ANALOG-DIGITAL SIGNAL PROCESSING FOR MULTI-CHANNEL RECEPTION

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ABSTRACT

In this paper an analog-digital signal processing scheme for multichannel signal reception with low-IF receivers is proposed and its performance is investigated. In the low-IF receivers, the signal in the mirror frequency causes interference to the desired signal. In the proposed analog-digital signal processing scheme, the interference signal is extracted with the analog filter and the interference to the desired signal is reconstructed by LSM algorithm. The interference is cancelled based on the reconstructed signal. Through computer simulation, it is shown that the proposed scheme can reduce the influence of the interference by about 5 [dB].

1. INTRODUCTION

Recently, access points for wireless LANs have been installed in many places such as airports or hotels. Though the roaming capability has been specified in the IEEE802.11 standard, it is not able to handle a quick roaming for VoIP applications. Therefore, multi-channel reception is required for VoIP over WLAN.

One of the receiver architecture suitable for such applications is the low-IF receiver [1][2]. In the low-IF receiver, the IF is set to be relatively higher than that in conventional IF receivers. The IF signal is sampled and converted to the digital signal with analog-digital converters (ADCs). The final process of down conversion is carried out in the digital domain. This architecture is suitable for multi-channel reception as the choice of the channel can be done with digital signal processing. However, in some cases the next access point may be far away from the current one and the dynamic range between the signals from the current access point and those from the next one may be quite large. This means that high resolution ADCs have to be employed to accommodate such a signal with large dynamic range. The increase of the resolution of the ADC causes higher power consumption and higher implementation cost.

In order to reduce the required resolution of the ADCs, an analog-digital signal processing technique has been proposed [4-5]. The proposed technique uses a band pass filter (BPF) for each WLAN channel. The BPFs reduce the adjacent channel interference (ACI) and ease the dynamic range of the ADCs. Neverthless, the problem of the analog BPFs is that it cannot eliminate the interference completely due to the restriction of the circuit size and the mismatch of the analog components. Thus, combination of the analog and digital signal processing is indispensable. The proposed analog-digital signal processing technique first approximates the characteristics of the analog filters to FIR filters and estimates the errors of the analog filters. It then compensates for the errors of the analog filters by adaptive digital signal processing. The results obtained from computer simulation show that the proposed technique enables multichannel reception with the low resolution ADCs.

2. MULTI-CHANNEL RECEPTION

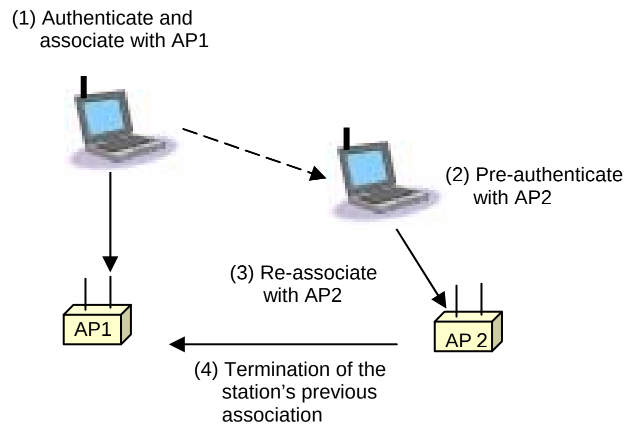

Fig.1 Roaming Example with IEEE802.11 MAC Protocol.

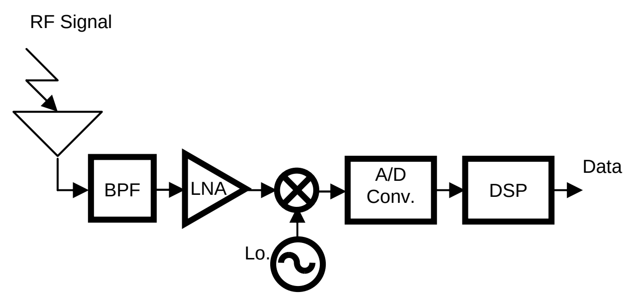

Fig.2 Low-IF Receiver Architecture.

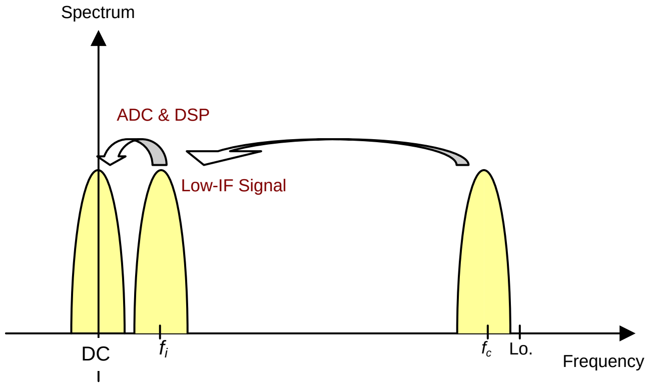

Fig.3 Down Conversion of the Received Signal in Low-IF Receiver Architecture.

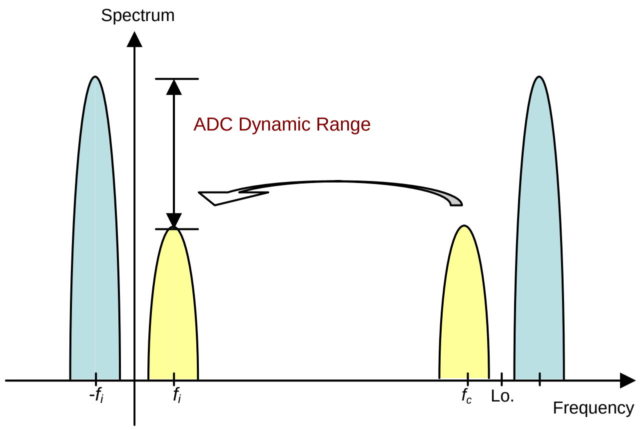

Fig.4 Interference due to the Signal in the Adjacent Channel

The mobility of WLAN terminals among multiple base stations is specified in Extended Service Set (ESS) of IEEE802.11 MAC protocol [3]. An example of roaming capability with IEEE802.11 MAC is shown in Fig. 1. As the terminal finds AP1, it will authenticate and associate with AP1. As the terminal moves, it may pre-authenticate with AP2. When the terminal determines that its association with AP1 is no longer desirable, it may reassociate with AP2. The reassociation causes AP2 to notify AP1 of the new location of the station, terminating the terminal's previous association with AP1.

In general, the terminal can be authenticated with many different stations simultaneously. However, it may be associated with only one base station at a time. Therefore, it is not suitable for quick roaming though some applications such as VoIP over WLAN requires.

In order to solve this problem, multi-channel reception capability is required. For example, in IEEE802.11b WLAN systems, 4 channels in 2.4 GHz band are utilized to cover large area for the services. Thus, if these 4 channels can be demodulated by one receiver, the quick roaming may be possible and the mobile VoIP service over WLAN can be provided.

There are several receiver architectures suitable for multiple channel reception. The simplest architecture is to combine 4 independent receivers in one package. However, this architecture has large redundancy in their circuits. Another candidate is the low-IF receiver. The structure of the low-IF receiver is shown in Fig. 2. The received signal is first down converted to the IF signal as shown in Fig. 3. Then the IF signal is converted to digital signal with ADCs and finally converted to the baseband signal with DSP. As the down conversion is carried out with DSP, it is possible to select one of 4 channels easily.

However, if the signal power in the adjacent channel is much larger than the desired signal, ADCs with very high dynamic range are required as shown in Fig. 4. This is not desirable as the large dynamic range leads to higher cost and power consumption of the ADCs.

3. PROPOSED ANALOG-DIGITAL SIGNAL PROCESSING SHCEME

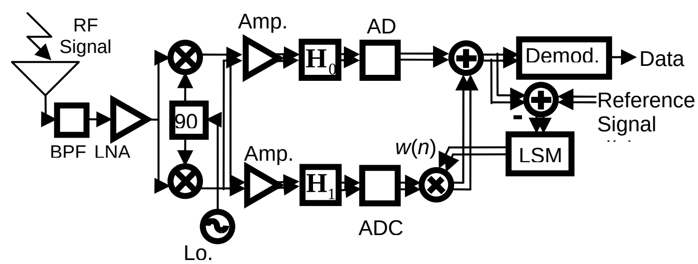

Fig. 5 Model of the Receiver with the Proposed Scheme

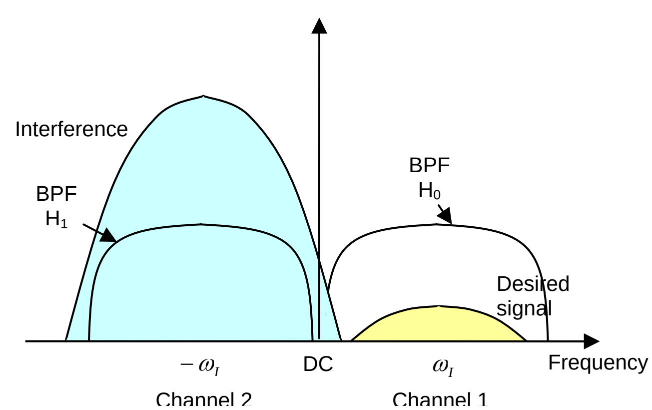

Fig. 6 Received Signal Model

In order to reduce the dynamic range of the ADCs, the analog-digital signal processing is utilized. The model of the receiver with the proposed scheme is shown in Fig. 5[4].

In [4-5] it has been shown that the analog filter bank can reduce the dynamic range of the ADCs. However, due to the restriction of the circuit size and the mismatch of the analog components, it is hard to realize high Q analog filter.
Here, in addition to the analog filter bank, adaptive digital signal processing is utilized to further reduce the required dynamic range of the ADCs. The received signal is first goes through the RF BPF and LNA. The output of the LNA is then multiplied with the local signal and converted to the IF. With the analog filters, H_0 and H_1, the signals on the difference channels are separated. However, if the power of the signal on the adjacent channel is large, it causes the interference to the desired signal. In order to reduce the adjacent channel interference, the analog filter bank and the adaptive digital signal processing are employed.

Suppose that the desired signal and the interference signal are received at the same time as shown in Fig. 6. The received signal is expressed as

$$\begin{aligned} r(k) &= d(kT_s)\exp(j\omega_I kT_s) \\ &\quad + I(kT_s)\exp(-j\omega_I kT_s) + n(kT_s) \end{aligned}, \qquad (1)$$

where $r(k)$ is the *k*-th sample of the received signal, T_s is the sampling interval, *d* is the desired signal, *I* is the signal on the adjacent channel, ω_I is the intermediate frequency of the desired signal, $-\omega_I$ is the frequency of the interference signal, and *n* is the noise. The received signal is put into the analog filters and converted to the digital signals.

$$y_0(n) = adc\left\{\sum_{k=0}^{L-1} h_0(k) r(n-k)\right\} , \qquad (2)$$

$$y_1(n) = adc\left\{\sum_{k=0}^{L-1} h_1(k) r(n-k)\right\} , \qquad (3)$$

where y_m is the output of the ADC for *m*-th channel, $adc\{X\}$ represents the analog-to-digital conversion of *X*, $h_m(k)$ is the *k*-th coefficient of the *m*-th filter.

In the training period, the reference signal $s(n)$ is given to train the coefficient, *w*, of the adaptive filter. The error between the received signal and the reference signal is given as

$$e(n) = y_0(n) - w(n) y_1(n) - s(n) \quad . \qquad (4)$$

The LMS algorithm is used to update the coefficient of the canceller as

$$w(n+1) = w(n) + \mu y_0(n) e^*(n) \quad , \qquad (5)$$

where μ is the step size. In the data reception period, the adjacent interference is canceled with the trained coefficient *w* and is given by

$$d(n) = y_0(n) - w y_1(n) \quad . \qquad (6)$$

The desired signal on the IF is then demodulated and decoded in the digital domain.

4. NUMERICAL RESULTS

Table 1 Simulation Conditions

Channel Model	AWGN Channel
Modulation Scheme	QPSK/OFDM
Number of Subcarriers	64
IF	$\pi / 2$
Number of Stages of Analog Filter	5
Number of Coefficients	31
Step Size	10^{-4}
Number of Bits	1280000
Training Period	128000

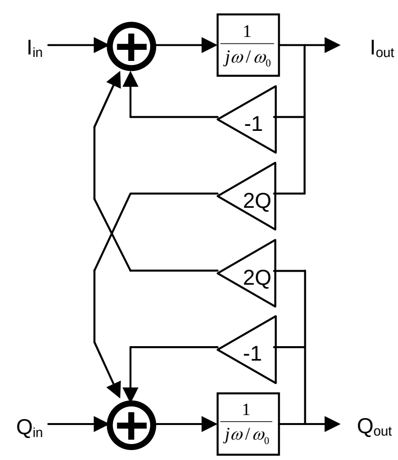

Fig. 6 Model of the Analog Filter

The performance of the proposed analog-digital signal processing scheme is investigated through computer simulation. The simulation conditions are shown in Table 1. The model of the complex analog band pass filter is shown in Fig. 6 [2]. The transfer function of the filter is given as

$$H_{bp}(j\omega) = \frac{1}{1-2\,jQ+\omega/\omega_0} \quad , \qquad (7)$$

where ω_0 is the cut off frequency of the filter, and $2Q$ is the center frequency of the BPF. 2Q is set to ω_I and $-\omega_I$ for the desired signal and the interference signal. The number of the serial stages of the analog filter is 5. In order to simulate the analog BPFs in the digital domain, the digital filters whose frequency response in every $\Delta f = 2\pi / 31$ is equal to that of the analog filter are constructed.

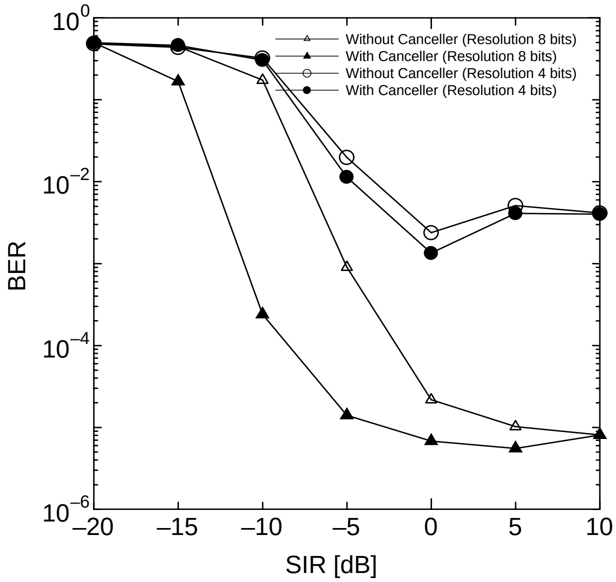

Fig. 7 BER vs. SIR, Eb/No=10 [dB].

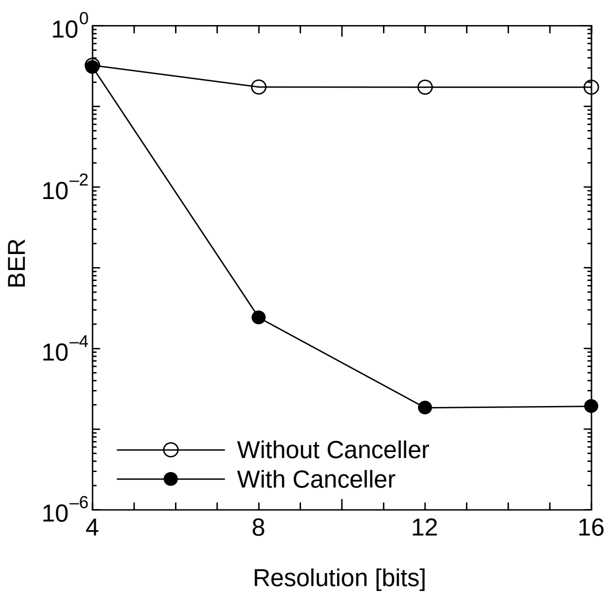

Fig. 8 BER vs. Resolution, SIR=-10[dB], Eb/No=10 [dB].

Fig. 7 shows the BER versus the SIR with and without the adaptive digital signal processing in the proposed receiver architecture. When the resolution of the ADCs is 4 bits, there is not much difference in the BER performance between the proposed adaptive digital signal processing scheme and the conventional analog filter bank. The reason is that the reconstructed interference through the adaptive signal processing includes the error be due to the quantization noise. On the other hand, if the resolution of the ADCs is 8 bits, and if the SIR is less than -10 [dB], the proposed scheme effectively cancells the interference and can improve the BER performance.

Fig.8 shows the BER versus the resolution of ADCs. It is clear that the canceller for the adjacent channel interference works well. It is also clear that 12 [bits] is enough for the resolution of the ADCs in the proposed scheme.

5. CONCLUTIONS

In this paper, the novel analog-digital signal processing scheme for multichannel signal reception has been proposed. It has been shown that the proposed scheme can mitigate the influence from the adjacent channel and enables multichannel reception with relatively low ADCs with the adaptive digital signal processing. Therefore, the proposed scheme can be applied to VOIP services with WLANs.

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