DESIGN AND IMPLEMENTATION OF AN EKF-BASED TRACKING SYNCHRONISM SCHEME FOR AN INDOOR DS/CDMA SYSTEM

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Abstract.- In a direct-sequence code-division multiple-access (DS-CDMA) system, code synchronisation is essential for the demodulation of the received signals from different users. In this paper, we propose two different structures based on the Extended Kalman Filter (EKF) for code synchronization by means of channel delay estimation. Performance of both schemes in terms of the normalised error jitter and the Mean-Time to Lose-Lock (MTLL) are shown through results obtained via extensive computer simulations under indoor mobile channel conditions. These results have been also validated by implementing a DSP-based DS/CDMA system demonstrator built for this purpose.

I. INTRODUCTION

In a direct-sequence code-division multiple-access (DS-CDMA) system, code synchronisation is a key factor of the system performance. Nevertheless, the combination of deep fading and large Doppler shifts in spread spectrum mobile communication channels leads to severe performance degradation in such systems.

DLL-based synchronism schemes have been traditionally considered for this purpose although these schemes have not been conceived for operation under Rayleigh multipath fading [1]. A first attempt to introduce specific synchronizers for operation in those environments was presented in [2], where an Extended Kalman Filter structure was proposed to jointly estimate the PN code delay in tracking and the channel impulse response. Nevertheless, this approach fails in the presence of high-interference environments with high processing gains, which are typical in mobile communication systems. The specific problem of optimal synchronization in a direct-sequence spread-spectrum receiver is also considered here as a particular application of the Extended Kalman Filter (EKF) delay estimator, but the scheme proposed in this paper have been conveniently modified in order to improve its performance in terms of convergence and other quality factors.

In particular, two different structures based on the EKF approach are proposed in this paper for code synchronization by means of channel delay estimation techniques. The difference between these two schemes lies in the place where the input samples to the EKF are taken. Performance of both schemes in terms of jitter and Mean-Time to Lose-Lock (MTLL) are shown through results obtained via extensive computer simulations under realistic indoor mobile channel conditions for square waveforms. Finally, these results have been also validated by implementing one of these schemes on a simple DSP-based demonstrator of a DS/CDMA system built for this purpose.

The organisation of this paper is as follows. In section II, the EKF theory is first revised, and the EKF-based schemes proposed for delay estimation as well as the complete system model are described in detail. Section III is devoted to present the simulations and system emulation implemented in order to assess the performance of the proposed schemes (in terms of the tracking error jitter and the MTLL) in an indoor mobile environment under a first-order AR model of channel delay

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generation. Finally, in Section IV some conclusions are drawn.

II. SYSTEM DESCRIPTION

A. *EKF formulation.*

The Kalman filter is a linear, discrete-time, finite-dimensional system. It can be understood as a feedback system, with the forward part driven by the innovations, which are a white noise sequence. The Extended Kalman Filter (EKF) is similar to a standard and linear Kalman filter for a nonlinear model that, if sufficiently smooth, can be approximated by a linear function by means of Taylor series.

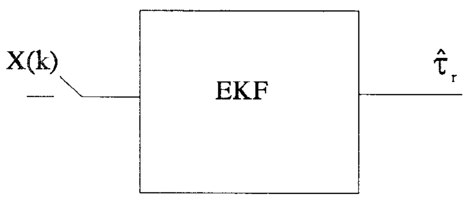

Fig. 1. The Extended Kalman Filter as delay estimator.

Given the X(k) samples of the signal at the input of the EKF, the filter task is to obtain a delay estimate. In the rest of the paper, the channel delay is assumed to follow a first-order AR model [1]:

$$\tau_r(k+1)=\tau_r(k)+w_{\tau_r}(k) \tag{1}$$

where w_{τ_r} is a white gaussian process with zero mean and σ^2 variance.

The equations that rule the Kalman filter behaviour, known as measurement up-date equations, are given by [3]:

$$\begin{aligned}
&\hat{\tau}_r(k/k)=\hat{\tau}_r(k/k-1)+K(k)[r(k)-H(\hat{\tau}_r(k/k-1))]\\
&K(k)=\frac{P_{k/k-1}\frac{\partial}{\partial\tau}H^*(\hat{\tau}_r(k/k-1))}{|\frac{\partial}{\partial\tau}H(\hat{\tau}_r(k/k-1))|^2P_{k/k-1}+\sigma_n^2}\\
&P_{k/k}=[I-K(k)\frac{\partial}{\partial\tau}H(\hat{\tau}_r(k/k-1))]P_{k/k-1}
\end{aligned} \tag{2}$$

where σ_n^2 is the variance of n(k), K(k) and $P_{k/k}$ are known as the gain vector and the error covariance matrix respectively, and the function $H(\tau(k))$ is given by

$$X(k)=H(\hat{\tau}_r(k/k-1))+\sum_{n=1}^{M}n_n(k)=H(\hat{\tau}_r(k/k-1))+n(k) \tag{3}$$

In addition, the prediction equations, according to the first order AR model that has been considered, can be expressed as:

$$\begin{aligned}
\hat{\tau}_r(k+1/k)&=Re[\hat{\tau}_r(k/k)]\\
P_{k+1/k}&=P_{k/k}+\sigma^2
\end{aligned} \tag{4}$$

B. *System model.*

The Rake receiver allows for a constructive recovering of signals that travel through different propagation paths to the receiver, provided that the delay between two different paths is larger than the system resolution, which is approximately given by the chip rate [4]. However, the delay spread of the channel in indoor environments is so low that it results extremely difficult to resolve more than one path. In order to have a better system performance by resorting to a diversity mechanism (as the use of a Rake receiver), a second propagation path has been artificially created by the use of a second antenna that transmits the signal with a delay of T_R seconds with regard to the main antenna.

In this situation, a two-ray time-variant mobile channel model with independent Rayleigh fading for every path has been considered for performance analysis purposes:

$$h_c(t,\tau')=\sum_{m=0}^{1}\alpha_m(t)\delta(\tau'-mT_R+\tau_r(t)) \tag{5}$$

where α_m represent the complex channel amplitudes of the propagation paths and τ_r the delay introduced by the channel with regard to the time reference at the receiver. The performance of the synchronism schemes proposed in this paper has been analyzed under the assumption of an open-loop power control of signals for the forward link (up-link) and an indoor environment with an equivalent low-pass Doppler spectrum, $S_c(f)$, given by [5]:

$$S_c(f)=\left[1-\left(\frac{f}{f_d}\right)^2\right]^{-\frac{1}{2}} \tag{6}$$

where f_d is the maximum Doppler shift of the channel.

A two-arm Rake receiver able to combine the signals from the two different paths has been adopted at reception. PN sequences and CBPSK have been considered as the spreading codes and the signalling scheme employed in this analysis respectively.

Let d(t) be the data sequence transmitted by the desired user, c(t) the PN sequence assigned to this user and S the mean power at transmission. The signal at the receiver input can be expressed as:

$$r(t)=\sum_{n=0}^{L-1}\alpha_n(t)s(t-nT_c+\tau_r)+n(t) \tag{7}$$

where n(t) represents the interfering power coming from other simultaneous users and s(t) is the transmitted signal, which can be expressed as

$$s(t)=\sqrt{2S}\,d(t)c(t) \tag{8}$$

In order to estimate the delay τ_r, a feasible solution might be to sample the received signal, r(t), every T_c seconds and to use these samples as the input of the Kalman filter (as a direct application of the Kalman theory) [2]. But taking into account that the interfering power-to-signal ratio at the receiver input is usually very small in the base station of a DS/CDMA system, under this assumption the synchronism would be lost in a very short period of time, as we could experience from the results obtained when several simulation chains were carried out using such a strategy.

With the aim of improving the estimator performance, there are two different approaches to solve the problem of convergence mentioned before. A first feasible solution results from the fact that the estimator is working in tracking, and so the delay τ_r, although unknown, is shorter than half a chip interval. This first procedure proposed to solve the convergence problem can be summarised as follows:

1. To multiply the received signal coming from each propagation path by the PN sequence generated in reception, delayed τ_r seconds with regard to the received sequence, and to filter low-pass the resulting signal:

$$X_n(k)=\int_{kT_b}^{(k-1)T_b} r(t+nT_c)c(t)dt=\sqrt{2S}\,\alpha_n(k)(T_b-N|\tau_r(k)|)d_k+n_n(k) \tag{9}$$

where N is the PN sequence length and T_b is the bit interval. Let us note that square pulses have been considered as waveforms in (9).

2. To add the output of the M branches:

$$X(k)=\sum_{n=1}^{M}X_n(k)=\sum_{n=1}^{M}\sqrt{2S}\,d_k\alpha_n(k)(T_b-N|\tau_r(k)|)+n_n(k) \tag{10}$$

Fig. 2 shows the procedure described above.

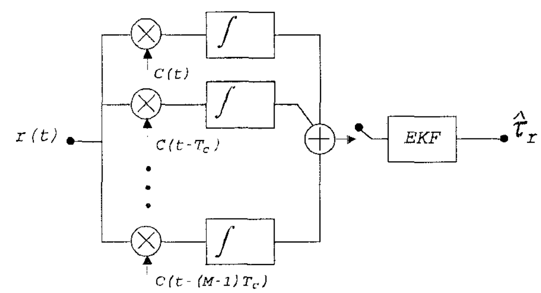

Fig. 2. Structure of the delay estimator using a signal pre-processing before the EKF.

With the X(k) samples, the Kalman filter is now able to compute the delay estimate every bit interval.

It can be demonstrated that the equilibrium point of the filter ($\hat{\tau}_r \rightarrow \tau_r$) must not be at the origin ($\hat{\tau}_r \rightarrow 0$). The physical reason for this is that the gain vector of the Kalman filter, K(k), which is proportional to the derivative of $R_c(\tau)$, is zero in the absence of noise, or not defined in the mentioned area depending on the pulse waveform employed. Therefore, if the gain vector is zero, the filter cannot follow the variations of the channel delay although the error term of the measurement update equation, $[X(k)-\hat{H}(k)]$, is not zero.

To avoid the origin area ($\hat{\tau}_r \rightarrow 0$), a control subroutine has been designed in order to force the filter to work in a suitable area. The proposed algorithm consists of having a "slightly misaligning" between the received code and the local code at the receiver.

Let us note that the structure detailed above works with iterations at the bit rate. The second procedure proposed to solve the synchronism problem works at the chip rate, and in this approach, the samples at the input of the EKF are directly taken from the received signal without the need of carrying out a signal pre-processing. To have a correct operation of this second implementation it is necessary also in this option to design a control subroutine in order to force the filter to work in the optimum area.

The election of one or other delay estimator structure will depend on the complexity or on technological factors: while the first implementation needs a kind of signal pre-processing at the EKF input, the second one works with iterations at a rate much higher than the other.

III. SIMULATIONS AND RESULTS

Extensive simulations of the whole system considering interference from other users have allow us to assess the performance of the filter for the two different structures detailed in the previous section: one of them directly takes the samples from the received signal, while the other takes them from the signal at the output of a correlator that tries to improve the performance of the synchronism scheme by increasing the SIR at the filter input.

Some system parameters have been set in our simulations as follows:

- The time interval between the two propagation paths of the mobile channel assumed for simulation purposes (see (5)): $T_R = 2T_c$.
- The maximum Doppler frequency normalized to the bit interval (see (6)): $f_d T_b$ = 5E-4.
- The bit rate, R_b = 10 Kbps.
- The number of chips/bit of the PN sequence: N=127.
- Square pulses as the waveform shaping.

System performance, basically in terms of the estimate error jitter and the Mean-Time to Lose-Lock, has been assessed via extensive computer simulations. The normalised estimate error jitter, σ_τ^2, which quantifies the tightness of the delay estimate made by the EKF, is defined as $\sigma_\tau^2=E[(\tau_r(k)-\hat{\tau}_r(k|k-1))^2/T_c^2]$. Concerning the Mean-Time to Lose-Lock (MTLL), it is defined as the mean time in which the filter is able to maintain the system synchronism without diverging. The performance of the structure with the pre-filter has been further validated by means of the implementation of a simple DS/CDMA system on a DSP-based core (see Fig. 3).

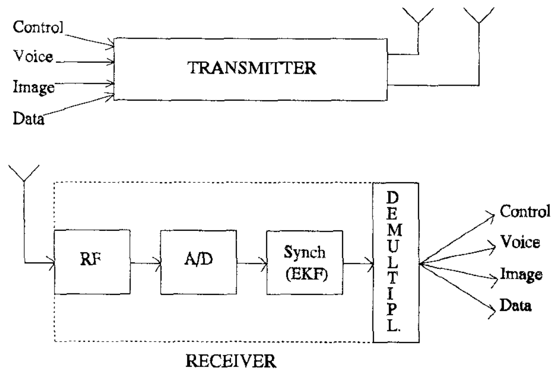

Fig. 3. Block diagram of the transmitter/receiver implemented on a DSP-based core for validation purposes.

Some representative figures have been built with the results obtained from different system simulations.

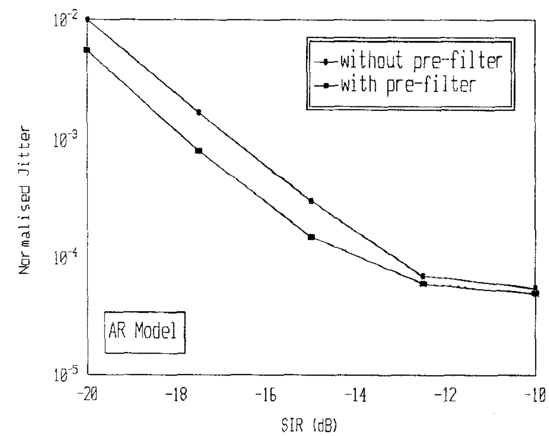

Fig. 4. System performace in terms of the normalised jitter for both synchronism schemes.

In particular, Fig. 4 and Fig. 5 have been chosen to illustrate the system performance of both structures in terms of the normalised jitter and the Mean-Time to Lose-Lock, respectively.

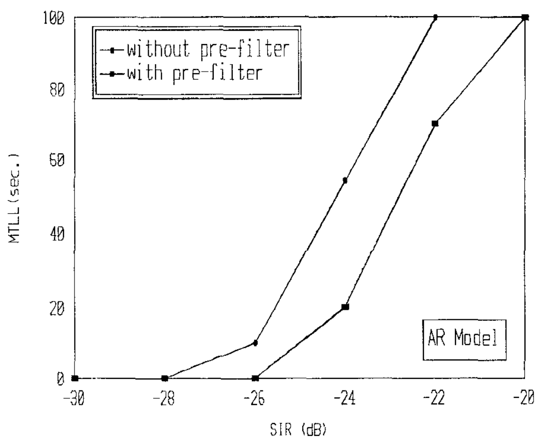

Fig. 5. System performance in terms of the MTTL for both synchronism schemes.

Fig. 4 shows how both implementations of the delay estimator are able to compute channel delay estimates with a very small jitter, even in the case of very low interference-to-signal ratios (as usual at the receiver input of an interference-limited DS/CDMA system). Fig. 5 illustrates the behaviour of both schemes in terms of the MTLL for a very low range of SIR (from -30 to -20dB). It was not feasible to compute this time of convergence for the same range of values considered to assess the jitter due to computational reasons. In any case, it may be extrapolated from this figure that for typical SIR values, the interval of convergence of the Extended Kalman Filter is large enough to ensure a correct operation as a delay estimator.

In summary, the main conclusion drawn from these two figures is that the performances of both schemes (with and without the pre-filter) are very similar. Therefore, they are very suitable for the support of code synchronisation in the tracking phase for a DS/CDMA system operating in a Rayleigh multipath fading environment. As the sampling rate is different in both cases, the election of one of them in a real implementation will depend on the state of the art of the technology used to design the system.

The DSP-based demonstrator built for validation purposes confirmed the feasibility of the estimator scheme with the pre-filter with results very close to those obtained via simulations.

IV. CONCLUSIONS

Two different schemes based on the Extended Kalman Filter have been proposed for tracking delay synchronism in a DS/CDMA system. In one of these two approaches, the EKF directly works with the samples of the received signal, while in the other, the EKF works with the samples of the signal at the output of a correlator, the task of which is to improve the performance of the delay estimator by increasing the SIR at the filter input. A control subroutine is needed in both cases to operate in the optimum working area.

From several simulations it has been shown that the proposed schemes have similar performance (in terms of the normalised error jitter and the Mean-Time to Lose-Lock) in an indoor mobile environment with Rayleigh multipath propagation. The results obtained from these simulations confirm their validity as delay estimators in a tracking synchronism scheme for a DS/CDMA system of third generation.

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