Analog-Digital Signal Processing for Multi-Channel Reception

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SUMMARY 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 LMS algorithm.

**key words:** low-IF receiver, filter bank, AD converter, software radio

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]. 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 signals with large dynamic range. In addition, as the WLAN channels are closely assigned, the signal on the adjacent channel will cause interference to the desired signal.

In order to reduce the required resolution of the ADCs and eliminate the interference, an analog-digital signal processing technique has been proposed [3], [4]. The proposed technique uses band pass filters (BPFs) for each WLAN channel. The BPFs reduce the power of the interference over the adjacent channel and ease the dynamic range of the ADCs. The problem of the proposed scheme is that it requires estimating the characteristics of the analog filters. While the estimation the receiver can not receive the signal that causes the interference to the desired signal.

LSM algorithm is then used in this paper to cancel the interference from the adjacent channel. LSM algorithm is less complex and can estimate the characteristics of the analog filters while it maintains the synchronization to the received signal though training sequence is required periodically. The results obtained through computer simulation show that the proposed scheme enables multichannel reception with the low resolution ADCs.

2. System Model

The model of the receiver with the proposed scheme is shown in Fig. 1. 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 adjacent 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 interference from the adjacent channel, an analog-digital signal processing scheme, the interference signal is extracted with the analog filter and the interference to the desired signal is reconstructed by LMS algorithm.

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

\[
r(t) = d(t) \exp(j\omega_1 t) + I(t) \exp(-j\omega_1 t) + n(t)
\]

where \(r\) is the received signal, \(d\) is the desired signal, \(I\) is the signal on the adjacent channel, \(\omega_1\) is the intermediate frequency of the desired signal, \(-\omega_1\) is the frequency of the interference signal, and \(n\) is the noise. The received signal is put into the analog filters.

\[
y_0(t) = \int_{-L/2}^{L/2} h_0(\tau)r(t-\tau)d\tau,
\]

\[
y_1(t) = \int_{-L/2}^{L/2} h_1(\tau)r(t-\tau)d\tau
\]

**Fig. 1** Model of the receiver with the proposed scheme.
where \( y_m \) is the output of the ADC for \( m \)-th channel, \( h_m \) represents the response of the \( m \)-th analog filter. The output of the ADCs are given as

\[
\hat{y}_0(n) = \text{adc}(y_0((n−1)T_s)), \tag{4} \\
\hat{y}_1(n) = \text{adc}(y_1((n−1)T_s)). \tag{5}
\]

In the training period, the reference signal \( s(n) \) is given to train the coefficients of the FIR filter, \( w(n) = [ w_0(n) \ w_1(n) \ldots \ w_{L−1}(n) ]^T \). The error between the received signal and the reference signal is given as

\[
e(n) = y_0(n) − w^H(n)\hat{y}_1(n) − s(n) \tag{6}
\]

where \( \hat{y}_1(n) = [ \hat{y}_1(n) \ \hat{y}_1(n−1) \ldots \ \hat{y}_1(n−L+1) ]^T \). The LMS algorithm is used to update the coefficients of the FIR filter as

\[
w(n+1) = w(n) + \mu\hat{y}(n)e^*(n). \tag{7}
\]

In the data reception period, the adjacent interference is canceled with the FIR filter and is given by

\[
d(n) = y_0(n) − w^H\hat{y}_1(n). \tag{8}
\]

### 3. Numerical Results

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 given [2]. The transfer function of the filter is given as

\[
H_{bp}(j\omega) = \frac{1}{1−2jQ + j\omega/\omega_0} \tag{9}
\]

where \( \omega_0 \) is the cut off frequency of the filter, and \( 2Q = \omega_1/\omega_0 \) is the relative center frequency of the BPF. In this simulation \( 2Q \) is set to \( \omega_1/\omega_0 \) and \( −\omega_1/\omega_0 \) for the desired signal and the interference signal. \( \omega_1 \) and \( \omega_0 \) are set to \( \pi/2 \). The number of the serial stages of the analog filter is 5. As a result, the suppression ratios of the filters, \( H_0 \) and \( H_1 \), to the adjacent channels are the same, which is about 15 dB at \( \pi/2 \) away from the center frequency. This means that the interference in the mirror frequency with the SIR of up to about

\(-15 \text{ dB} \) can be suppressed with this analog filter. 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.

Figure 2 shows the BER versus the resolution of the ADCs. Even though the analog filter can suppress the interference in the mirror frequency, the part of the interference signal overlaps with the desired signal. This interference cannot be eliminated with the analog filter and the BER without the canceller is quite large.

On the other hand, it has been shown that the canceller for the adjacent channel interference works well. It has been also clear that at least 8 [bits] are required for the resolution of the ADCs in the proposed scheme.

Figure 3 shows the BER versus the SIR. It is also shown that the proposed canceller works when the resolution of ADC is 8 [bits]. The reason is that the accuracy of the reconstructed interference improves as the resolution of

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**Table 1** Simulation conditions.

<table>
<thead>
<tr>
<th>Channel Model</th>
<th>AWGN Channel</th>
</tr>
</thead>
<tbody>
<tr>
<td>Modulation Scheme</td>
<td>QPSK/OFDM</td>
</tr>
<tr>
<td>Number of Subcarriers</td>
<td>64</td>
</tr>
<tr>
<td>IF</td>
<td>( \pi/2 )</td>
</tr>
<tr>
<td>Number of Stages of Analog Filter</td>
<td>5</td>
</tr>
<tr>
<td>Suppression Ratio [dB] @ Frequency Separation</td>
<td>-15@( \pm \pi/2 )</td>
</tr>
<tr>
<td>-35@( \pm \pi )</td>
<td></td>
</tr>
<tr>
<td>Number of Coefficients</td>
<td>31</td>
</tr>
<tr>
<td>Step Size</td>
<td>10^4</td>
</tr>
<tr>
<td>Number of Bits</td>
<td>1280000</td>
</tr>
<tr>
<td>Training Period</td>
<td>128000</td>
</tr>
</tbody>
</table>
the ADC. It is also clear that the analog filter can suppress
the interference with the SIR of up to $-5$ [dB] when the res-
olution of ADC is 8 [bits] and the BER is about $10^{-3}$. The
proposed canceller can suppress the additional interference
about 5 [dB].

4. Conclusions

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

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References

software-defined radio in handheld terminal applications,” IEEE
performance analog front ends of fully integrated receivers,” IEEE
cfficient errors of complex filter bank parallel A/D converter in low-IF