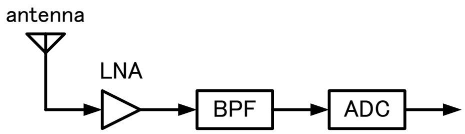
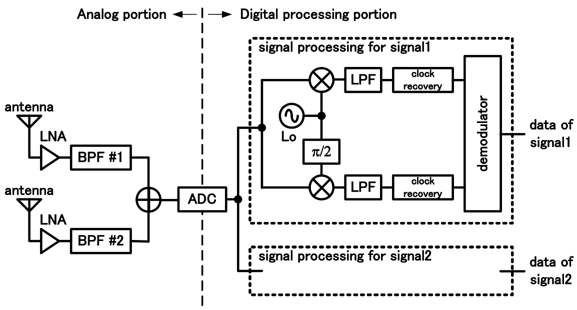
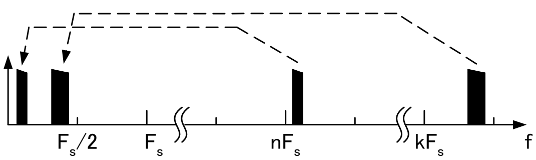


Fig. 1. Block diagram of RF undersampling receiver.



(a) Receiver structure example for undersampling multiple system signals.



(b) Conceptual view of spectrum.

Fig. 2. Undersampling for multiple system signals.

of an RF undersampling receiver. The signal received at the antenna is amplified by the LNA (Low Noise Amplifier). Thereafter, only the desired signal is passed by the BPF (Band Pass Filter). Using an appropriate sampling frequency F_S, the desired signal is frequency-converted to a frequency range between 0 and $F_S/2$. Out-band interference signals and noises that cannot be rejected by the BPF are also frequency-converted to the same frequency range. Thus, by undersampling, the signal can be frequency-converted without oscillators or mixers that usually obstruct the receiver flexibility for the receivable frequency range. Here, the relation between the desired signal's center frequency F_C and the frequency-converted signal's center frequency F_{IF} is indicated by Eq. (1).

$$\text{if} \ \ \mathrm{fix}\left(\frac{F_C}{\frac{F_S}{2}}\right) \ \text{is} \begin{cases} \text{even}, & F_{IF} = \mathrm{rem}(F_C, F_S) \\ \text{odd}, & F_{IF} = F_S - \mathrm{rem}(F_C, F_S) \end{cases} \tag{1}$$

In Eq. (1), fix(a) is a function used to obtain the value after omitting the decimal places of the value a, and rem (a, b) is a function to obtain the remainder of the value a divided by the value b. The center frequency of the signal after frequency-conversion by undersampling is obtained by Eq. (1). An appropriate sampling frequency by which the entire bandwidth of the desired wireless system is converted to a frequency range between 0 and $F_S/2$, should be selected to avoid interference from the aliasing signal of the desired system. Such a sampling frequency F_S and the frequency-converted signal's center frequency F_{IF} are required to satisfy both Eqs. (2) and (3).

$$0 < F_{IF} - \frac{BW}{2} \tag{2}$$

$$F_{IF} + \frac{BW}{2} < \frac{F_S}{2} \tag{3}$$

In Eqs. (2) and (3), BW indicates the bandwidth of the desired system. The undersampling scheme is applicable when the sampling frequency F_S satisfying Eqs. (2) and (3) exists.

To sample the signals of multiple systems simultaneously using a single ADC, the signals must be pre-bandpass filtered, and the frequency bandwidths of the signals restricted. As an example, the structure of a receiver in which the signals of multiple systems are sampled simultaneously is shown in Fig. 2, as well as a conceptual view of the spectrum.

First, the center frequency of each system after the frequency-conversion is calculated by Eq. (1). Then the sampling frequency, by which the signals of each system avoid interferences from the system itself, is selected by finding the value satisfying Eqs. (2) and (3). A condition should be added for sampling the signals of multiple systems simultaneously, and the sampling frequency is selected. Equation (4) is the condition for sampling the signals of two systems without mutual interference.

$$\left|F_{IF_1} - F_{IF_2}\right| \geq \frac{BW_1 + BW_2}{2} \tag{4}$$

The subscript in Eq. (4) is the number identifying the two systems. For example, F_{IF1} indicates the frequency-converted center frequency of System-1. Similarly, BW_1 indicates the bandwidth of System-1. Equation (4) can be expanded to be applicable for N systems. Equation (5) is the condition for sampling the signals of N systems by single ADC simultaneously.

$$\left|F_{IF_b} - F_{IF_a}\right| \geq \frac{BW_b + BW_a}{2} \qquad \begin{cases} a=2, \cdots, N \\ b=1, \cdots, a-1 \end{cases} \tag{5}$$

The subscripts a and b in Eq. (5) identify the systems' number. By applying Eqs. (1)-(3) and (5), the sampling frequency that can be used for sampling the signals of N systems is selected.

3. SAMPLING FREQUENCY SELECTION FOR MULTIPLE SYSTEMS

The signals of entire multiple systems can be sampled using the conventional sampling frequency selection scheme described in the previous chapter. However, the sampling frequency more than doubles the sum of all systems' bandwidths to satisfy Nyquist's sampling theorem. Therefore, the scheme has a problem in that the sampling frequency becomes high when the bandwidth of any receiving system is wide or the receiving of many systems is desired.

To solve the problem, a novel sampling frequency selection scheme was proposed[1]. The scheme offers a sampling frequency selection method according to the bandwidth of a transmitting channel, and is not heavily dependent on system bandwidth. As a result, the minimum sampling frequency becomes lower compared to the conventional scheme described in the previous chapter.

F_{IF} and F_{IFch}, the center frequencies of the system and the desired channel after frequency-conversion respectively, are revealed by Eqs. (1) and (6).

$$\text{if fix}\left(\frac{F_{ch}}{\frac{F_S}{2}}\right) \text{ is } \begin{cases} \text{even}, & F_{IF_{ch}} = \mathrm{rem}(F_{ch}, F_S) \\ \text{odd}, & F_{IF_{ch}} = F_S - \mathrm{rem}(F_{ch}, F_S) \end{cases} \quad (6)$$

In Eq. (6), F_{ch} indicates the center frequency of the desired channel before frequency-conversion. The center frequency of the desired channel is revealed by Eq. (6). In order to receive and demodulate the signal of the desired channel, the frequency-converted signal of the desired channel must not be influenced by any aliasing signals. Therefore, it is necessary to select a sampling frequency satisfying Eqs. (7) and (8).

$$F_{IF_{ch}} - \frac{BW_{ch}}{2} > -\left(F_{IF} - \frac{BW}{2}\right) \quad (7)$$

$$\frac{F_S}{2} - \left(F_{IF_{ch}} + \frac{BW_{ch}}{2}\right) > \left(F_{IF} + \frac{BW}{2}\right) - \frac{F_S}{2} \quad (8)$$

In Eqs. (7) and (8), BW_{ch} indicates the bandwidth of the desired channel. Using a sampling frequency satisfying Eqs. (7) and (8), the signal of the desired channel is received without being subject to the influence of the aliasing image.

In addition to the constraint of the sampling frequency described above, it is necessary to select a sampling frequency by which the frequency-converted signals of each desired channel are not influenced by the aliasing signals of other systems, to receive and demodulate every signal of each desired channel. Here is an example of receiving each signal of two channels in two systems. Firstly, select the sampling frequency by which each system and each desired channel satisfy Eqs. (1) and (6)-(8). Then pick up the sampling frequency by which the signals of each desired channel are not influenced by the signals of the aliasing images of other systems, using Eqs. (9) and (10).

$$\left|F_{IF_{ch_2}} - F_{IF_1}\right| \geq \frac{BW_{ch_2} + BW_1}{2} \quad (9)$$

$$\left|F_{IF_2} - F_{IF_{ch_1}}\right| \geq \frac{BW_2 + BW_{ch_1}}{2} \quad (10)$$

By the sampling frequency satisfying both Eqs. (9) and (10), the signals of the desired channels in each system can be undersampled.

Furthermore, by select a sampling frequency so that the frequency difference (ΔF) of the center frequencies of the two desired signals may increase, the BER (Bit Error Rate) characteristics improve at a multiple signal receiver using undersampling. So, this paper proposes selecting such sampling frequency from the sampling frequencies satisfying this process.

4. EVALUATION OF BER CHARACTERISTICS

In this chapter, the effect of increasing the frequency difference of the center frequencies of the two desired signals, is evaluated by computer simulation.

4.1. System model

The system model used to evaluate the effect of increasing the difference of the center frequencies of the two desired signals is shown in Fig. 4. Signal-1 which is one of the channels of system-1, and signal-2 which is one of the channels of systems-2 are generated and mixed before an ADC. Sampled data using undersampling scheme is demodulated and compared with transmitted data. Pseudorandom data is used for the simulation and QPSK modulation is used for signal-1 and signnal-2 to simplify the simulation. In this simulation, the frequency parameters of system-1 are assumed to be the cdma2000 system of band class 3, and the frequency parameters of system-2 are assumed to be the WiMAX system of 2.5GHz band. So, the parameters, F_{C1}=800MHz, F_{C2}=2.5GHz, BW_1=10MHz, BW_2=20MHz, BW_{ch1}=1.25MHz, BW_{ch2}=10MHz are used in the simulation. Here, the subscript number n corresponds to the signal number or the system number. And, F_{cn}, BW_n and

BW_{chn} indicate the center frequency of the desired signal-n, the system bandwidth of the system-n and the signal bandwidth of the signal-n, respectively. Assuming that the desired signal is assigned to just center of the corresponding system. The parameters are summarized in Table 1.

4.2. Sampling frequencies for sampling two desired signals

Figure 4 (a), (b) show the relation between the sampling frequency and the center frequency of the signals after frequency-conversion by the conventional scheme and the proposed scheme in Reference [1], respectively. In Fig. 4 (a), (b), the horizontal axis indicates the sampling frequency, and the vertical axis indicates the center frequency after frequency-conversion. For example, by undersampling the

Table 1. Simulation parameters

	Signal-1	Signal-2
System assumption	cdma2000 (band class 3)	WiMAX
Center frequency	800 [MHz]	2.5 [GHz]
Signal bandwidth	1.25 [MHz]	10 [MHz]
System bandwidth	10 [MHz]	20 [MHz]
Modulation type for simulation	QPSK	QPSK
LPF	root nyquist α=0.5	root nyquist α=0.5

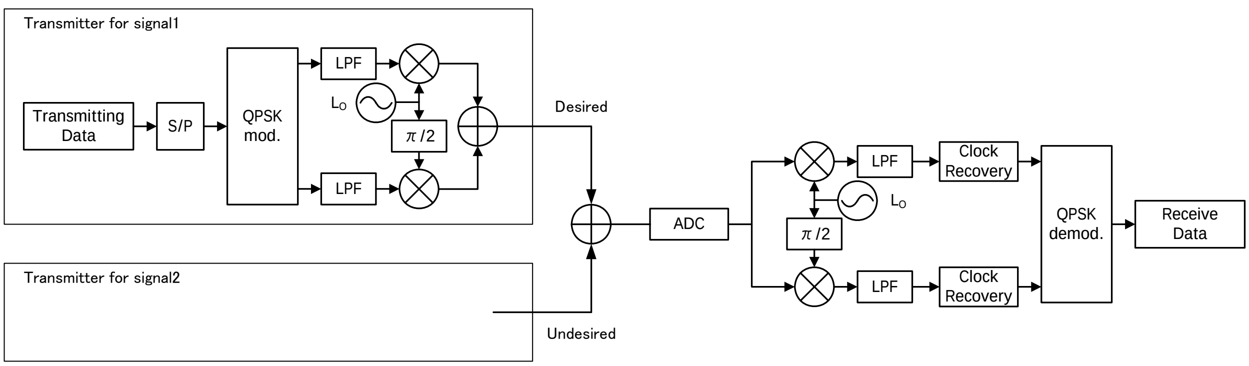
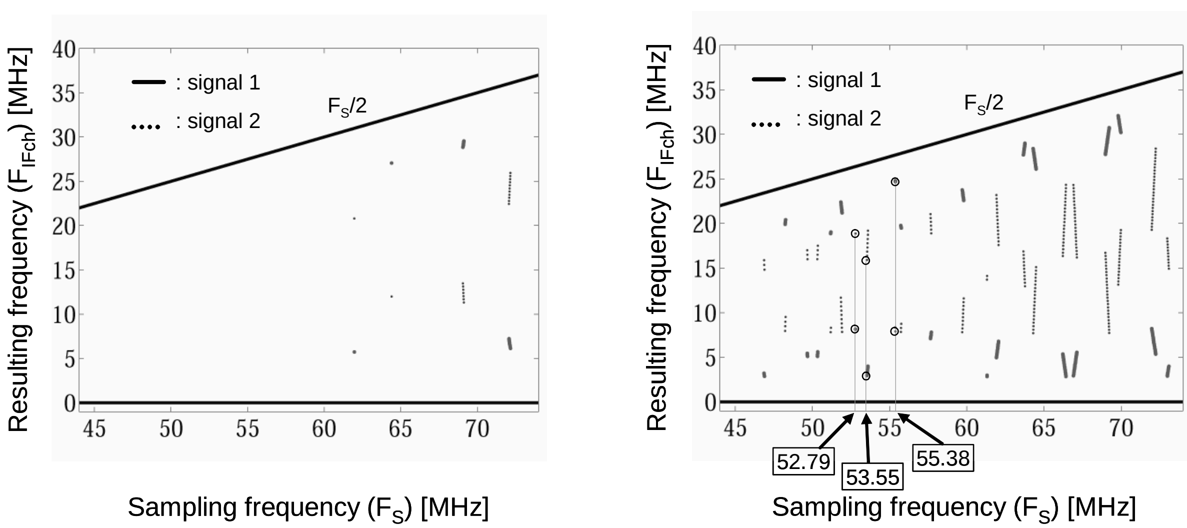


Fig. 3. System model in simulation



(a) Conventional. (b) Proposed [1].

Fig. 4. Relation between sampling frequency and frequency after frequency-conversion. (Sampling frequency between 45MHz and 75MHz)

signal using a sampling frequency of 69.1 MHz, signal-1 is converted to a frequency of 29.2 MHz, and signal-2 is converted to a frequency of 12.4 MHz.

4.3. Evaluation of BER characteristics

According to the calculation in the previous chapter, the available sampling frequency Fs=52.79 MHz, 53.55 MHz and 55.38 MHz can be found for undersampling. And the center frequency of two signals can be calculated as Table 2. The relation of the sampling frequency and the signals are also indicated in Fig. 4 (b). Consequently, the frequency difference (ΔF) of center frequency of the two desired signals become 10.72 MHz, 13.60 MHz and 16.78 MHz, respectively.

Figure 5 (a) and (b) shows the BER characteristics of signal-1 in the cases of D/U (Desired-to-Undesired signal power ratio), = -45 dB and -50 dB, respectively. Thus, the signal level of signal-1 is assumed to be lower than that of signal-2. Moreover, signal-1 is investigated as a desired signal in this simulation, because the low level signal is influenced far more by the high level signal. From Tabel 2, ΔF becomes 10.72MHz when the sampling frequency Fs=52.79 MHz. ΔF can be enlarged only by Fs expanding slightly. Figure 5 indicates that the BER characteristics can be improved if the sampling frequency is suitably selected so that ΔF becomes large. This is because the amount of interference from signal-2 decreases if ΔF increases.

The signal transmitted from a practical transmitter, has an out-band spectrum due to the limited linearity of the power amplifiers or the mixers in the transmitter. So the desired channels are influenced from the out-band of another desired signal. The spectrum component of the out-band signal typically becomes so small that its frequency separates from the center frequency. Therefore, by selecting

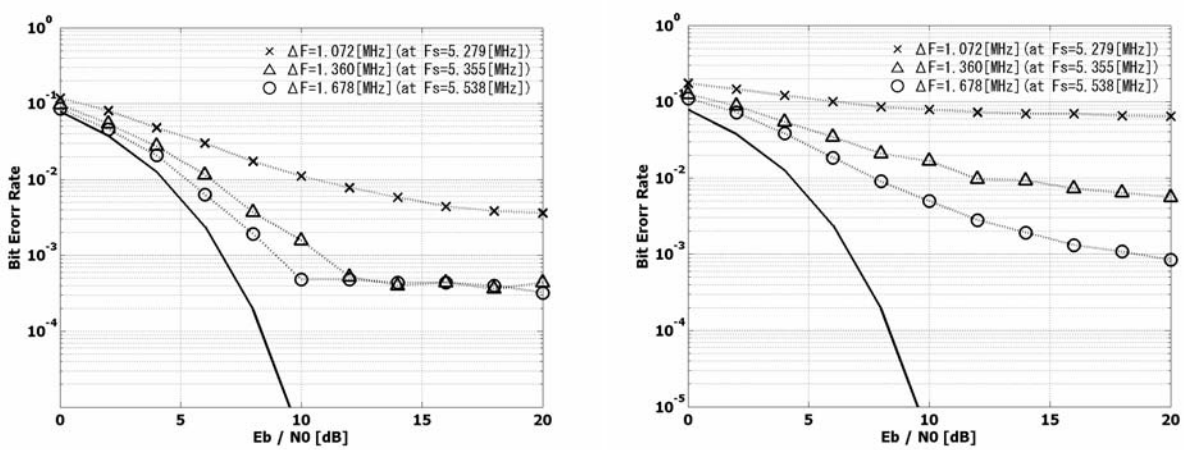
Table 2. Center frequency of frequency converted signals

Sampling frequency (F_S)	52.79 [MHz]	53.55 [MHz]	55.38 [MHz]
Center frequency of signal-1 (F_{IFch1})	8.15 [MHz]	3.25 [MHz]	24.68 [MHz]
Center frequency of Signal-2 (F_{IFch2})	18.87 [MHz]	16.85 [MHz]	7.90 [MHz]
Frequency difference (ΔF)	10.72 [MHz]	13.6 [MHz]	16.78 [MHz]

a sampling frequency so that the frequency difference (ΔF) of the center frequencies of the two desired signals may increase, the BER (Bit Error Rate) characteristics improve at a multiple signal receiver using undersampling. This scheme can expand the tolerance of signal level difference without greatly increasing the sampling frequency in multiple signal receivers using undersampling.

5. CONCLUSION

This paper describes a sampling frequency selection scheme to undersample the signals of multiple systems in different frequency bands, simultaneously. And this paper proposed to select effective sampling frequency from the available sampling frequencies for undersampling, to improve the BER characteristics. The computer simulation confirms the improvement of the BER characteristics if the frequency difference of the center frequencies of the desired signals increases. Furthermore, the tolerance of the signal level



(a) D/U = -45 [dB] (b) D/U = -50 [dB]

Fig. 5. BER characteristics

difference can be expanded by the proposed scheme. Consequently the requirement to the quantization bits of the ADC can be reduced. The sampling frequency can be selected from the frequencies available for undersampling as if the frequency difference of the converted signals become as large as possible, within reasonable value of the sampling frequency. This scheme is particularly effective in future broadband wireless systems where a wide dynamic range is required for the receivers and multiple communication systems are used cooperatively.

6. REFERENCES

[1] Y. Kunisawa, "Study on a Multiple Signal Receiver using Undersampling Scheme," 2005 Software Defined Radio Technical Conference & Product Exposition.

[2] Joe Mitola, "The Software Radio Architecture," IEEE Communications Magazine, Vol. 33, No. 5, pp. 26-38, May 1995.

[3] K. Araki, "Software Defined Radio and Its Applications," Sipec, 2002.

[4] D. Akos, M. Stockmaster, J.B.Y. Tsui, and J. Caschera, “Direct bandpass sampling of multiple distinct RF signals,” IEEE Trans. Commun., Vol. 47, No. 7, pp. 983-988, July 1999.

[5] G. Hill, “The benefits of undersampling,” Electron. Design, pp. 69-79, July 1994.

[6] Vaughan, R.G. Scott, N.L. White, and D. Rod, “The theory of bandpass sampling,” IEEE Trans. Signal Processing, Vol.39, pp.1973-1984, Sept. 1991.