

Broadband DSP based feedforward amplifier lineariser

R.G. Randall, J.G. McRory and R.H. Johnston

The performance of the feedforward linearisation technique begins to degrade for signals with very wide bandwidths due to the non-ideal amplitude and phase frequency response of the components. By implementing part of the lineariser in a digital signal processor (DSP), a digital equaliser can be used to compensate for the non-ideal behaviour and improve the cancellation in the first loop.

Introduction: The feedforward lineariser was developed in the 1920s [1], and is commonly used for wireless base station amplifiers. It provides good distortion cancellation over a reasonably broad bandwidth because all of the components in the signal path are analogue and can be made broadband in nature. However, the feedforward lineariser is very sensitive and difficult to design. To achieve proper cancellation of the two signals in each loop, the amplitude, phase, and delay of the signals must be very closely matched [2]. These stringent requirements become even more of an issue as the bandwidth of the signal to be amplified increases. Even with careful design and choice of components, there will still be some non-ideal behaviour over frequency. When the responses of all the components in the signal path are cascaded, the amplitude, phase, or delay ripple across the signal bandwidth can become excessive and degrade the cancellation performance of the lineariser [3]. This is especially true when manufacturing variations and component tolerances are considered. A further disadvantage of the conventional feedforward lineariser is that the delay is usually realised as a fixed length of coaxial line and thus cannot compensate for any variation in delay due to aging or signal frequency.

DSP generated reference signal implementation: To compensate for the component frequency response and the non-adaptive nature of the delay line, a new lineariser has been developed that is a hybrid of the conventional feedforward lineariser and a digital signal processor (DSP). The signal cancellation loop of the hybrid lineariser is shown in Fig. 1. It can be seen that both the amplifier input signal and the reference signal are generated by the DSP. The reference signal is then used to cancel the linearly amplified component of the distorted amplifier output signal, leaving an error signal containing only the amplifier distortion. The remainder of the circuit is not shown, but is identical to the conventional feedforward, where the error signal is amplified and used to remove the distortion at the output.

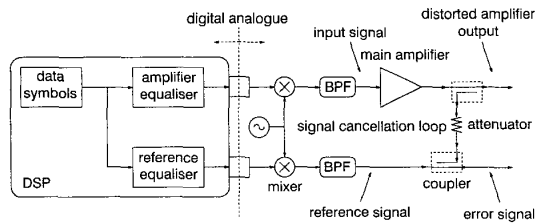


Fig. 1 Hybrid lineariser with DSP generated reference signal

By generating the reference signal in the DSP, rather than using an analogue splitter, some of the analogue hardware can be moved into a simpler digital implementation. This also allows the main amplifier input signal and the reference signal to be manipulated independently inside the DSP, so that an equaliser in the form of a digital filter can be applied to each signal. The equalisers perform the scaling, phase shift, and delay required to achieve the proper cancellation to generate the error signal. The equalisers also act to correct the non-ideal frequency response of the analogue components. The overall cascaded response of each path through the loop will thus have a flat amplitude response and zero phase. In theory this allows the two signals to cancel perfectly across the entire band corrected by the equalisers. This allows the cancellation to be maintained or improved over a broad bandwidth even with less ideal components. By improving the cancellation of the first loop, a more accurate error signal is generated that consists only of the distortion from the amplifier. This lowers the input power to the error

amplifier, which will generate less distortion and improve the overall linearity.

The ability to compensate for non-ideal components is also an advantage from a manufacturing standpoint. The feedforward lineariser has traditionally been a very difficult system to produce in quantity, as the extreme sensitivity required a great deal of tuning, including hand-building a custom delay line. The use of the equalisers in the first loop of the lineariser has an advantage in that the tuning previously done by hand has been moved back into the DSP where it can be done adaptively. Also, the DSP implementation of the equalisers allows them to be adapted to compensate for aging and changes in the signal characteristics, maintaining the cancellation performance of the first loop.

Experimental results: The measured signal cancellation achieved by the new lineariser for various bandwidths is shown in Table 1. The lineariser was built using commercial connectorised components, and was based around a Mini-Circuits ZHL-4240 amplifier. The centre frequency of the design was 1.8 GHz. We can see that excellent cancellation of 40 dB is achieved for a 125 kHz bandwidth signal, but the cancellation drops with increasing bandwidth. This is due to the implementation used for this prototype. An arbitrary waveform generator was used to generate the two signals rather than a true adaptive DSP implementation. This meant that the analogue component characteristics were measured, and the equalisers were designed and applied to the signals off-line. Since there was no way to adapt or fine tune the equalisers in this implementation, some error in the frequency response correction was unavoidable, which decreases the achievable cancellation. An additional phase shift and scaling was included in the experimental setup to tune the cancellation as best possible, which was effective for a narrow bandwidth. However, since the accuracy requirements to achieve high levels of cancellation are so tight, this was not sufficient to achieve good cancellation at broader bandwidths. When an adaptive DSP implementation is used, this will no longer be an issue, and the cancellation should improve to 40 dB or more regardless of the signal bandwidth.

Table 1: Hybrid lineariser signal cancellation loop performance

Approximate signal bandwidth [MHz]	Cancellation [dB]
0.125	40
48	25
190	21

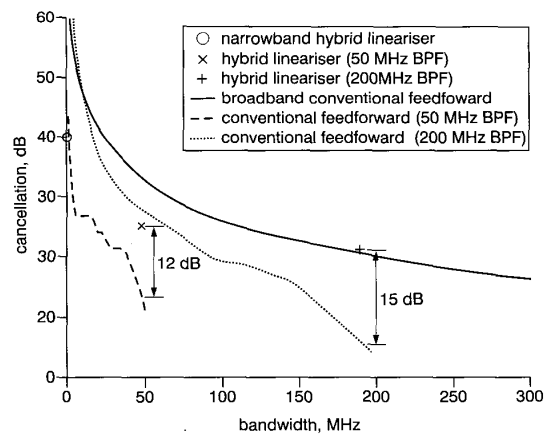


Fig. 2 Signal cancellation loop performance comparison

The performance of the first cancellation loop for a variety of lineariser configurations is shown in Fig. 2. The three measured results for the hybrid lineariser from Table 1 are shown in the Figure, as well as comparison curves from some other possible configurations. The first curve is for a very broadband implementation of the conventional feedforward lineariser. We can see that if our goal was 30 dB of cancellation, it could be achieved over a 63 MHz bandwidth. It is only at very large bandwidths that the non-adaptive hybrid lineariser

begins to outperform this conventional feedforward implementation. However the conventional feedforward lineariser shown here is not a practical system in that it was massively over-designed, used expensive, nearly ideal components, and required extensive tuning to achieve the performance shown.

It is interesting to see the effects of additional impairments on the conventional feedforward, which would be unavoidable in a more practical design. For a more realistic comparison, the same bandpass filters used in the hybrid lineariser were included in the conventional feedforward structure to determine the performance drop caused by the additional non-ideal frequency response of the filters. The second curve in Fig. 2 shows the maximum achievable cancellation when the 50 MHz bandpass filters (BPF) were added, and the third curve shows the impact of the 200 MHz filters. It can clearly be seen from the graph that even the addition of a small amplitude and phase impairment severely impacts the performance of the conventional feedforward. The bandwidth for 30 dB cancellation has dropped to 3 and 38 MHz, respectively, a severe decrease. However, with the same impairments, we can see that even without adaptation the hybrid lineariser achieves 12 dB better cancellation for a 48 MHz bandwidth signal, and 15 dB better for a 190 MHz bandwidth signal. Even at lower bandwidths, there is still a significant increase in performance by using the new lineariser. These results show the advantage of the hybrid lineariser. In a realistic design, where component and manufacturing variations exist, the ability of the equaliser to correct the frequency response dramatically increases the cancellation of the first loop for broader bandwidth signals. It is also important to note that with an adaptive implementation, the hybrid lineariser should achieve 40 dB cancellation over a large bandwidth range, and would outperform any implementation of the conventional feedforward lineariser.

Conclusions: The integration of the first loop of the feedforward lineariser into a DSP implementation has a number of advantages. It replaces some of the analogue components with a digital implementation, simplifying the design. This structure also allows equalisers to be used to correct for the non-ideal frequency response of the components. This relaxes the constraints on the components and manufacturing, while at the same time improving the cancellation performance of the first loop of the lineariser. When made adaptive, the equalisers also track any variations of the system, and adapt to maintain the cancellation performance.

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References

- 1 BLACK, H.S.: 'Translating system', U.S. Patent 1,686,792, October 9, 1928
- 2 SANG-GEE, KANG, and IL-KYOO, LEE: 'Cancellation performance of a linearisation loop of a feedforward amplifier', *Electron. Lett.*, 1997, **33**, (6), pp. 444-446
- 3 SMITH, A.M., and CAVERS, J.K.: 'A wideband architecture for adaptive feedforward linearization'. Proc. 48th IEEE Vehicular Technology Conf., 1998, Vol. 3, pp. 2488-2492

Combined transmit and receive diversity schemes for WCDMA in multipath fading channels

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The performance of various receiver antenna diversity combining schemes is analysed in combination with transmit diversity on the downlink of a WCDMA system. The analytical framework developed can handle an arbitrary number of transmit and receive antennas, as well as the study on the effect of dissimilar received signal strengths at the mobile handset on the average symbol error rate performance of a multitude of digital modulation schemes.

Introduction: In future third generation cellular systems, wireless multimedia and data services are expected to place higher demands on the downlink than on the uplink. It is important to find techniques that can improve the capacity of the downlink. Space-time processing technology and multiple input and multiple output (MIMO) architectures hold great promise to improve the capacity of the downlink. Transmit antenna diversity has already been accepted as part of the third generation partnership project [1]. The capacity of the downlink can be improved further by employing receiver antenna diversity in conjunction with transmit diversity, creating a MIMO architecture [2].

In [2], the bit error rate (BER) performance of a MIMO system with two transmit and receive antennas has been studied for BPSK modulation in Rayleigh fading. Further, it is assumed that the amplitudes of fading between each transmit and receive antenna are independently fading and that the average signal powers at each receiver antenna from each transmitter antenna are identical. In this Letter we extend these results by studying the average symbol error rate (ASER) performance of a MIMO system with an arbitrary number of transmit and receive antennas for M-ary PSK signals in a generalized fading environment (includes Rician, Nakagami-m, Nakagami-q and mixed fading channels). Our analytical framework allows us to examine the effect of non-identical average signal powers at each receiver antenna from each transmitter antenna as well as non-identical fading parameters on the ASER performance. This is particularly interesting because in actual mobile links, radio waves take different paths and may undergo different amounts of fading before arriving at the receiver.

Average symbol error rate analysis: The average symbol error rate (ASER) performance of the M-ary PSK modulation scheme is given by [3]

$$\bar{P}_s = \frac{1}{\pi} \int_0^{\pi-\pi/M} \phi_s \left(\frac{\sin^2(\pi/M)}{\sin^2(\theta)} \right) d\theta \quad (1)$$

where $\phi_s(s)$ is the moment generating function (MGF) of the signal to noise ratio (SNR) at the rake receiver output. Our immediate goal is to derive the MGF of SNR at the rake receiver output for various operational modes.

Open loop schemes: In an open loop system, space-time block coding is employed at the transmitter to improve the spatial diversity gain at the receiver. Since we have multiple receive antennas, we can employ either maximum ratio combining (MRC) or selection diversity combining (SDC) at the receiver. The former method is optimal in the maximum likelihood sense, whereas the latter is more cost effective.

Open loop MRC: In a MRC receive structure, the signals received from multiple receive antennas are co-phased and individually weighted to provide the optimal SNR at the output. For a system employing T transmitters and R receivers, SNR at the output of the rake receiver is given by

$$\gamma = \frac{1}{T} \sum_{j=1}^R \gamma_j = \frac{1}{T} \sum_{i=1}^T \sum_{j=1}^R \gamma_{i,j} = \frac{1}{T} \sum_{i=1}^T \sum_{j=1}^R \sum_{k=1}^L \gamma_{i,j,k} \quad (2)$$

where L is the length of the multipath intensity profile, $\gamma_{i,j,k}$ is the SNR of the k th multipath of the channel between i th transmitter and j th receiver. The MGF of γ can be easily shown to be

$$\phi_\gamma(s) = \sum_{i=1}^T \sum_{j=1}^R \sum_{k=1}^L \phi_{\gamma_{i,j,k}} \left(\frac{s}{T} \right) \quad (3)$$

where $\phi_{\gamma_{i,j,k}}(s)$ is the MGF of $\gamma_{i,j,k}$.

Open loop SDC: In a SDC receiver structure, the signal from the receiver antenna that provides the highest instantaneous SNR is demodulated. The SNR at the output of the rake receiver is given by

$$\gamma = \max \left\{ \frac{\gamma_1}{T}, \frac{\gamma_2}{T}, \dots, \frac{\gamma_R}{T} \right\} \quad (4)$$

If $F_j(x)$ denotes the cumulative distribution function (CDF) of (γ_j/T) , then the CDF of γ may be computed as