Adjacent Channel Interference Cancellation Scheme for Low-IF Receiver in Multi-Channel Reception

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SUMMARY In this paper a new adjacent channel interference (ACI) cancellation scheme for multi-channel signal reception with low-IF receivers is investigated through the experiment. 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, channel selection is made by analog complex band pass filter and the signal is reconstructed by Wiener filter to eliminate the interference effect in order to improve the performance.

key words: low-IF receiver, VoIP over WLAN, multi-channel reception, adjacent channel interference

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 applicable for such applications is the low-IF receiver [1], [2]. In the low-IF receiver, the IF is set to be relatively lower than that in conventional IF receivers. The low-IF receiver are not using SAW filter so that this architecture would have low cost and low power consumption compare to conventional super-heterodyne architecture. 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 applicable 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. Moreover the adjacent channel interference (ACI) component may directly overlap with the desired signal if the interference is much larger than desired signal.

In order to reduce the required resolution of the ADCs and reduce the interference, an analog-digital signal processing technique has been investigated [3], [4]. This technique uses a band pass filter (BPF) for each WLAN channel. The BPFs erase the dynamic range of the ADCs and makes the modulation of multi-channel possible. Nevertheless, 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 problem of this scheme is that it requires estimating the characteristics of the analog filters. This system requires the signal generator in the receiver to generate known waveform for the estimation. While the estimation the receiver can not receive the signal and loses the synchronization.

In this paper, a new ACI cancellation scheme with the analog filter bank has been proposed. The proposed scheme automatically estimates the characteristics of the interference signal and cancels it from received signal through Wiener filter. The proposed scheme can estimate the characteristics of the interference signal while it maintains the synchronization to the received signal through training sequence is required periodically. The results obtained from experiment show that the proposed technique enables multi-channel reception and work with the low resolution ADCs.

2. Multi-Channel Reception

The mobility of WLAN terminals among multiple base stations is specified in Extended Service Set (ESS) of IEEE802.11 MAC protocol [5]. 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 reassocation 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 IEEE802.11g 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 applicable 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 by using 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, in the low-IF receivers, ADCs with very high dynamic range may required if the signal power in adjacent channel is much larger than the desired signal as shown in Fig. 4. In addition the desired signal may suffer from the interference from signals which are adjacent in frequency to the desired signal. This is called adjacent channel interference (ACI). This interference occurs because the adjacent channel generates side lobe energy that falls into the pass band of the desired signal that make desired signal cannot be demodulated.

3. System Model

In order to reduce the dynamic range of the ADCs and reduce the ACI, the analog-digital signal processing is utilized. The model of the receiver with the proposed scheme is shown in Fig. 5. In [3], [4] 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. Moreover the ACI may directly overlap with the desired signal if the interference is much larger than the desired signal.

Here, in addition to the analog filter bank, adaptive digital signal processing is utilized to reduce the ACI. 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 ACI, the adaptive digital signal processing, Wiener filter, is employed. However, the proposed scheme is only required when the roaming channel is an adjacent channel. When the channel is not adjacent, the receiver will demodulate the channel without the canceller process. Since the low-IF receiver requires higher sampling-frequency than conventional architecture such as super-heterodyne, undersampling technique can be applied in this system in order to lower the required sampling frequency.

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
where \( r(t) \) is the sample of the received signal at the time \( t \), \( d(t) \) is the desired signal, \( l(t) \) is the signal on the adjacent channel, \( \omega T \) is the intermediate frequency of the desired signal, \( -\omega T \) is the frequency of the interference signal, and \( n \) is the noise. The received signal is put into the analog filters for channel selection. This can be expressed by

\[
y_0(t) = \int_{0}^{(L-1)T_s} h_0(\tau)r(t-\tau)d\tau, \tag{2}
\]
\[
y_1(t) = \int_{0}^{(L-1)T_s} h_1(\tau)r(t-\tau)d\tau \tag{3}
\]

where \( y_m \) is the output of \( H_m \) analog complex band pass filter for \( m \)-th channel, \( h_m(\tau) \) is the impulse response of the \( m \)-th filter. The output of the analog filters is then converted to digital signals.

\[
Y_0(n) = \text{adc}\{y_0((n-1)T_s)\}, \tag{4}
\]
\[
Y_1(n) = \text{adc}\{y_1((n-1)T_s)\} \tag{5}
\]

where \( Y_m \) is the output of the ADC for \( m \)-th channel, \( \text{adc}[X(t)] \) represents the analog-to-digital conversion of \( X(t) \) at the time \( t \).

Because the interference signal is much larger than the desired signal, the interference signal is still remaining in \( Y_0 \). In training period, the signal is processed by using Wiener filter to cancel the interference signal that remains in \( Y_0 \). The model of the Wiener Filter is shown in Fig. 7. The input of the Wiener filter is express by \( \bar{Y}(n) = [Y_0(n), Y_1(n), Y_1(n-1), ..., Y_1(n-M+1)]^T \), that is combination of \( Y_0 \) and \( Y_1 \) and tap coefficient vector is given by \( w(n) = [w_0(n), w_1(n), ..., w_M(n)]^T \). Autocorrelation of input signal, \( \bar{Y} \) is expressed by,

\[
R = E[\bar{Y}(n)\bar{Y}^H(n)] \tag{6}
\]

and cross-correlation of input signal, \( \bar{Y} \) and reference signal, \( s \) is expressed by

\[
p = E[\bar{Y}(n)s^*(n)]. \tag{7}
\]

This reference signal, \( s \), is created with random sequence which is known to the receiver. The optimal tap coefficient vector \( w_{opt} \) is calculated by using this equation

\[
w_{opt} = R^{-1}p. \tag{8}
\]

In the reception period, the ACI is canceled with the trained optimal coefficient and is given by,

\[
d_0(n) = w_{opt}^H(n)\bar{Y}(n). \tag{9}
\]

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

4. Experiment System

Figure 8 shows the model of the Experiment System. Table 1 shows the experiment conditions. In the experiment, IF signals are generated by using a dual-channel modulation signal generator. There are 4 outputs of the signal generator which 2 of them are I-phase and Q-phase of the signal, and the other 2 are the differential part of I/Q. The signal's data is made by MATLAB program and signal generator is used to generate the IF signal. OFDM signals with difference signal to interference ratio (SIR) are generated to investigate performance of the proposed scheme.

The output of the signal generator is goes through the analog filter for channel selection. Channel selection in the proposed scheme is made by using complex analog band pass filters, \( H_0 \) and \( H_1 \). The model of the complex analog band pass filter is shown in Fig. 9 [2]. With a complex filter it is possible to discriminate between the negative and positive frequencies and therefore, the mirror frequency will
Fig. 8 Model of the experiment system.

Table 1 Experiment conditions.

<table>
<thead>
<tr>
<th>Component</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Signal Generator</td>
<td>2ch(I,Q), 14 bit 80 Msps</td>
</tr>
<tr>
<td>Tx and</td>
<td>64 Carrier OFDM, DQPSK</td>
</tr>
<tr>
<td>Interference</td>
<td></td>
</tr>
<tr>
<td>Demodulation</td>
<td>Differential Decodes</td>
</tr>
<tr>
<td>Signal bandwidth</td>
<td>2.4 [MHz]</td>
</tr>
<tr>
<td>ADC</td>
<td>4ch, 12 bit, 10 Msps</td>
</tr>
<tr>
<td>FPGA</td>
<td>Xilinx 1M Gate Virtex II</td>
</tr>
<tr>
<td>DSP</td>
<td>167 MHz TMS320C6701 32bit Floating Point</td>
</tr>
</tbody>
</table>

Fig. 9 Model of the analog filter.

be filtered out. $H_0$ allows only positive part of certain frequencies to pass and cut off negative frequency, while $H_1$ only allows same negative frequencies to pass. The transfer function $H_0$ and $H_1$ is given as

$$H_0(j\omega) = \left(\frac{1}{1 - 2jQ + j\omega_0/\omega_I}\right)^n,$$

$$H_1(j\omega) = \left(\frac{1}{1 + 2jQ + j\omega_0/\omega_I}\right)^n,$$

where $n$ is the number of the stages of the complex filters. $\omega_0$ is the cut off frequency of the filter and 2Q is the relative center frequency of the complex filter. There are many techniques can be used for the complex filters. In the experiment, single complex pole with active-RC filter technique is used. Figure 10 [2] shows the circuit of the filter that used in the experiment.

Analog signal is converted to digital signal by using ADC with FPGA board. The FPGA is programmed by using VHDL language to controls the sampling of the ADCs. In the experiment, 4 channels of ADCs are used, 2 channels are used to sample I-phase and Q-phase signals from output of $H_0$ filter and the other 2 channels are used to sample output from $H_1$ filter. Both boards are set to start sample at the same time and sampled with same sampling clock to synchronize the boards. The sampling frequency is set to 10 MHz. The input level of the received signal is adjusted to the maximum amplitude of the output of filter each. The resolution of ADCs is adjusted by changing the least significant bit (LSB) and most significant bit (MSB) of the ADCs. The maximum system resolution is 12 [bits].

In the digital domain, the signal is processed with Wiener filter remove the remaining interference part. Finally the desired signal is demodulated and the performance in BER of the system is investigated.

5. Experiment Results

Firstly, the characteristics of the analog complex band pass filter are investigated by using spectrum analyzer. Then the result from the spectrum analyzer is drawn in Fig. 11 for $H_0$ and Fig. 12 for $H_1$. The characteristics of the analog filters are shown in Table 2. From the results, the center frequency of the $H_0$, $\omega_0$ is 1.64 [MHz] with bandwidth of 1.15 [MHz] and the center frequency of the $H_1$, $\omega_1$ is −1.64 [MHz] with bandwidth of 1.15 [MHz].

Spectrum analyzer cannot separate positive frequency and negative frequency, thus, to investigate positive and negative frequency, the output of the filter is analyzed with computer program. Figures 13–16 show computer analysis of input and output of analog filter for 64-carrier DQPSK/OFDM signal. Signal with 0 [dB] of SIR would have about the same power spectrum for both positive and negative frequency such as shown in Fig. 13. When the signal is filtered by the $H_0$ analog filter, only the positive part of the signal re-
main such as shown in Fig. 14. This means the filter only let the positive frequency pass through and select only the desired channel. However, when SIR is $-12 \,[\text{dB}]$ such as in Fig. 15, interference part would have much larger power than the desired signal. When the signal is pass through $H_0$, the negative frequency of the signal is still remaining in a small power as in Fig. 16.

When interference power is much larger than desired signal, negative frequency part would still remain in $Y_0$ although it had filtered by $H_0$ such as shown in Fig. 17(a). Meanwhile, only negative frequency part left in $Y_1$ which is output of $H_1$ such shown in Fig. 17(b). When the signal is processed with Wiener filter, the output of the filter (Fig. 17(d)) is seemed similar to reference signal (Fig. 17(c)) which only includes desired signal. In other words, the proposed scheme effectively cancels the interference. However when SIR is about $-24 \,[\text{dB}]$, the output of the Wiener filter is still not similar to the reference signal.

Figure 18 shows BER versus number of coefficient for Wiener filter with different SIR. The Wiener filter cannot approximate the characteristic of the interference if the number of coefficient is too small. The performance of the Wiener filter is converging when the number of coefficient is
Fig. 16  Output of $H_0$, SIR=$-12$ [dB] 64 carrier QPSK/OFDM signal.

Fig. 17  SIR=$-12$ [dB], SNR=$20$ [dB], Spectrum of (a) $Y_0$ (b) $Y_1$ (c) reference signal, $s$, (d) output of Wiener filter, $d$.

Fig. 18  BER vs. number of coefficient, SNR=$10$ [dB].

Fig. 19  BER vs. SIR, SNR=$10$ [dB], resolution [8 bits].

Fig. 20  BER vs. SNR, SIR=$-12$ [dB], resolution 8 [bits].

enough. However, if the number of coefficient is too large, much more noise will be included while approximates the interference. This will make the performance worse. In

Fig. 18, the performance seems to be converging when the number of coefficient is between 10 and 50. In this experiment, the number of coefficient is set to 40.

Figure 19 shows BER versus SIR with and without the adaptive digital signal processing in the proposed receiver architecture. In this figure, ‘no side lobe energy’ means that the signal on the adjacent channel does not have the side lobe energy and no interference occurs. When the SIR is more then $-18$ [dB] the Wiener filter seems effectively cancel the interference and improves the BER performance. However, when the SIR is $-24$ [dB], there is no much difference in performance. If the adjacent channel signal level is very high, the quantization error of the signal is also high. This will make the canceller hard to approximate the characteristics of the interference signal. As the result, the interference can not be eliminated completely and the performance degraded. From the figure, it can concluded that the effect of ACI can be reduced by about $18$ [dB] when the BER is $10^{-2}$.

Figure 20 shows BER versus SNR when SIR is fixed.
There is not much difference in BER performance when the SNR is varied for the signal without interference cancellation. The Wiener filter work better in low power of noise however it still can work effectively in high power of noise.

Figure 21 shows BER versus the resolution of ADCs when SIR is set to $-12 \,[\text{dB}]$. If the resolution of ADCs is 6–8 [bits], the Wiener filter effectively improve the BER performance. When higher resolution then 6 [bits] are used for SIR=$-12 \,[\text{dB}]$, the improvement is very small. However when the ADC is 4 [bits], there is not much difference in the BER performance between the filtered signal and not filtered signal. The reason is that the reconstructed signal through the adaptive signal processing includes the error due to the quantization noise. From the result, it is clear that 6 [bits] is enough for resolution when the SIR is higher then $-12 \,[\text{dB}]$ and SNR=$10 \,[\text{dB}]$.

6. Conclusion

In this paper, the ACI cancellation scheme with analog filter bank for multi-channel reception has been proposed. From experiment results, it has been shown that the proposed scheme can mitigate the influence from the adjacent channel and enables multi-channel reception with relatively low resolution of ADCs with the adaptive digital signal processing. Therefore, the propose scheme can be applied to VoIP services with WLANs.

From the results, the proposed scheme effectively cancels the interference and improved the BER performance when the resolution of ADCs is 6 [bits] and more, and the SIR is more then $-18 \,[\text{dB}]$. The effect of ACI can be reduced by about 18 [dB] for BER=$10^{-2}$.

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References


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