A list Viterbi equalizer with simple liner channel interpolation for acoustic communications

Kazuki Shimomura¹, Syota Nagano¹, and Hiroshi Kubo²a)

¹ Graduate School of Science and Engineering, Ritsumeikan University,
1–1–1 Noji-Higashi, Kusatsu-shi, Shiga 525–8577, Japan
² Faculty of Science and Engineering, Ritsumeikan University,
1–1–1 Noji-Higashi, Kusatsu-shi, Shiga 525–8577, Japan

a) kubohiro@fc.ritsumei.ac.jp

Abstract: This paper discusses a list Viterbi equalizer (LVE) with liner channel interpolation (LCI), which can cope with time and frequency selective channels, i.e., doubly-selective channels of acoustic communications. Features of the proposed LVE are (1) to cope with delay spread over 127 bits with small computational complexity, (2) to cope with time selective channels by simple channel estimation and its LCI and (3) to cope with non-minimum phase condition by metric-criteria combining. Next, computer simulation results show that the proposed LVE has better performance on doubly-selective channels than a decision-feedback equalizer employing recursive least square (RLS) algorithm. Finally, field evaluation results confirm that the proposed LVE is suitable for acoustic communications environment.

Keywords: acoustic communications, doubly-selective channels, list Viterbi algorithm, linear channel interpolation, offset-quadrature phase-shift keying

Classification: Wireless Communication Technologies

References

[1] A. C. Singer, J. K. Nelson, and S. S. Kozat, “Signal processing for underwater acoustic communications,” IEEE Commun. Mag., vol. 47, no. 1, pp. 90–96, Jan. 2009. DOI:10.1109/MCOM.2009.4752683
[2] Z. Yu, Z. Kuang, M. Wu, and J. Yang, “Aerial acoustic communication for long-range applications based on single-carrier frequency domain equalization,” Proc. IEEE International Conference and China Forum on Signal and Information Processing, pp. 1007–1011, July 2015. DOI:10.1109/ChinaSIP.2015.7230556
[3] S. U. H. Qureshi, “Adaptive equalization,” Proc. IEEE, vol. 73, no. 9, pp. 1349–1387, Sep. 1985. DOI:10.1109/PROC.1985.13298
[4] H. Kubo, K. Murakami, and T. Fujino, “A list-output Viterbi equalizer with two kinds of metric criteria,” Proc. IEEE ICUPC’98, (Florence, Italy), Oct. 1998. DOI:10.1109/ICUPC.1998.733689
[5] H. Kubo and M. Miyake, “Single carrier modulation scheme employing a list
Viterbi equalizer with a metric-criteria combining scheme,” Proc. PIMRC’99, (Osaka, Japan), Sept. 1999.

1 Introduction

Recently, acoustic communications have been studied in land and underwater environment [1, 2]. A propagation environment of acoustic communications suffers from large delay spread and fast time-varying channels, i.e., severe doubly-selective channels.

In mobile communications, channel state information (CSI) varies frequently between minimum phase condition and non-minimum one. In order to maintain good communication quality, equalizers have to track and compensate these channel conditions. In these environments, a decision-feedback equalizer with a feedforward filter (FFF-DFE) employing recursive least square (RLS) algorithm has good performance [3]. However, computational complexity of RLS algorithm increases in proportion to the number of required taps. From an implementation point of view, it is important for equalizers to reduce computational complexity even on large time dispersive channels.

This paper proposes a list Viterbi equalizer (LVE), which employs simple channel estimation and its liner channel interpolation. The list Viterbi algorithm (LVA) is one of state-reduction algorithms including decision-feedback sequence estimation (DFSE) and M-algorithm, and it can reduce computational complexity, keeping good performance. In addition, the LVE with the metric-criteria combining (MCC) scheme [4, 5] has excellent performance in non-minimum phase condition environment. This paper evaluates performance of the proposed LVE on doubly-selective channels. Finally, field experimental results confirm that the proposed LVE is suitable for acoustic communications.

2 Communication system model

Fig. 1 shows a communication system model in the presence of inter-symbol interference (ISI), where $k$ denotes discrete time, $b_k (b_k \in \{+1, -1\})$ is a transmitted information sequence, $r_k$ is a received sequence and $\hat{b}_k$ is an estimated information sequence.
sequence. This paper employs offset-quadrature phase shift keying (OQPSK) in order to suppress influences of non-linear distortion of the amplifier and sampling timing error [5]. The information sequence \( b_k \) and the modulated signal \( x_k \) are described as follows:

\[
\begin{align*}
    u_k &= \exp(j\pi b_k), \\
    x_k &= u_k \exp\left(j\frac{\pi}{2}k\right).
\end{align*}
\]

(1)

(2)

Bit rate sampling of the OQPSK signals generates to \( \pi/2 \)-shift BPSK signals in the presence of ISI. Demodulators employing equalizers can compensate ISI caused by both OQPSK and channels. The received sequence \( r_k \) is as follows:

\[
r_k = \sum_{i=0}^{L} h_k[i] x_{k-i} + \eta_k,
\]

(3)

where \( L \) denotes a channel memory length, \( h_k[i] \) denotes channel impulse response (CIR) including ISI caused by OQPSK, channels and phase rotation of \( \pi/2 \) at the receiver side, and \( \eta_k \) denotes additive noise.

Maximum delay spread in acoustic communications is much larger than that of radio frequency (RF) band wireless communications. This paper assumes that the symbol rate is 8 ksps and the maximum delay spread is from several msec to several tens of msec. This paper selects \( L = 127 \) in order to cope with the delay spread of about 15 msec.

3 Functions of receiver part

A demodulation part employs a reverse phase rotator, a timing estimator, a channel estimator, LVE and forward error correction (FEC) processing.

3.1 Channel estimation based on synchronization word (SW)

Assuming 8th-order M-sequence with \( L \) bit cyclic prefix (CP), the estimated CIR \( \hat{h}[i] \) can be simply derived without performance degradation as follows [5]:

\[
\begin{align*}
    \hat{h}[i] &= \frac{1}{2^8} p_i + \frac{1}{2^{15}} \sum_{i=0}^{L} p_s, \\
    p_s &= \sum_{j=0}^{2^8-2} r_{i+s+x_i^M},
\end{align*}
\]

(4)

(5)

where \( x_i^M \) is the M-sequence of the transmitted sequence, and \( p_s \) is a cross-correlation value.

3.2 List Viterbi equalizer

The LVA is a generalized concept including DFSE and M-algorithm, where \( V \) is a memory length, \( 2^V \) is the number of states and \( S \) is the number of surviving paths. The LVE employs the metric-criteria combining scheme \( \Gamma_k^{MCC} \) which combine the squared Euclidean metric-criterion, \( \Gamma_k^{SQR} \), with the modified metric-criterion based on the matched filter, \( \Gamma_k^{MOD} \), as follows:

\[
\Gamma_k^{MCC} = \Gamma_k^{SQR} + \lambda \Gamma_k^{MOD}
\]

(6)
\[
\Gamma_k^{\text{SQR}} = |r_k - \hat{r}_k|^2
\]

\[
\hat{r}_k = \sum_{i=0}^V \hat{h}_i \tilde{u}_{k-i} + \sum_{i=1}^L \hat{h}_i \tilde{u}_{k-i}^{\text{SV}}
\]

\[
\Gamma_k^{\text{MOD}} = -2 \text{Re}[y_k \tilde{u}_{k-1}^*] + s_0 |\tilde{u}_k|^2 + 2 \text{Re} \left[ \sum_{j=1}^L s_j \tilde{u}_k^* \tilde{u}_{k-j} \right]
\]

\[
y_k = \sum_{i=0}^L \hat{h}_i^* r_{k+i}
\]

\[
s_j = \sum_{i=0}^{L-j} \hat{h}_i^* \tilde{h}_{i+j},
\]

where \(\tilde{u}_k\) is a candidate of the transmitted information sequence, \(\tilde{u}_k^{\text{SV}}\) is a candidate of the transmitted information sequence based on the surviving path and \(\lambda (\lambda > 0)\) is a parameter of the combining ratio.

### 4 Linear channel interpolation (LCI)

This section proposes LCI in order to track time-varying channels. LCI derives the estimated CIR \(\hat{h}_k^{\text{LCI}}\) interpolating the estimated CIR \(\hat{h}_k^{\text{pre}}\) and \(\hat{h}_k^{\text{post}}\), where \(\hat{h}_k^{\text{pre}}\) is estimated from SW at the present frame, and \(\hat{h}_k^{\text{post}}\) is estimated from SW at the next frame. Let us define that \(T_{\text{frame}}\) is a frame duration, \(k'(0 \leq k' \leq T_{\text{frame}})\) is the discrete time number at the present frame. The proposed LCI tap coefficient \(\hat{h}_k^{\text{LCI}}\) is derived as follows:

\[
\hat{h}_k^{\text{LCI}}[i] = \frac{T_{\text{frame}} - k'}{T_{\text{frame}}} \hat{h}_k^{\text{pre}}[i] + \frac{k'}{T_{\text{frame}}} \hat{h}_k^{\text{post}}[i].
\]

### 5 Computer simulation

This section compares packet error rate (PER) performance of MCC-LVE and SQR-DFSE, where MCC-LVE is the LVE with the MCC scheme \(\Gamma_k^{\text{MCC}}\), SQR-DFSE is the DFSE with only the squared Euclidean metric-criterion \(\Gamma_k^{\text{SQR}}\). We also evaluate influence of LCI. This paper assumes that a modulation scheme is OQPSK, transmit/receive filters are 20% root cosine roll-off, the number of received antennas \(N_R\) is 1 and FEC is Reed-Solomon code \((240, 192, 8)\). Channels suffer from 2-path independent Rayleigh fading with the delay spread of 32 bits and desired to undesired signal power ratio of 6 dB, where the maximum Doppler frequency normalized by symbol rate, \(f_D T\), of 0% corresponds to quasi-static fading channels. FFF-DFE with RLS algorithm in a decision-directed mode is also evaluated for reference, where the number of feedforward filter taps is 64 and the number of feedback filter taps is 64. Fig. 2(a) shows PER performance as a function of average \(E_S/N_0\) on quasi-static channels, Fig. 2(b) shows PER performance in the absence of noises as a function of \(f_D T\). From Figs. 2(a) and 2(b), we can obtain the following results:

- MCC-LVE has better PER performance than SQR-DFSE;
- the equalizers employing the proposed LCI have better tracking performance than the others;
- FFF-DFE suffers from PER floor on the quasi-static channels; this is because the tap coefficients do not sufficiently converge, and FFF-DFE suffers from the error propagation.

Thus, computer simulation results show that the proposed MCC-LVE employing the LCI can improve not only tracking performance but also PER performance in the required signal to noise power ratio (SNR), keeping small computational complexity.

6 Field experiment

This section evaluates PER performance of the proposed LVE on actual field experiment. In this experiment, a carrier frequency is 22.6 kHz and a transmission rate is 8 ksps. We employ PM0.3H (FOSTEX) of a speaker and LS-100 (OLYMPUS) of a microphone. Fig. 3(a) shows a sounded CIR on quasi-static channels where the distance between the speaker and the microphone is 10 m and Fig. 3(b) shows PER performance as a function of $f_D T$ at the same point. This section evaluates the same equalizers in 5. From Fig. 3(b), we can obtain the following results:

- similar to the simulation results, the equalizers employing the proposed LCI improve tracking performance;
- in actual fields experiment, FFF-DFE seriously degrades tracking performance; FFF-DFE seems to suffer from severe error propagation compared with the simulation result.

These results confirm that the proposed LVE is suitable for acoustic communications environment.
 Conclusion

This paper has proposed the LVE employing the LCI for large delay time-dispersive channels of acoustic communications. It can reduce computational complexity without performance degradation. In addition, computer simulation results have confirmed that the proposed LVE employing the LCI shows better tracking performance and better performance in the required SNR than FFF-DFE employing RLS algorithm. Finally, field evaluation results have confirmed the proposed LVE is suitable for acoustic communications.

Acknowledgments

This work was partly supported by JSPS Grants-in-Aid for Scientific Research (KAKENHI) Grant Number JP16K06374.