TRANSDUCER EQUALISATION : SIGNAL REPRESENTATION AND FILTER STRUCTURE

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1. Introduction

Equalisation of transducer frequency response is a recurrent requirement in high fidelity Sonar instrumentation, such as the echo repeaters used to provide synthetic targets. This paper considers available filter structures, applicable filter design techniques, and practical and performance issues for hard real time Sonar instrumentation operating at sampling rates of ~96 kHz.

When designing an equalisation filter the designer has freedom to select the signal representation (eg passband, complex baseband or 'single-sideband'), the filter structure (eg finite impulse response, infinite impulse response or wave digital filter) and the filter design technique (eg Remez exchange or least squares). This paper reviews the available techniques.

The 'optimum' design for an equalisation filter is strongly application dependent. It is influenced by the fractional bandwidth to be equalised, the fidelity required (eg linear phase, amplitude and group delay tolerance specifications) and by other application requirements and constraints (eg prescribed signal representation, maximum group delay). This paper considers the trade-offs afforded by the available techniques and tools.

Examples of equalisation filter implementations and their supporting infrastructure (particularly measurement and design tools) are presented to exemplify the design issues, and provide a selection of practical examples.

2. Equalisation Filter Requirements and Design Algorithms

Equalisation filter specifications can impose amplitude, phase, passband flatness and group delay requirements. The requirements for echo repeaters (figure 1) are typically limited to a detailed specification of the amplitude response (with a tolerance of +/- 0.5 dB), and either linear phase or minimum group delay (with a tolerance of a sample period). Most commonly, only the amplitude response is specified - over 10 to 50% of the passband, for implementation by a linear phase finite impulse response (FIR) filter.

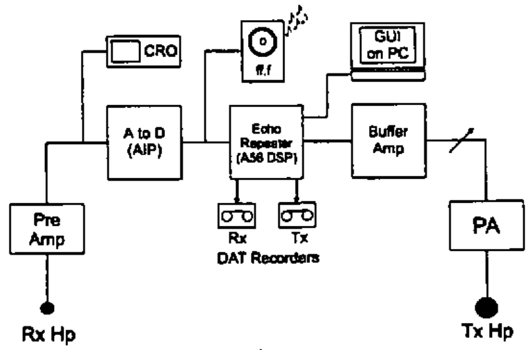

Figure 1 Echo Repeater System

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2.1 Signal Representation

If simultaneous equalisation of a significant fraction of the passband is required (the wideband case), the signal representation will generally be that of the analog to digital converter, typically real passband samples (figure 2). When simultaneous equalisation of only a small fraction of the passband is required (the narrowband case), the complex baseband and single-sideband (SSB) signal representations [1] are preferable (figures 3 and 4 respectively). Typically [2], a complex baseband representation is employed with multirate filter structures [1, 3, 4] and wave digital filters [5 - 12] used to mechanise the decimator and interpolator.

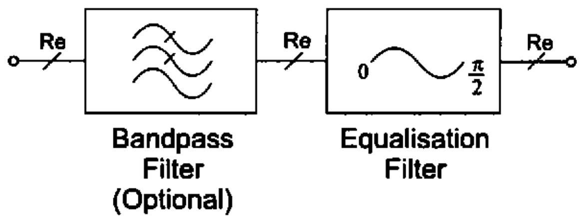

Figure 2 Passband Equalisation Filter

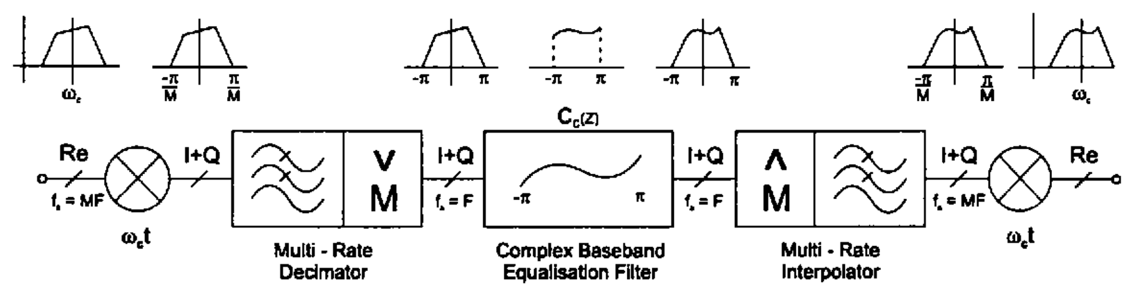

Figure 3 Complex Baseband Equalisation Filter

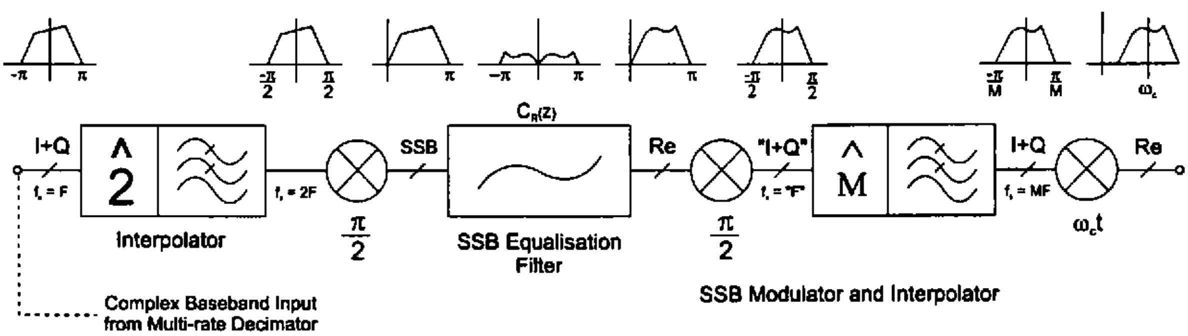

Figure 4 Single Side Band Equalisation Filter

2.2 Filter Structure

Typically, a real symmetric FIR filter is used for equalisation at passband. Equalisation of a passband or SSB signal can be performed by a real FIR, recursive or lattice filters, while complex baseband signals can be equalised by complex FIR or lattice filters. The key in each case is the availability of design tools.

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2.3 Filter Design Techniques
The techniques available for the design of recursive filters with arbitrary amplitude responses [13, 14] are limited, and the scope for second order problems a source of concern. By contrast, the literature is replete with design techniques for FIR filters.

The real FIR design technique in common use remains the Parks-McClellan implementation of the Remez exchange algorithm [13, 15, 16]; several modifications to its initialisation have been proposed to improve its convergence properties [17 - 19]. The eigenfilter method [20] can incorporate both frequency and time domain constraints. Steiglitz et al [21] have proposed a technique which permits the incorporation of more general kinds of constraints than Parks-McClellan, using the simplex algorithm. Iterative design techniques [22] using the LMS algorithm have also been proposed.

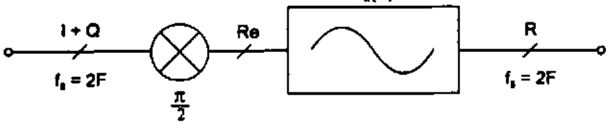
Algorithms are available for the design of complex FIRs using: the Chebyshev error [23], generalised Remez algorithms [24, 25], an extension to the eigenfilter method [26], and weighted least squares [27, 28]. Use of these design techniques enables the group delay to be minimised, typically at the expense of ripple in the group delay, potentially a worthwhile trade-off.

3. The Mechanisation of Equalisation Filters

For baseband implementations, either a complex filter can be used at baseband (figure 3), or a single-sideband filter can be used (figure 4), which operates at twice the baseband rate (which we refer to as twice-baseband rate) As the real FIRs designed by the Parks-McClellan algorithm are suitable for amplitude only equalisation requirements, we shall develop the SSB mechanisation and compare the use of real FIRs at passband and twice-baseband, with complex FIRs at baseband.

3.1 Direct Mechanisation of the SSB Equalisation Filter
Direct mechanisation of an equalisation filter using the SSB signal representation (figure 4), while conceptually straightforward is evidently inefficient. Only alternate samples from the first interpolator are output by the SSB mixer. Similarly, the input to the second, M-stage interpolator will have interleaved zeros in both the phase and quadrature limbs.

These inefficiencies can be eliminated, for example, by merging the SSB modulator with the FIR - note the use of interleaved phase and quadrature samples in figure 5, and by use of an SSB interpolator[1] (figure 6). However, these reductions in computation require increased control complexity; for example, the implementation of a half-sample advance in the SSB interpolator.



$$
\begin{aligned}
R_0 &= C_0I_0 - C_1Q_{-1} - C_2I_{-2} + C_3Q_{-3} \ldots \\
R_1 &= C_0Q_1 + C_1I_0 - C_2Q_{-1} - C_3I_{-2} \ldots \\
R_2 &= -C_0I_2 + C_1Q_1 + C_2I_0 - C_3Q_{-1} \ldots \\
R_3 &= -C_0Q_3 - C_1I_2 + C_2Q_1 + C_3I_0 \ldots
\end{aligned}
$$

Figure 5 Mechanisation of SSB Equalisation Filter

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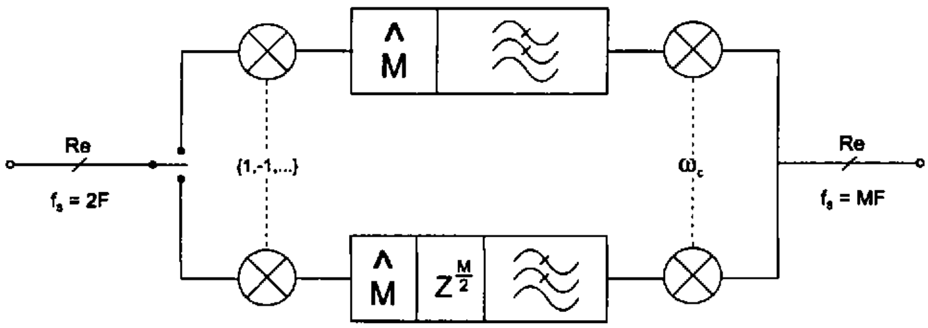


Figure 6 Mechanisation of SSB Modulator and Interpolator

3.2 Semi-Complex Mechanisation of the equalisation filter

An alternative mechanisation [29] of the twice-baseband equalisation filter (figure 7), changes the equalisation filter (figure 8), to match the complex signal representation. The equivalence between this and the SSB mechanisation can be readily confirmed by algebraic manipulation.

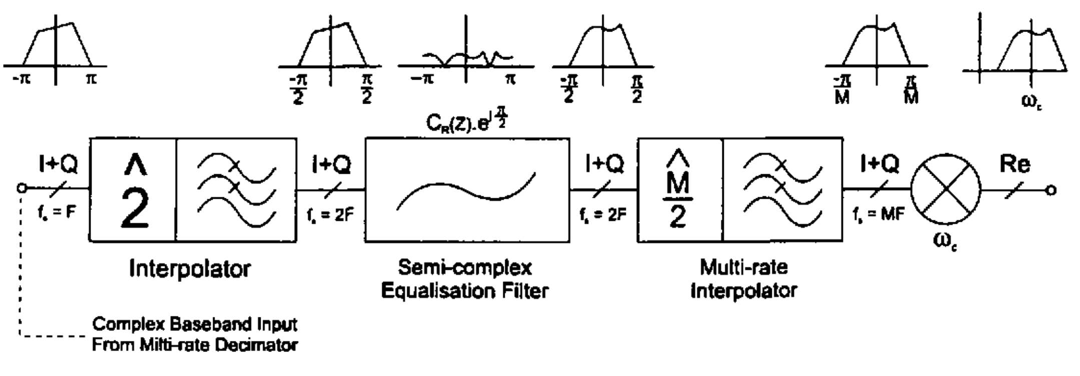
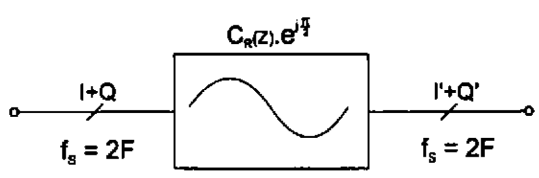


Figure 7 Semi-complex Equalisation Filter



$$I' = C_0 I_0 - C_1 Q_{-1} - C_2 I_{-2} + C_3 Q_{-3} + \dots$$
$$Q' = C_0 Q_1 + C_1 I_{-0} - C_2 Q_{-1} - C_3 I_{-2} + \dots$$

Figure 8 Mechanisation of Semi-Complex Equalisation Filter

The interleaved use of phase and quadrature samples in the twice-basband equalisation filter (figure 8), suggested its description as semi-complex, and as the arithmetic required is not complex, the complexity of the equalisation filter is no greater than that of two real FIRs. The equalisation filter, of course, remains essentially a real coefficient linear-phase FIR designed for a single-sideband application..

4. Design Considerations for Transducer Equalisation Filters

The pragmatic selection of the equalisation filter design algorithm and its mechanisation is now addressed. Four filter mechanisations are available to the designer: passband, single-sideband, semi-complex and complex.

4.1 Filter Specification

If the filter specification is restricted to amplitude, a Parks-McClellan real FIR design can be used to implement a passband (in the wideband case) or SSB/semi-complex (in the narrowband case) filter. However, where a linear phase FIR is unacceptable, for example to minimise group delay, a complex FIR is required.

4.2 System Integration

The computational costs for passband, SSB, semi-complex and complex mechanisations are tabulated in table 1. It is assumed that if an N tap FIR is used for equalisation at passband, then half that number of taps will be used for equalisation at baseband. While this assumption may seem unduly conservative, because available computational effort typically constrains the length of passband equalisation filters, it is considered realistic, as at passband performance must generally be compromised.

	Rate	Re/SSB/I+Q (MACS)		Taps	MACS	Rel. Cost	Notes
Passband	M	Re	(*1)	N Re	N.M	1	
SSB	2	SSB	(*1)	N/2 Re	N	1/M	Requires interpolation by 2*M
Semi-complex	2	I+Q	(*2)	N/2 Re	2N	2/M	
Complex Baseband	1	I+Q	(*4)	N/2 Cx	2N	2/M	Could Reduce Taps to N/4

Table 1 Equalisation Filter Mechanisation : Computational Cost Comparison

The reduction in computation obtained by using an equalisation filter at baseband is 2/M. As M (the decimation factor) is approximately inversely proportional to the fraction of bandwidth to be equalised, significant savings can be achieved in narrowband applications. Note that the SSB equalisation filter requires an additional interpolation stage which will typically outweigh the reduced compuational cost of mechanising only one limb of the equalisation filter.

Interestingly, for constant filter order, the computational cost of a twice-baseband semi-complex equalisation filter is equivalent to that of a baseband complex equalisation filter. The improvement in filter characteristics available from the additional degrees of freedom available in the complex filter is an interesting topic.

5. Design, Implementation and Evaluation of Transducer Equalisation

The Parks-McClellan FIR design algorithm and analysis software for both wideband and narrowband compensation filters are both implemented by PC software. Filter order estimation is performed empirically, although as noted earlier it is subject to such strong computational constraints that there is often negligible latitude in its selection. The measurement of unequalised systems, bench testing of equalisation filters and verification of equalised systems are also performed in a single integrated design environment.

5.1 In-Water Measurement

In-water echo repeater calibration (figure 9), uses several DSPs and a sampling digital voltmeter (SDVM) to measure transmitter sensitivity (S), receiver sensitivity (M) and system gain. The calibration is unusual in that the sensitivities are referenced to full scale deflection in the DSPs, dB re DSP FSD (cryptically shortened to dBF).

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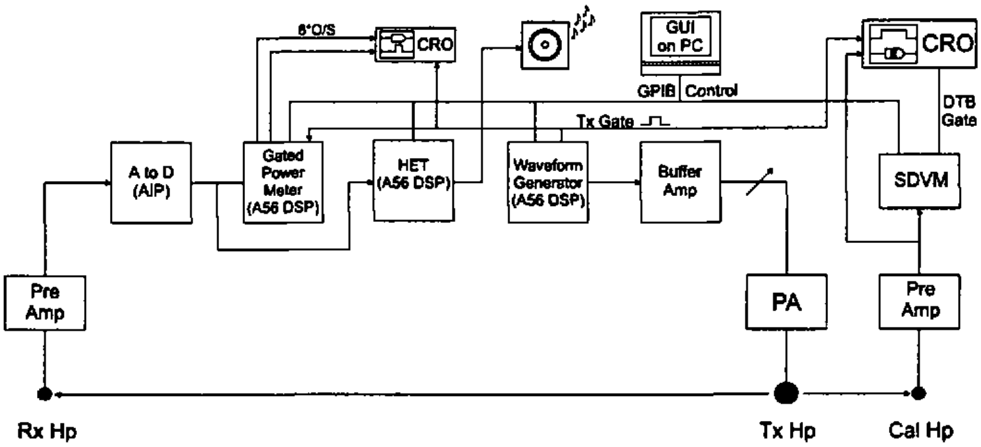

Figure 9 Echo Repeater Calibration

All calibration pulses are generated by the DSP components used in the deployed system, thus calibrating the output DAC and its analog reconstruction filter. Typically the A56 DSP [30] which drives the PA acts (inter alia) as waveform generator (figure 10), another A56 DSP measures the (gated) digital signals output by the receiver, thus calibrating the receiver's analog components and ADC, while a third A56 DSP provides a heterodyne for aural monitoring.

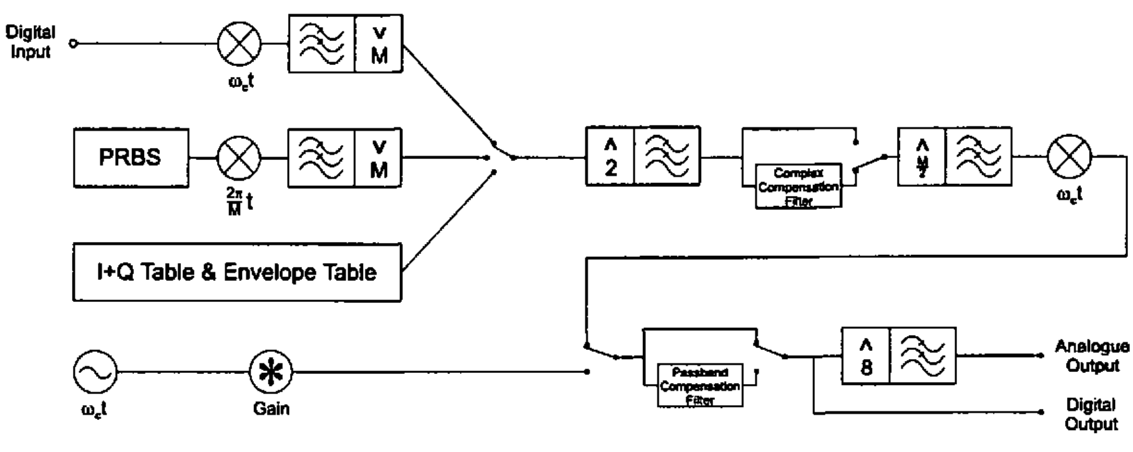

Figure 10 Echo Repeater Mechanisation

The calibration procedure is controlled by a PC which controls the generator, heterodyne and measurement DSPs, and the SDVM over GPIB. Positioning of the receiver sampling gate is under PC control, with the calibration hydrophone sampling gate generated by an oscilloscope.

5.2 The Integrated Design Environment

All filter analysis and evaluation is performed in an integrated design environment, which runs under Windows and is implemented in Delphi [31]. The control pages of the IDE (for example, figure 11) provide facilities for frequency and linearity scans, filter analysis and validation, etc. The results of analysis and measurements are reported to child windows for review prior to archival (for example, figure 12).

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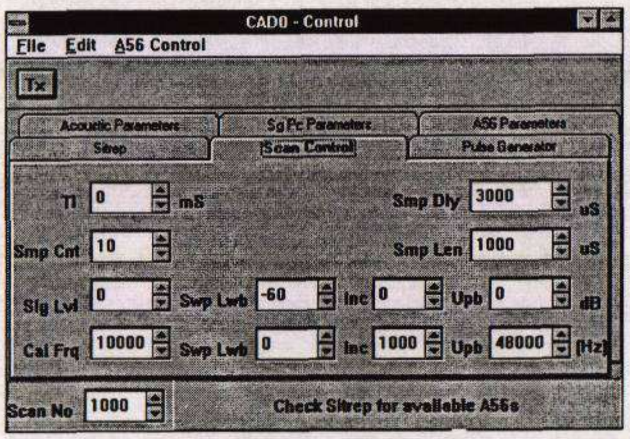

Figure 11 IDE Scan Control Page

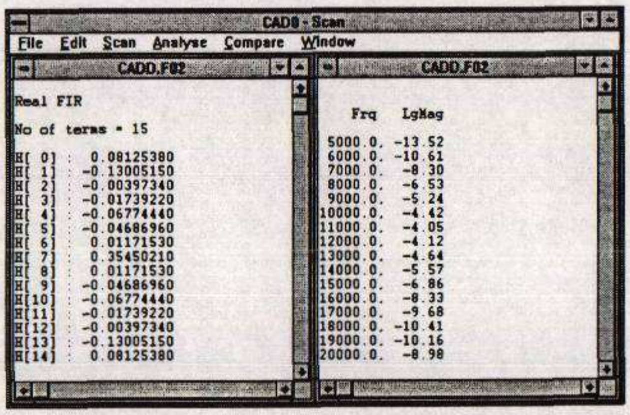

Figure 12 IDE Filter Coefficient and Analysis Windows

5.3 Equalisation filter design examples

Typically equalisation filters are required to provide 4..10 dB of spectrum shaping. Figure 13 shows a typical (wideband) requirement, and the gains of 31, 15 and 7 tap FIRs designed using Parks-McClellan to provide equalisation at passband. The gain errors (figure 14) show that a 31 tap FIR is required for a gain error of ~0.1 dB. Figure 15 shows the gains of 15, 7 and 5 tap FIRs designed using Parks-McClellan to meet the most difficult of the narrowband requirements by mechanising a semi-complex equalisation filter at the twice-baseband rate. The gain errors (figure 16) show that a 7 tap FIR is adequate for a gain error of ~0.1 dB.

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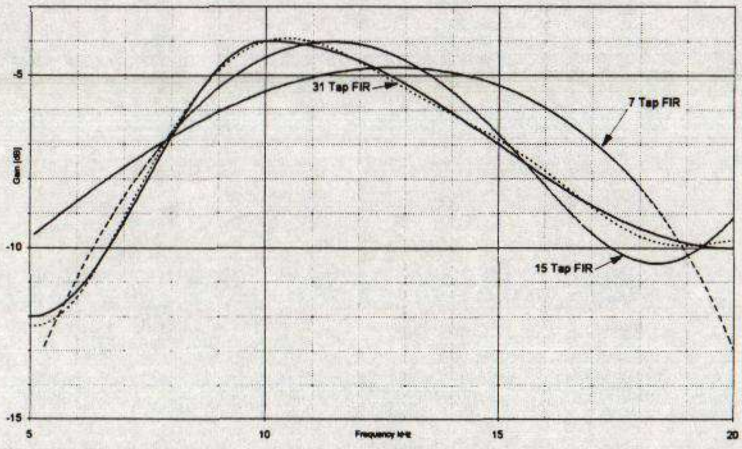

Figure 13 Wideband Equalisation Filter Response (31/15/7 Taps)

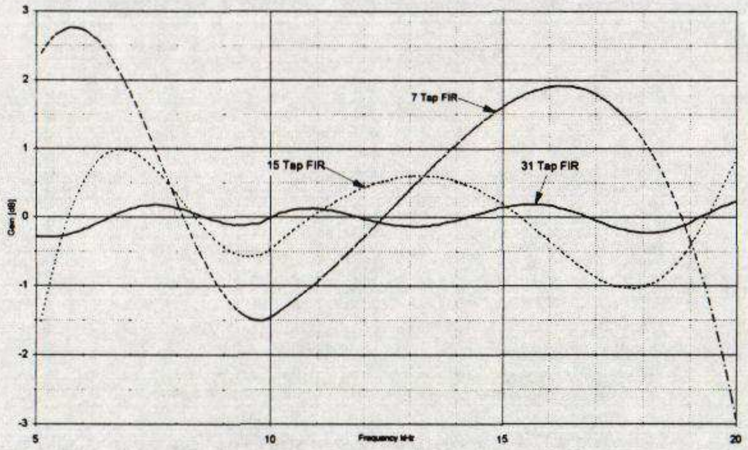

Figure 14 Wideband Equalisation Filter Error (31/15/7 Taps)

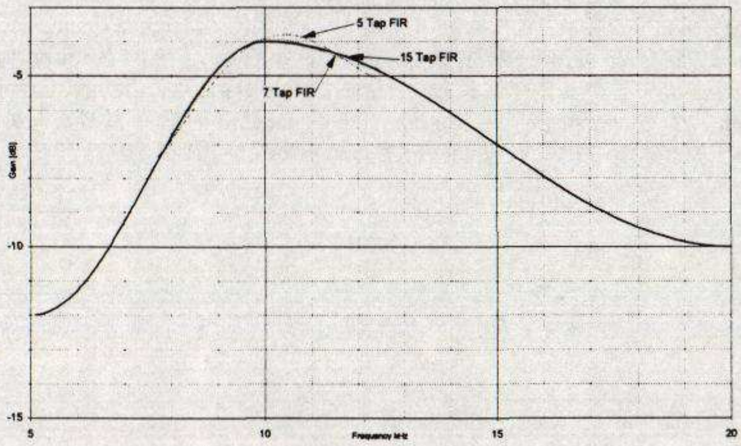

Figure 15 Narrowband Equalisation Filter Response

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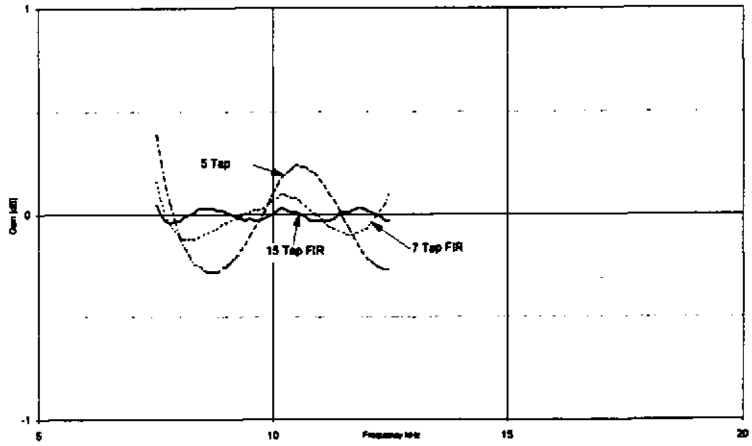

Figure 16 Narrowband Equalisation Filter Error (15/7/5 Taps)

6. Conclusions

Wideband equalisation of transducer responses can be performed using FIR filters designed by the Parks-McClellan algorithm at passband. However, this approach is computationally costly and the Remez exchange algorithm does not always converge.

Real FIR filters designed by the Parks-McClellan algorithm can efficiently perform narrowband equalisation of transducer responses at twice-baseband rate, providing computational savings proportional to the narrowband - wideband ratio, and an easier FIR design problem.

The utility of complex FIR filters for transducer equalisation remains to be evaluated. Also, considerable work is necessary before best practice algorithms for real FIR design are readily available to the systems engineer.

7. References

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