TECHNIQUE FOR ALLEVIATING DYANMICA-RANGE REQUIREMENTS FOR WIDEBAND SOFTWARE RADIO RECEIVER

Kaibin Huang (Institute for Communications Research (ICR), Singapore, Email: huangkb@icr.a-star.edu.sg), Quee Seng. Quek (MIT), Syed Naveen Altaf Ahmed (ICR), Bo Jin (ICR), Marath Ashok Kumar (ICR)

ABSTRACT

In this paper, a wideband digitization technique for digitizing narrow band signal in the presence of strong interference is proposed. The proposed technique can be implemented in two methods, the feed-forward and the feedback method. Analysis and simulation results are provided. Based on a WCDMA/GSM software radio receiver, it is shown that the proposed technique provides about 3-bit improvement on digitization resolution compared with the conventional approach.

1. INTRODUCTION

The rapidly growth of wireless communication and the emergence of all kinds of standards have generated interest in developing re-configurable multi-mode tranceivers. This leads to the creation of a new research area, called software radio [1][2]. In an ideal software-radio system, a tranceiver can intelligently recognize the characteristics of the received signals and re-configure itself in real time accordingly; programm itself in such a way that the channel resource is made use of in a optimized way; employ advanced techniques e.g. interference cancellation, equalization, error correction to achieve high data rate. In this initial stage, one important research area of software radio is the design of re-configurable multi-mode receiver. The realization of such a receiver is challenged in many practical aspects, including broad-band antenna, linear power amplifier, wideband frequency synthesis, and analog-to-digital (A/D) conversion. This paper is related to one of them, the A/D conversion.

A multi-mode software radio receiver is required to be wideband, as shown in Figure 1. A/D converter (ADC) remains as the main bottleneck in realizing such a wideband receiver. The stringent dynamic-range requirements for the ADC rise from the fact that the channelization function of the wide-band receiver is carried out digitally and a ADC is hence exposed to wideband radio signal, which may contain very strong interfering signals. This problem of digitizing wideband signal can't be solved using the conventional method in which a single ADC alone is employed to digitize the received wideband signal because the high-dynamic range requirements for such a scenario can't be satisfied even by the state of the art. This prompts the necessity for developing alternative techniques to alleviate the ADC dynamic-range requirements. The technique proposed in [3] though provides improvement in ADC resolution for wideband signal, does not ensure the signal in the desired channel is recovered with minimized quantization noise. A particular problem we address in this paper is that how to efficiently receive a very weak narrow band signal with high digitization resolution in the presence of strong interfering signals using a wideband multi-mode receiver.

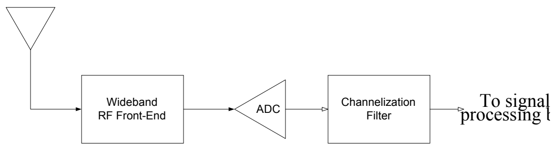


Figure 1 Wideband software radio receiver

2. SYSTEM MODEL

In this paper, a two-mode (WCDMA/GSM) [4][5] receiver is used as the platform for developing our wideband digitisation techniques and evaluate their performance. For such a receiver, the RF-filter is required to accommodate the signals of both standards, and hence is wideband (5MHz). The digitisation problem occurs only for GSM system as the wideband admits both the interference and the desired signal. And this is not the case for the WCDMA system. The power profile of the GSM interference at the output of the wideband receiver filter is specified in Table 1.

Table 1 Power profile of GSM interference

Frequency offset	Interfering signal level	Wanted signal level
200 KHz	-73dBm	-82dBm
400 KHz	-41dBm	-82dBm
600 KHz-1400 KHz	-43 dBm	-99dbm
1600KHz–2800 KHz	-33dBm	-99dBm
>= 3MHz	-23dBm	-99dBm

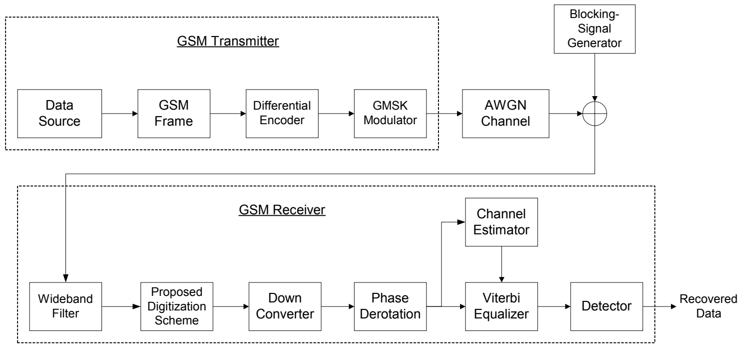


Figure 2 GSM system model with wideband receiver.

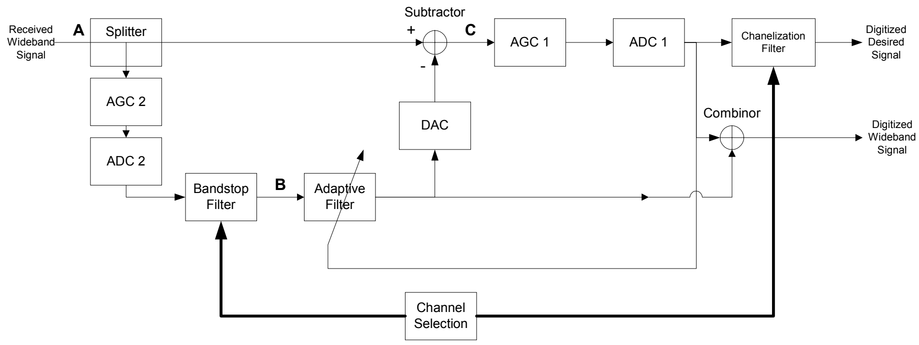


Figure 3 Block diagram of the proposed technique for alleviating ADC requirement.

For the reason described above, our analysis concentrates on digitisation of wideband GSM signal. The system model is shown in Figure 2. The designs of the transmitter and receiver are based on [7]. As specified in [6], the GSM signal is modulated using GMSK and the symbol rate is 270kHz. And the blocking signals (interference) are generated according to Table 1.

3. WIDEBAND DIGITIZATION TECHNIQUE

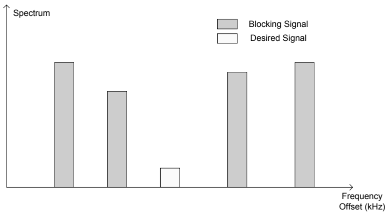
A technique for digitising with improved resolution a narrow-band, in this case GSM, signal at the presence of strong interference is proposed. Two possible implementations, the feed-forward and feedback methods, are discussed in the following sections.

3.1. Feedforward Method

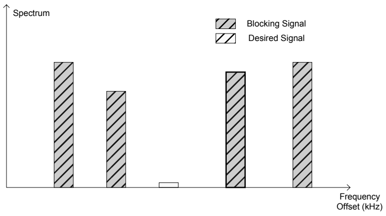
The feedforward method of our technique is shown in Figure 3. Its basic concepts is to suppress the interference using an adaptive filter after the first stage of coarse digitisation such that high resolution can be achieved in the second stage of digitisation. The details of its operation is described as follows.

The spectral content of the received signal, showing the signal power level, at different nodes of the system is shown in Figure 4. Assume that the received signal consists of multiple interfering signals at different carrier frequencies whose power is much larger than the desired signal, as illustrated in Figure 4(a).

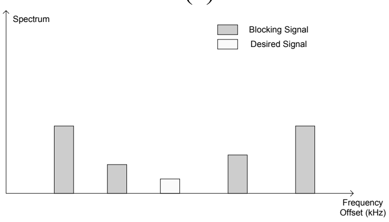
The received signal split by the splitter is firstly scaled by the AGC2 and digitised by ADC2 with relatively low resolution. The scaling function of AGC2 ensures that the input signal of ADC1 occupies the full input range of the ADC. The digitised signal is then passed through a band-stop filter that attenuates the desired signal and allows the interfering signal outside of the desired signal band to pass without attenuation. However, the filtering operation introduces phase-shift to the blocking signal as well as to the small residual desired signal, whose existence is due to the non-ideality of the band-stop filter. The spectrum of the filtered signal is illustrated in Figure 4(b), where slantwise parallel lines are used to distinguish phase-shifted signals from the original ones. In order to achieve substantial spectrum subtraction and hence interfering signal suppression, the phase-shifted blocking signals have to undergo phase-correction before they can be subtracted from the received signal. This task is carried out by the adaptive filter indicated in Figure 3. The operation of the adaptive filter shall be elaborated shortly and for this moment we just assume it carries out the task effectively. Therefore, the output of the adaptive filter consists mostly of the interfering signals that are almost in-phase with the original signals at node **A** of Figure 3. It is converted into analogue signal by a DAC. Having synchronous blocking signals at both inputs, the subtractor results in substantial suppression of the interfering signals. Because only very small residue of the desired signal lies at one of the subtractor input, the desired signal is not significantly affected by the subtraction. AGC1 serves the same purpose as AGC1. Since the strong interfering signals are suppress, the dynamic range requirements of ADC1 is consequently mitigated. Optionally, the outputs of ADC1 and adaptive filter can be summed to provide the digitised wideband signal for multiple channel signal processing.



(a)



(b)



(c)

Figure 4 Spectrums of the signal at node (a) A, (b) B, (c) C in Figure 3.

Analysis of the feed-forward method is carried out as follows. The architecture of the adaptive filter in Figure 3 can be a FIR filter, IIR filter, or Lattice filter. Very popular algorithms such as LMS and RLS [8] can be used to update the tap weights of the filter. To simplify the analysis, let's assume both the band-stop filter (BSF) and the adaptive filter are FIR filter so that linear transfer functions can be used and the channel is AWGN channel. The received signal to be digitised is given by:

$$r(k) = s(k) + \tilde{s}(k) + n(k) \qquad (1)$$

where $s(k)$, $\tilde{s}(k)$ and $n(k)$ are the desired signal, interfering signal and channel noise respectively. The output of ADC2 is:

$$r'(k) = A_2 \cdot r(k) + e_2(k) \qquad (2)$$

where A_2 is the gain provided by AGC2 and $e_2(k)$ is the quantization noise at ADC2. Denote the tap weights of the BSF as $\underline{h} = [h_0, h_1, \Lambda, h_M]^T$ where M is the number of tap. Then the BSF output is

$$r_b(k) = \underline{h}^T \underline{r'}(k) \qquad (3)$$

Because the desired signal $s(k)$ is removed by the BPF,

$$r_b(k) = \underline{h}^T [A_2 \underline{\tilde{s}}(k) + A_2 \underline{n}(k) + \underline{e}_2(k)] \qquad (4)$$

where

$$\underline{\tilde{s}}(k) = [\tilde{s}(k), \tilde{s}(k-1), \Lambda, \tilde{s}(k-M)]^T,$$

$$\underline{n}(k) = [n(k), n(k-1), \Lambda, n(k-M)]^T,$$

$$\underline{e}_2(k) = [e(k), e(k-1), \Lambda, e(k-M)]^T.$$

Define $\tilde{s}_h(k) = \underline{h}^T \underline{\tilde{s}}(k)$, $n_h(k) = \underline{h}^T \underline{n}(k)$ and $\underline{e}_2^h(k) = \underline{h}^T \underline{e}_2(k)$. Hence

$$r_b(k) = A_2 \tilde{s}_h(k) + A_2 n_h(k) + e_2^h(k) \qquad (5)$$

Using LMS algorithm, the weight updating function of the adaptive filter is:

$$\underline{w}(k+1) = \underline{w}(k) + \mu e(k) \underline{r}_b(k) \qquad (6)$$

where $\underline{r}_b(k) = [r_b(k), r_b(k-1), \Lambda, r_b(k-N)]^T$ and N is the number of taps of the adaptive filter. The error function $e(k)$ can be expressed as :

$$e(k) = A_1[r(k) - \underline{w}^T(k) \cdot \underline{r}_b(k)] + e_1(k) \qquad (7)$$

where $e_1(k)$ is the quantization noise at ADC1. Substitute (1) and (5) into (7)

$$\begin{aligned} e(k) = {} & A_1 s(k) + A_1[\tilde{s}(k) - A_2 \underline{w}^T(k) \cdot \underline{\tilde{s}}_h(k)] + \\ & A_1[n(k) - A_2 \underline{w}^T(k) \cdot \underline{n}_h(k)] - \\ & A_1 \underline{w}^T(k) \cdot \underline{e}_2^h(k) + e_1(k) \end{aligned} \qquad (8)$$

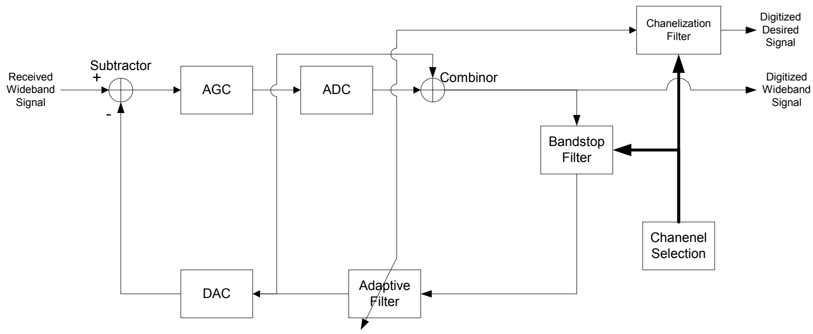


Figure 5 Feedback method of the proposed technique.

It can be observed from the above equation that by minimizing the error function, the adaptive filter in fact reduce the magnitude of the interfering signal $\tilde{s}(k)$ and the channel noise $n(k)$ leaving the desired signal $s(k)$ intact.

The function of AGC1 and AGC2 is to scale the input signals so that they are within input ranges of ADC1 and ADC2. Assume the input range of both ADC's are $[-1,1]$ and the power of the interfering signal is much larger then the desired signal. Then

$$A_1 = 1/\sqrt{E[|e_o(k)|^2]} \tag{9}$$

where $e_o(k)$ is the sample of the error function of the adaptive filter after it converges and

$$A_2 \approx 1/\left(\sqrt{E[|\tilde{s}(k)|^2]} + \sigma_n\right) \tag{10}$$

where σ_n is the normalized variance of the channel noise.

3.2. Feedback Method

The feedback method of the proposed technique is conceptually equivalent to the feedforward method and obtained by rearranging the data flows and component positions. The corresponding architecture is shown in Figure 5 and its operation is described as follows.

Like feedforward method, the feedback method is able to suppress the strong interfering signals but by using only one set of ADC and AGC instead of two by the former. The received signal is initially scaled by the AGC and digitised by ADC. This AGC ensures that the input signal of ADC occupies the full input range of the ADC. The digitised signal is then passed through a band-stop filter that only allows the interfering signals to pass without attenuation. Like feedforward form, the adaptive filter in the feedback form phase corrects the interfering signals from the band-stop filter. The phase-corrected interfering signals then converted into analogue signal by a DAC before it is subtracted from the incoming received signal. Such a feedback loop will suppress the strong interfering signals in the wideband-received signal before it is digitises. Hence, in order to retain the statistical information of the interfering signals required by the adaptive filter, the reconstructed interfering signals from the adaptive filter is added back to the suppressed digitised signal before it is passed through the bandstop filter. Since the suppressed digitised signal serves the *error function* [8] for tuning the taps weights of the adaptive filter. Depending on the required signal, this feedback form can generate either the digitised wideband signal or the digitised desired signal for further signal processing.

The channelization filter in both methods extracts the desired signal from the input for further signal processing. Both being digital, the channelization filter and the band-stop filter can be reconfigured by the channel selection block to receive different channels.

Comparing Figure 3 and Figure 5, it can be observed that the feedback method of the technique requires only one set of ADC and AGC instead of two by the former, hence the latter is more power saving and more suitable for mobile handset implementation. It is also shown in the next section that the two forms have comparable performance. It is, however, possible to remove the AGCs in both forms for simpler implementation as long as the signal input to the ADC is not too small to cause significant degradation in the digitizer.

In the cases that the power of the desired signal and interfering signal is comparable, it may not be necessary to employ the proposed technique and an ADC with ordinary resolution alone may be able to provide sufficient dynamic-range for the desired signal.

4. SIMULATION RESULTS

Simulations have been carried out to evaluate the performance of the proposed technique. The system model for the simulation is shown in Figure 2 and the power level of the interfering signals are defined in Table 1. Being both GSM channels, the desired signal and the interfering signal are both GMSK modulated and have identical bandwidth. Simulation models were developed on SPW© platform.

In the first simulated scenario, four blocking signals are present at frequency offset of –3.4MHz, 2MHz, 2.6MHz, 3MHz respectively with power level 66dB above that of the desired signal, and the channel is AWGN. In Figure 6, the performance of proposed technique is compared with the conventional method with different ADC resolution. Firstly, it can be observed that for the conventional system, gain in performance for 1-bit improvement in ADC resolution is smaller for higher ADC resolution. And the performance difference between the systems with 13-bit and 14-bit ADC's are insignificant. It can be also observed that the feed-forward method of the proposed technique performs almost equally well as the conventional system with 13-bit ADC. This suggests that the proposed technique that employs two ADC's of 10 and 8 bit each provides effective resolution of 13 bits in this setting.

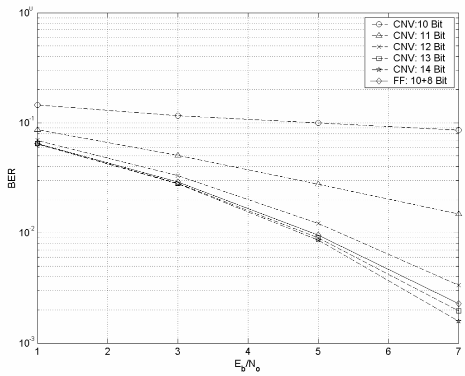


Figure 6 Performance of the proposed and conventional technique for AWGN channel with interference at frequency offset of –3.4MHz, 2MHz, 2.6MHz, 3MHz.

Next, the performance of the proposed and the conventional technique is simulated at the presence of one pair of interfering signals at different frequency offset, and the results are shown in Figure 7. It is observed that the feed-forward method is insensitive to the offset frequency of the interfering signals while the feedback method performs better for some values of interference offset frequencies than others. Generally, in this setting, the feed-forward method performs better than the conventional system with a 13-bit ADC. On the other hand, the performance of the feedback method is comparable to that of the feed-forward method.

The performance of the proposed technique for flat fading channel has been simulated. The simulation setting are the same as the first scenario described above where four interfering signals are present except that both the desired signal and the interference are each pass through a independent Rayleigh fading channel with Dopler frequency of 50Hz. From Figure 8 that shows the simulation results, it can be observed that the performance of both the feedforward and feedback method and the conventional method with 13-bit ADC are close to each other.

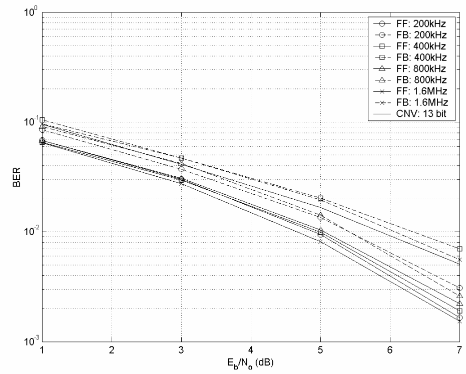


Figure 7 Performance of the proposed technique for AWGN channel with interference at different frequency offset.

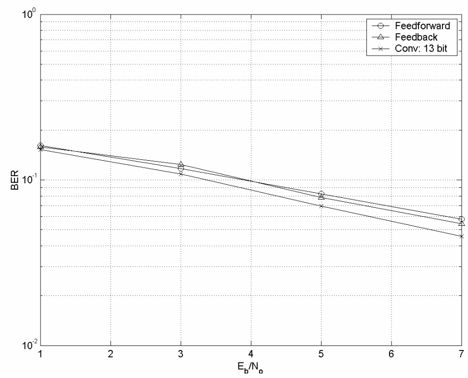


Figure 8 Performance of the proposed technique for flat fading channel

5. CONCLUSION

In this paper, a new digitisation technique for wideband software radio receiver is proposed. The proposed technique can be implemented in two methods: feed-forward and feedback. By cancelling the interference signal, the proposed technique provides improvement on digitisation resolution. In the simulated scenarios, by using two ADC's of 10-bit and 8-bit (feed-forward method), the feed-forward method of the proposed technique provides performance equal to or better than that of a 13-bit ADC in the conventional approach. Though the feedback method performs slightly worse than the feed-forward method in some cases, it promises more compact architecture.

REFERENCES:

[1] Mitola, J., III , "*Software radios: Survey, critical evaluation and future directions***",** IEEE Aerospace and Electronics Systems Magazine, Volume: 8 Issue: 4, April 1993

[2] P. B. Kenington, "Emerging technologies for software radio", *Electronics & Communication Engineering Journal*, April 1999.

[3] H. Nie, A. K. Salkintzis, and P. T. Mathiopoulos, "A new approach for mitigating A/D conversion requirements in software radio base stations", *Proceedings of IEEE Vehicular Technology Conference*, pp. 2810-2814, Sep 1999.

[4]T.OjanperaandR.Prasad, ``An overview of air interface multiple access orIMT-2000/UMTS", IEEECommun.Mag.,vol.36,pp.82–95,Sept.1998

[5]``Mobile station — base station compatability standard for dual-mode wideband spread spectrum cellular system," TIA/EIA, TIA/EIA/IS-95, 1993.

[6] *GSM 05.05 technical specifications: radio transmission and reception*, version 5.8.1, Oct. 1998.[7] Bjerke B.A., Proakis, J.G., Lee, K.Y.M., Zvonar Z.A, "Comparison of GSM receivers for fading multipath channels with adjacent and co-channel interference", *IEEE JSAC*, pp. 2211-2219, Vol. 18, No. 11, Nov 2000.

[8] Simon Haykin, *Adaptive Filter Theory*, Prentice Hall, 1986.