Switching between Two Different Fractionally Spaced Subband Equalisation Structures

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Abstract. This paper discusses methods for the fast adaptation of fractionally spaced equalisers. Subband methods offer advantages in adapting large scale equaliser, whereby two structures have been previously proposed. A first structure is faster, but can only be operated in training mode. A second structure permits the use in both training and decision directed mode at the cost of slower convergence. We propose and demonstrate a switching method to exploit the various benefits as best as possible.

1. INTRODUCTION

Fractionally spaced equalisers have recently been shown to exhibit slow convergence in the feed-forward (FF) part when applying the popular and low-cost least means squares (LMS) family of adaptive algorithms [1]. The FF part of the equaliser, when implemented in polyphase form as shown for the $T/2$ spaced equaliser realisation shown in Fig. 1, forms a multichannel adaptive filter. The slow convergence can be assigned to the correlation of the inputs between both polyphase components a_0 and a_1. To overcome this problem, subband equalisation techniques have been investigated, whereby the input to the feed-forward part is decorrelated. This is accomplished by splitting fullband signals into several decimated frequency bands ("subbands") [2, 3], in which independent equalisers can be operated.

In [3], two subband structures for fractionally spaced equalisers are discussed, which exhibit different characteristics. A first structure (structure I) can be operated in both training and decision directed mode, while a second structure (structure II), showing faster convergence, but can only be updated during training. Therefore, this paper focuses on a switching method, to run structure II during the initial fast training and thereafter switch to structure I in decision directed mode. A reverse operation can re-initialise structure II in training mode with the adapted result of structure I operated in decision directed mode.

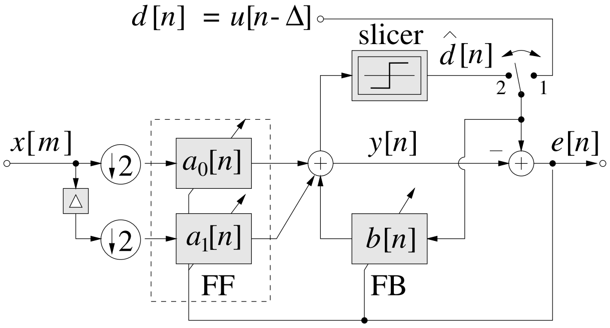


Fig. 1: Fractionally spaced equaliser with feed-forward (FF) and feedback (FB) part.

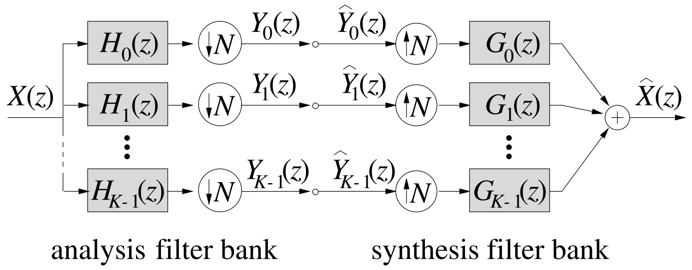


Fig. 2: K-channel filter bank decimated by N with analysis filters $H_k(z)$ and synthesis filters $G_k(z)$.

Subband structures I and II are reviewed in Sec. 2. Sec. 3 introduces the steps that allow us to switch from structure II in training mode to structure I in decision directed mode, and vice versa. Simulations and results are presented in Sec. 4, and conclusions are drawn in Sec. 5.

2. SUBBAND EQUALISER STRUCTURES

Equalisation in subbands projects the fullband equalisation problem into decimated frequency bands (socalled subbands), where smaller scale independent equalisation problems can be efficiently solved. First, the subband decomposition performing this projection is introduced. This will be used to operate various parts of the fractionally spaced equaliser in Fig. 1 in subbands as discussed in [3].

2.1 Subband Decomposition

Oversampled analysis filter banks as given in Fig. 2 with K channels and decimation by N are employed for subband decomposition [4, 5]. As aliasing in the subbands limits the performance of any subsequent adaptive filtering, oversampling with $N < K$ is mandatory [6]. To reconstruct a fullband signal from the subband domain, a synthesis filter bank is used, consisting of an expansion by N, interpolation filtering and summation over the K channels.

2.2 Subband Equaliser Structure I

The polyphase components of the FF part in Fig. 1, $a_0[n]$ and $a_1[n]$, are projected into subbands. The resulting architecture is shown in Fig. 3, whereby $\mathbf{H}$ and $\mathbf{G}$ denote analysis and synthesis filter bank blocks including decimation and expansion as given in Fig. 2, respectively. The matrices $\mathbf{A}_0$ and $\mathbf{A}_1$ are of diagonal polynomial form representing independent filters within each of the K subbands. As the FB has to be performed at symbol rate, the error is evaluated based on the FF outputs reconstructed by $\mathbf{G}$, and is projected back into the subband domain to update the filters in $\mathbf{A}_0$ and $\mathbf{A}_1$. The synthesis / analysis bank system in the error path causes a delayed update of the FF part [7], which is prone to slow convergence [8].

2.3 Subband Equaliser Structure II

If both FF and FB parts are implemented in the subband domain, the subband errors can be used to delaylessly update all filter components, as shown in Fig. 4. Similarly to the earlier notation, $\mathbf{B}$ is of diagonal polynomial form holding the adaptive FB filters running independently within each subband. Since the non-linear slicer cannot be transfered into the subband

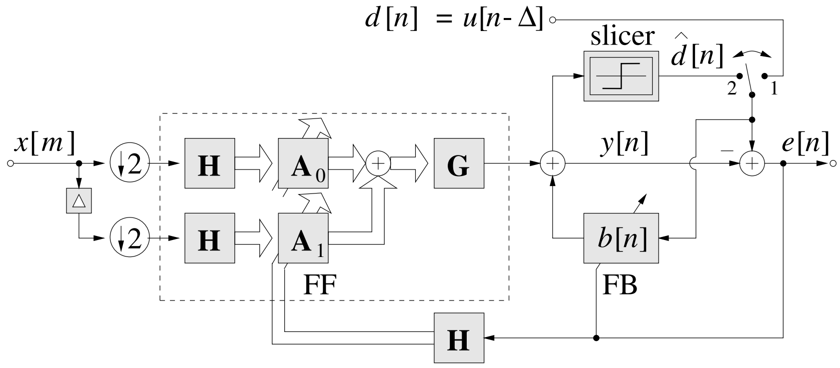


Fig. 3: Subband adaptive equaliser structure I. Depending on the switch position, the equaliser operates in training or decision directed mode.

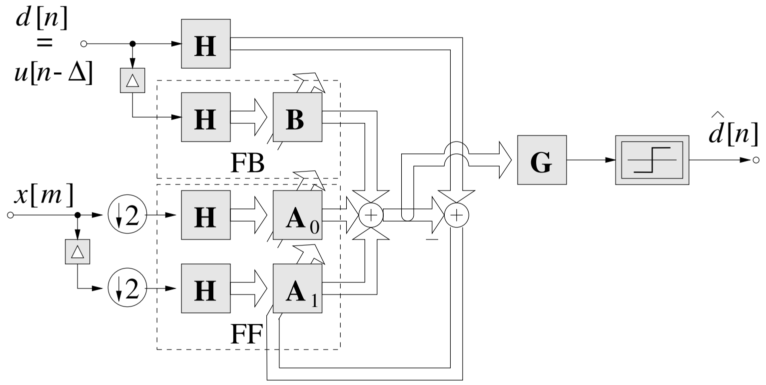


Fig. 4: Subband adaptive equaliser structure II with additionally the FB part in subbands.

domain, structure II can only performed in training mode. In training mode however, it has a faster convergence than structure I [3].

3. SWITCHING METHODS

In order to operate the faster structure II in training, but be able to update and track any channel changes in normal transmission mode, switching between structures I and II must be accomplished. In the following, we first comment on the transition from training with structure II to structure I in decision directed mode, followed by the reversed operation.

3.1 Switching from Training Mode to Decision Directed Mode

The proposed method to switch from structure II to structure I is based on the inclusion of a synthesis and analysis filter bank into the error path, as shown in Fig. 5. While the series of an analysis / synthesis system as shown in Fig. 2 only represents an overall delay z^{-L}, if the filter banks are perfectly reconstructing, a series of synthesis / analysis banks will only approximately delay the subband signals and introduce interference between at least neighbouring subbands. This has two implications on the updating in Fig. 5: (i) a delayed updating algorithm is required [8] to compensate for the approximate delay, and (ii) there is likely to be additional disturbance to the subband adaptive filters in $\mathbf{A}_0$ and $\mathbf{A}_1$ due to the subband interference.

It is however only the updating that is affected by stepping from Structure II to the one in Fig. 5. For the equaliser itself, it is now possible to pull the synthesis bank $\mathbf{G}$ across the summation point. In the FB branch of the equaliser, the equivalence of Fig. 6(a) and the projection of the subband FB equaliser in Fig. 6(b) permit to feed the new reconstructed FB equaliser $\tilde{b}[n]$ with a correctly sliced version of the equaliser output, $\hat{d}[n]$. The obtained circuit is now identical to structure I.

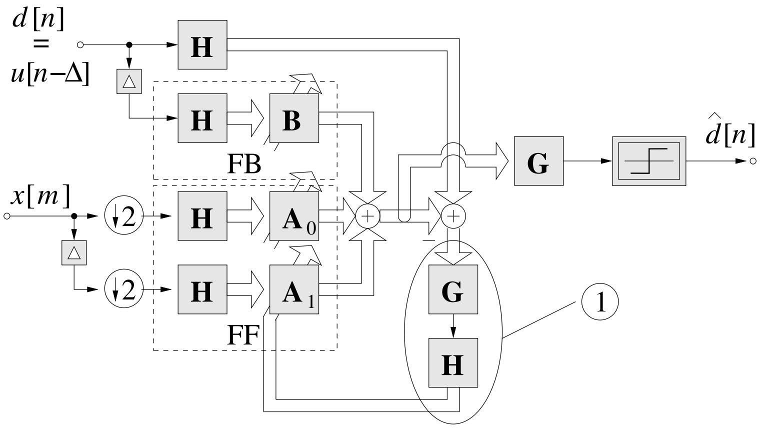
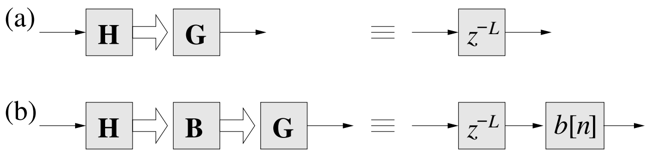


Fig. 5: Inclusion of a non-spurious synthesis / analysis system (1) into the error path of structure II.

Fig. 6: (a) Equivalence of the analysis / synthesis system to a delay, and (b) projection of the subband FB part into the fullband domain.



3.2 Switching from Decision Directed Mode to Training Mode

Operating structure I in decision directed mode is given by the architecture in Fig. 3 with the switch in position 2. When the equaliser returns to training mode, this switch is brought into position 1. From here onwards, the steps introduced in Sec. 3.1 are executed in reverse order.

We first move to the representation of structure I in Fig. 7, where the desired signal is defined with a delay L equivalent to the delay of the analysis-synthesis bank system in Fig. 6(a), according to which it can be substituted. The result of the substitution is the upper branch of Fig. 8, with a serial arrangement of the feedback response $b[n]$ and the analysis-synthesis filter system. The projection of $b[n]$ into subbands is accomplished by a subband system identification setup [5], whereby the subband errors in Fig. 8 can be directly used to independently update every subband filter. If the system identification is successful, the lower brach of Fig. 8 will by identical to its upper branch.

If the subband responses collected in the notation $\mathbf{B}$ are substituted into Fig. 8, the transition to structure II has been made. The remaining action is to draw the linear synthesis bank operation block $\mathbf{G}$ to the system output, and switch adaptation from the original delay-LMS to a standard LMS-type updating mode.

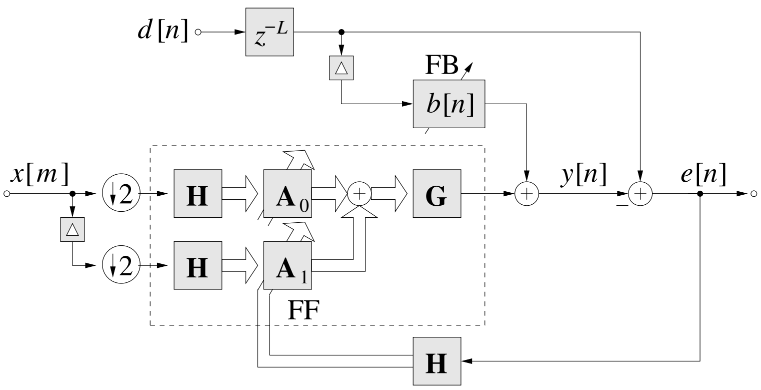
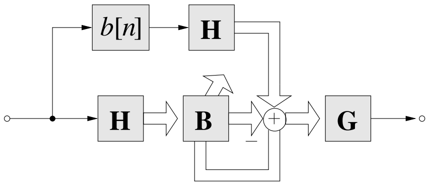


Fig. 7: Modification of structure I in training mode: a delay is extracted from the desired signal $d[n]$.

Fig. 8: Projection of $b[n]$ into subbands via subband adaptive system identification.



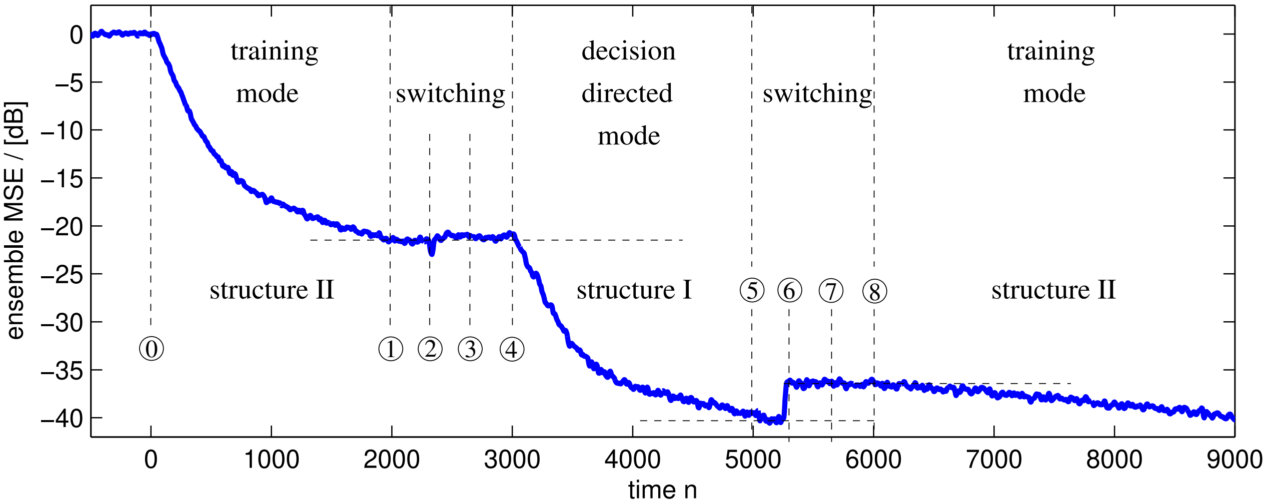


Fig. 9: MSE curve for switched subband equaliser: (0) structure II, adaptation switched on; (1) adaptation halted, projection of $b[n]$ for the FB part; (2) feeding $b[n]$ with $d[n]$; (3) feeding $b[n]$ with the slicer output $\hat{d}[n]$ (decision feedback); (4) structure I in decision directed mode (Fig. 3, switch position 2); reversed switching: (5) adaptation off, feedback filter fed by delayed training signal $d[n]$; (6) operation with projected subband equalisers; (7) synthesis bank transfered to output; (8) structure II adapted in training mode.

4. SIMULATION AND RESULTS

To demonstrate the switching method, a 16-QAM excited channel with 4 multipath components and 50 symbols delay spread is considered. The subband equalisers use $K = 4$ subbands decimated by a factor $N = 3$. The delay Δ of the desired signal $d[n]$ for training is set such that the FB part of structure II only operates on the post-cursor of the channel. To consider the effect of switching only, channel noise is neglected in the simulation.

The mean square error (MSE) curve of the switched equaliser structure is shown in Fig. 9, with various stages of the switching. The switching stages have to ensure that any switched systems are properly initialised. While in Fig. 9 no degradation is incurred by switching from structure II to structure I, the reversed operation introduces a decrease in MSE performance. This is due to the way how projections are performed in the two switching operations. In the transition II $\longrightarrow$ I, the length of $b[n]$ is set to be N times the subband equaliser length, while ideally the relation is given in [9] as

$$L_B \approx \frac{L_b + L}{N} \quad , \tag{1}$$

where L_B and L_b the lengths of the fullband and subband equaliser respectively, and L is the length of the filter bank's prototype filter. According to (1), the projection in switching II $\longrightarrow$ I is unlikely to truncate the projected result, while in switching I $\longrightarrow$ II a small error is likely to be made.

The projection error could be traded off between the projection lengths of the two switching operations. Additionally, the steps 1-4 and 5-9 shown in Fig. 8 can be grouped together and be performed at once.

5. CONCLUSION

A switching method has been introduced that permits to combine the benefits of two different subband equaliser structures. Structure II, which allows fast convergence due to delayless updating of all adaptive equaliser parts, is operated during training. Thereafter, with some approximations and a projection of the feedback part into the fullband domain, structure I can be performed in decision directed mode. In a similar projection operation, switching from decision directed mode back into training with structure II can be accomplished. This allows fractionally spaced subband equaliser to be applied to packed transmission scenarios such as in UMTS, where each packet contains a segment of training data.

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