Low complexity multichannel adaptive turbo equalizer for large delay spread sparse underwater acoustic channel

Wu Yanbo[a,b], Zhu Min[a,b]

[a]Ocean Acoustic Technology Laboratory, Institute of Acoustic,
Chinese Academy of Sciences, Beijing, China 100190
[b]State Key Laboratory of Acoustics, Institute of Acoustic,
Chinese Academy of Sciences, Beijing, China 100190

Abstract: A scheme of multichannel adaptive turbo equalizer for large delay spread sparse underwater acoustic channel is presented, with a complexity of single DSP chip implementation. Channel equalizer without input of decoder's soft information is structured as multichannel feed-forward (FF) filter with phase lock loop (PLL), and during iterative processing with decoder, an intersymbol interference canceller (IC) filter is added in the channel equalizer structure to further utilize soft information of the decoder. FF filter and IC filter are sparselized based on magnitude truncation of full order equalizer filters. Multichannel waveforms in training period are used to calculate filter taps with non-sparse least mean square(LMS) adaptive algorithm, with an order larger than 1000 (for example: 4 receiving channels, 2 samples per symbol each channel, FF filter duration is 160 symbol, then the multichannel FF filter order is 1280, and real-time DSP chip implementation is possible with LMS algorithm). After sparse tap position selection, the order of equalizer filter is reduced to less than 200. Multichannel receiving waveforms of both training and data symbols are processed with fast self-optimized LMS (FOLMS) adaptive sparse equalizer with PLL. Turbo code, bit-interleaving coded modulation (BICM), and QPSK/8PSK with gray mapping are adopted as they perform well in fading channel. The information exchanges between equalizer and decoder are realized by symbol/bit converters. The scheme is robust to combat delay spread with a duration larger than hundred symbols and easily implemented on a single DSP chirp. Extrinsic information transfer chart (EXIT) is used to analysis the performances of sparse equalizer, channel decoder and their iteration. Shallow lake and ocean experiments verified the performance of the scheme, with PSK symbol rate 4k Baud, half rate turbo code. Communication distance was from 1 km to 5 km, and there was no error frame with iteration of turbo equalization, even when delay spread larger than 40 symbols and no dominant channel path.

Keywords: **Underwater acoustic communication, sparse channel, large delay spread, turbo equalizer.**

Introduction

The characteristic of underwater acoustic communication channel is sparse, time-varying, and multipath, especially in long distance shallow water[1, 2]. Multipath is caused by the reflection of the surface or bottom, and curving sound ray. The variability of amplitude and phase of channel impulse response is mainly caused by movement of communication devices, when communication devices are lowed from floating ships, rather than moored from water bottom, the channel variability is especially obvious. The difficulty of shallow water communication is to confront the varying long multipath, and reverberation. Multichannel adaptive equalizer with phase tracking is a classic method[1], which can work well in sufficient signal-to-noise ratio (SNR). Turbo equalizer[3], using the turbo iteration principle, decreases the SNR requirement. Adaptive turbo equalizer was studied in [3], to tracking the variability of underwater channel[5]. Both adaptive equalizer and adaptive turbo equalizer face the problem of large number of equalizer taps, which decreases the tracking speed, increases the misadjustment noise and computation complexity. Sparse underwater channel estimation was studied in[4]. However, the optimum tap selection of equalizer based on sparse channel estimation is rather complicated, especially with multichannel and soft decision feedback[2]. This paper gives a low complexity method to realize adaptive sparse turbo equalizer. The sparse tap selection is simple but efficient. A turbo equalizer with 4 receiving channels is finally real-timely realized on a single DSP chip, which can adaptively deal with 1~5km distance with different multipath channel structures.

Proposed Sparse equalizer

To utilize the benefit of sparse characteristic, several considerations should be taken into account. Firstly, sparse position is constant during one frame. The frame duration used in the proposed scheme is less than 1 second, during which there is no new path emerging. Secondly, each path is a cluster, not a single ray. For the channel impulse response, the sparse position is not isolated but continuous with several symbols. Thirdly, sparse position need to be jointly optimized among different receiving array elements. With these considerations, the proposed method of sparse position initialization contains the following steps as shown in figure 1. Step 1, equalize adaptively during training period with full taps. The taps length is constant, according to maximum delay spread duration, and least mean square algorithm is used as it has advantages of robust and low complexity. Step 2, find positions of equalizer taps by magnitude truncation. A constant ratio to maximum amplitude of taps is used. Step 3, expand selected positions (to ±3 nearby positions). Figure 2, shows 4 channel equalizer taps and sparse position, with a distance of 1470 m. The full order number is 1288, which is reduced to 148 by sparse selection, as shown in red marks.

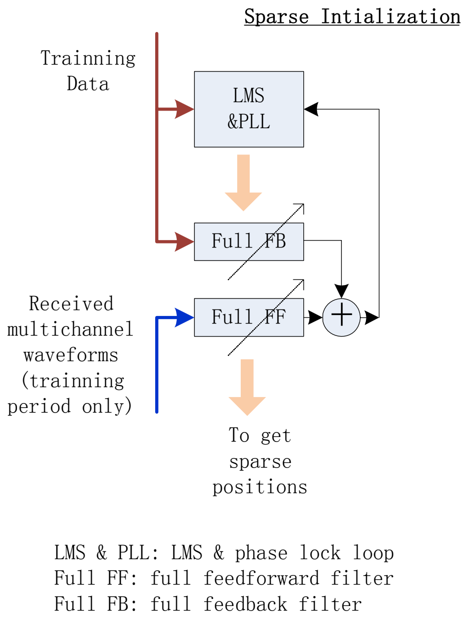

FIGURE 1. Proposed method of sparse position initialization

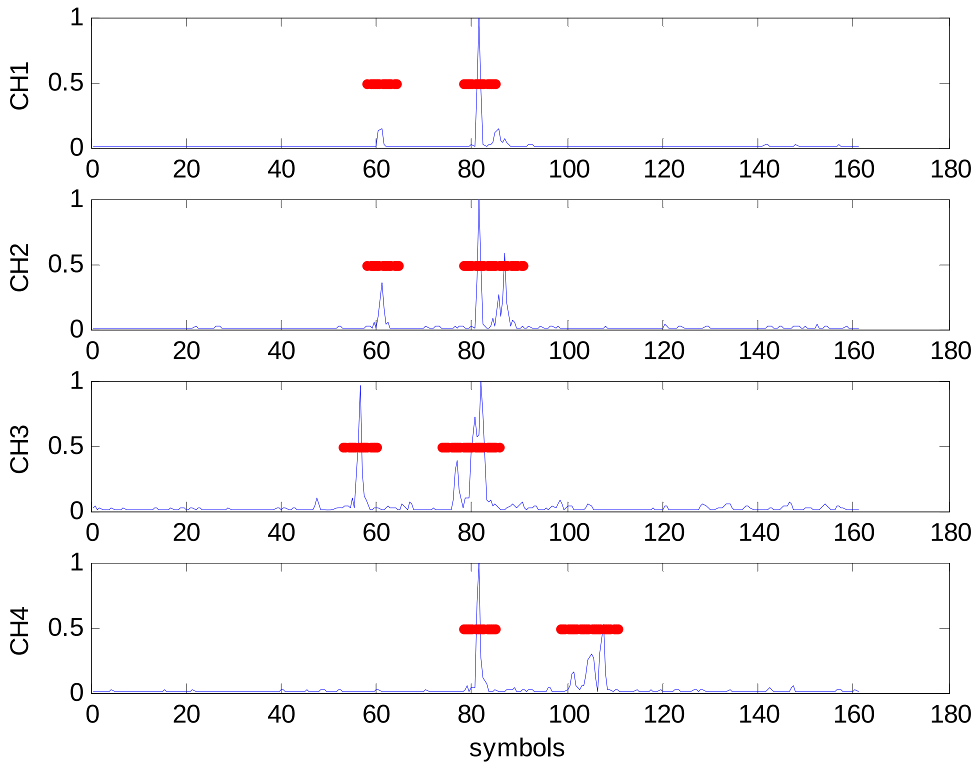

Figure 2 Four channel equalizer taps and sparse position

Adaptive Turbo Equalizer with Sparse Taps

Adaptive multichannel linear turbo equalizer is used in with the basic idea in [3]. With the first equalization, linear equalizer without feedback is adopted to avoid error propagation. During iteration of turbo equalization, soft output of decoder is fed into feedback equalizer filter with both post cursor and precursor. The proposed method here has following differences with others, as shown in Figure 3. The Equalizer is a sparse filter structure. The input data of equalizer is not circular after sparse selection, so reduced complexity RLS cannot be used. Meanwhile order of input data is also a variable depended on the channel spread length; the step factor must be adaptively selected. Fast optimized LMS is preferred, and normalized LMS may be suitable for real-time implementation. For the channel code, turbo code is used, as less iterations (3~5 is sufficient) of turbo equalization is need compared to RSC code (10~20). Another reason is that turbo code performs better in AWGN channel than RSC code.

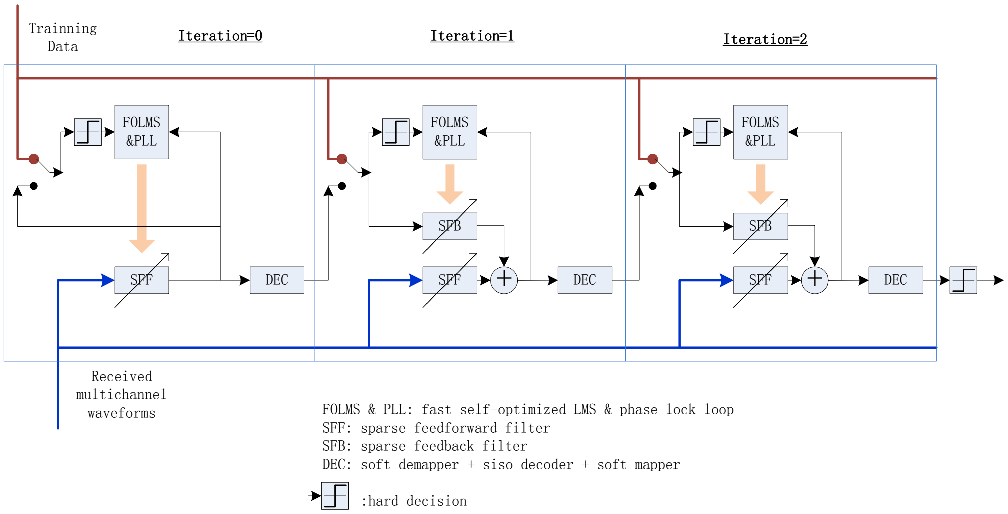

Figure 3 Structure of proposed Adaptive Turbo equalizer with sparse taps

Shallow Sea Experiment

Shallow sea communication experiment was carried out in South China Sea, in Aug/Sep 2013. The sound speed profile is shown in Figure 4. The water depth is 60 m, and one transducer and 4 receiving hydrophones were lowed from two ships with depth of near 30 m. QPSK was used with symbol rate of 4 kBaud, and carrier frequency was 8 kHz. ½ Turbo code was used, with a frame length of 1936 bit. Channel impulse response and extrinsic information transfer (EXIT) chart[6] of different distance are shown in figure 5~7. In the EXIT chart, the mutual information curves of equalizers (without / with feedback) and channel decoder are shown in red or blue lines. From the figures, we can see that the channel with moderate distance is favorable, and the SNR requirement is low, however the benefit of iteration is not so obvious.

1624 frames were collected, with varying distance from 1 km to 5 km. The proposed scheme (equalizer spans 160 symbols) was compared to traditional adaptive DFE scheme (equalizer spans 32 symbols) with turbo decoder [7]. The results are shown in Figure 8. The sparse equalizer make the FER decreasing from 19% to 7%, and after 5 iteration of turbo equalizer, the FER reduces to 0.

The real-time implementation of proposed scheme was developed on TigerSharc101, a float-point DSP, with CPU clock 300MHz. The processing abilities are listed as following: information bit rate 4kbps, full equalizer filter order 640(20ms

4ch), 4 iterations of equalizer and decoder per frame, twice iterations of turbo decoder after each equalization.

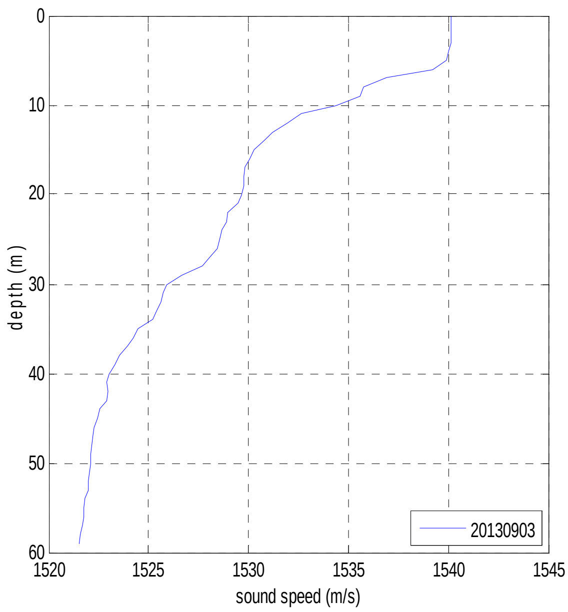

FIGURE 4 Sound speed profiles in South China Sea (3rd Sep, 2013)

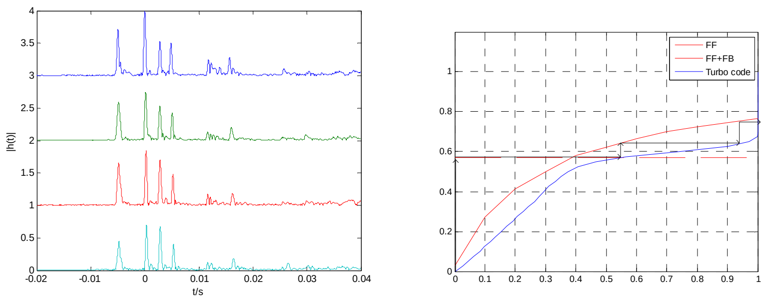

FIGURE 5 Channel impulse response and EXIT Chart (1)
(Distance=1159 m, SNR=1dB /channel)

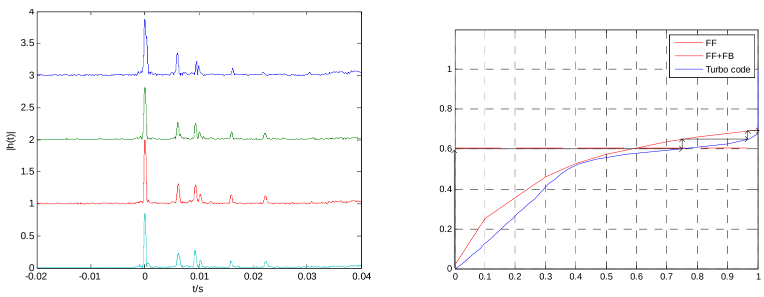

FIGURE 6 Channel impulse response and EXIT Chart (2)
(Distance=2421 m, SNR=-1.5dB /channel)

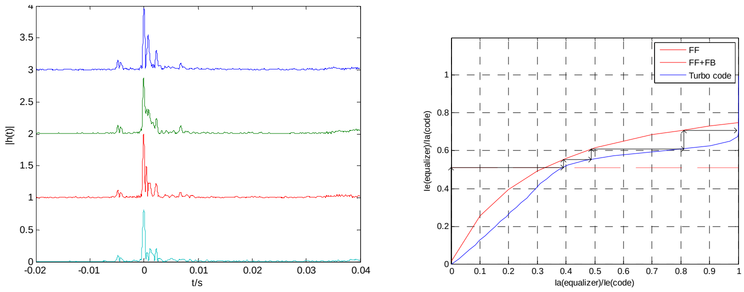

FIGURE 7 Channel impulse response and EXIT Chart (3)
(Distance= 4258 m, SNR= -0.5dB /channel)

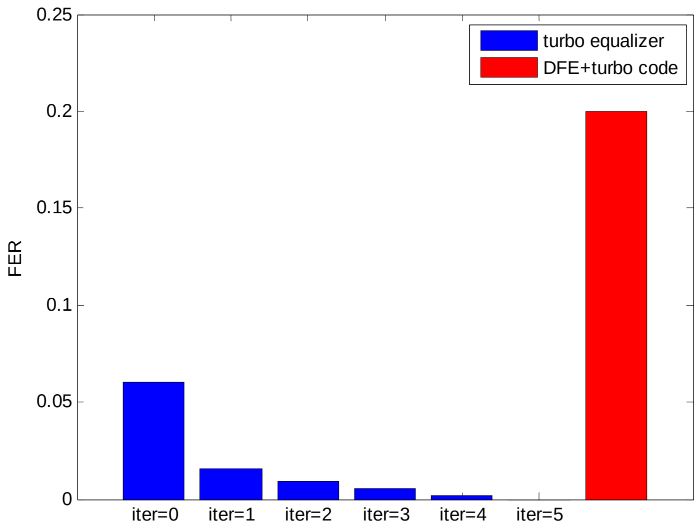

FIGURE 8 Frame error rate comparisons

ACKNOWLEDGMENTS

This work was supported by the Chinese National 863 Projects (2009AA093301, 2009AA093601).

REFERENCES

[1] M. Stojanovic, J. A. Catipovic, and J. G. Proakis, "Phase coherent digital communications for underwater acoustic channels," IEEE J. Ocean. Eng. 19, pp. 100-111, 1994.

[2] M. Kocic, D. Brady, M. Stojanovic, and S. O. C. Marine Technol, Sparse equalization for real-time digital underwater acoustic communications, 1995.

[3] C. Laot, A. Glavieux, and J. Labat, "Turbo equalization: Adaptive equalization and channel decoding jointly optimized," Ieee Journal on Selected Areas in Communications, vol. 19, pp. 1744-1752, Sep 2001.

[4] S. F. Cotter and B. D. Rao, "Sparse channel estimation via matching pursuit with application to equalization," Ieee Transactions on Communications, vol. 50, pp. 374-377, Mar 2002.

[5] J. W. Choi, T. J. Riedl, K. Kim, A. C. Singer, and J. C. Preisig, "Adaptive Linear Turbo Equalization Over Doubly Selective Channels," Oceanic Engineering, IEEE Journal of, vol. 36, pp. 473-489, 2011.

[6] S. J. Lee, A. C. Singer, N. R. Shanbhag, and Ieee, Analysis of linear turbo equalizer via EXIT chart. New York: Ieee, 2003.

[7] Zhu Weiqing, Zhu Min, Wu Yanbo and et al. Signal processing in underwater acoustic communication system for manned deep submersible "Jiaolong". Chinese Journal of Acoustics, 2013; 32(1)