Preamble design for symbol synchronisation in frequency-selective fading channels

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Abstract: In this study, a two-step method for designing multi-tone preambles is proposed and its performance in the frequency-selective fading channel is optimised. The first step is to find an analytical solution for obtaining the optimum normalised tone frequencies of the multi-tone preamble in order to perform better in frequency-selective fading channel. To measure the quality of the preamble, the authors also propose a timing metric that provides a measure of impulsiveness of the auto-correlation function of the preamble. The second step is to de-normalise the tone frequencies to obtain the best performance in the presence of additive white Gaussian noise regarding a specific number of tones in a specific frequency interval. The tone frequencies are de-normalised using a factor that maximises the impulsiveness of the auto-correlation of the preamble. To compare the authors’ work with other methods, computer simulations are done using Monte Carlo method and the mean square error of the timing estimation is then calculated. Their work is mostly suitable for use in applications with high signal-to-noise ratio (SNR) such as M-quadrature amplitude modulation high rate links because they show that their method outperforms other methods in high SNR scenarios.

1 Introduction

Symbol synchronisation is the process of finding the instant at which the signal arrives. There are two main methods for performing synchronisation; namely, data-aided and blind synchronisation. Data-aided methods insert a known preamble in front of each data packet such that the preamble could be easily used by the receiver to achieve synchronisation. Blind methods usually exploit the periodic structure of cyclic prefixes or orthogonal frequency division multiplexing (OFDM) frames to accomplish the estimation task. Each of these methods has its own advantages, but in this paper we focus on data-aided methods.

Generally, timing estimation involves finding the maximum of the cross-correlation of the received preamble and the original preamble at the receiver. In this way, the instant of occurrence of the maximum is the timing estimation. In the rest of this section methods developed for synchronisation are covered.

Schmidl and Cox [1] proposed a synchronisation method based on preambles for OFDM. This method uses a preamble containing two identical halves. Although this preamble provides a substantial time and frequency synchronisation potential, the timing of Schmidls method has a plateau, causing large variances in the timing estimate.

Minn et al. [2] modified Schmidl’s preamble to reduce this uncertainty. The modified preamble proposed by Minn improved estimation performance in low signal-to-noise ratio (SNR) scenarios with the impulsive characteristics of the timing metric.

Park et al. [3] proposed a method using a central symmetric sequence. This method provides pulse-like shapes in timing metric; however, it does not deal with the frequency offset. Therefore Park’s method cannot estimate the frequency offset common for frequency-selective fading channels.

Umari et al. [4] used a pseudo-noise-based preamble for synchronisation. Obviously, auto-correlation of random signals is similar to an impulse; therefore by using pseudo-noise signals as preambles, it is possible to sharpen the peak of the cross-correlation at the receiver. This solely enhances the preamble performance in low SNR conditions. The preamble used by Umari consists of two parts with equal lengths. The first part is a monotone and the next part is the same monotone with a random phase. This random phase is used to reduce the correlation of the first and second parts of the preamble. In this way, the average value of the cross-correlation of the received preamble and its original version at the receiver is diminished and impulsiveness of the cross-correlation which is the timing metric in Umari’s method increases. This method uses a single tone and the possibility of missing the incoming tone at the receiver is high because of severe distortions in specific instants in frequency-selective fading channels which cause problems for synchronisation. Our work is mostly similar to Umari’s work, but we have developed a method for designing multi-tone preambles such that synchronisation would be possible in very poor frequency-selective fading channels.

Zhou et al. [5] used a central symmetric preamble that is structured as a sequence of time samples having four main parts as the vector [C D C D] in which D is a sequence of samples and is a rotated reverse conjugate version of the sequence C. They used a metric to synchronise the receiver and the transmitter. The metric used is the product of the
symmetric correlation profile and the delayed correlation profile of the received preamble that was normalised with respect to the power of the received preamble.

Schellmann [6] analytically investigated the multi-periodic preambles for synchronisation in OFDM. Schellmann shows that the multi-periodic preamble is able to choose the correlation shift and the correlation window length independently. This is very suitable while adapting the synchronisation process to the desired carrier frequency (CFO), estimation accuracy and noise suppression.

Kang et al. [7] proposed a preamble-based synchronisation for OFDM-based systems with an arbitrary preamble. They used a circular shift operation on the given preamble at the receiver and thus, in a sense, they made their method independent of the preamble. The most significant advantage of this method is that it obtains an impulsive correlation feature from any given preamble which enables the method to perform well in low SNR channels. Compatibility of this method with a wide range of preambles is also of great significance.

Gul et al. [8] use the Zadoff–Chu sequences introduced in Q2 [9, 10] as the preamble for timing estimation. Zadoff–Chu sequences are non-binary unit-amplitude sequences, which satisfy a constant amplitude zero auto-correlation property. This kind of sequence is widely used in long-term evolution [11]. One of the most important merits of this sequence is that its peak to average power ratio (PAPR) is very low. Low PAPR provides the capability of using this sequence in high-efficiency non-linear power amplifiers such as class E or F [12].

Hsieh and Wu [13], Van de Beek et al. [14], Zhang [15] and Lee and Seo [16] introduced maximum likelihood (ML)-based method for synchronisation. ML-based synchronisation methods are computationally high cost, but suitable for using in additive white Gaussian noise (AWGN) and slow-fading channels because a good performance is obtainable with a reasonable computational cost. However, some low-complexity ML-based synchronisation methods are also introduced in [17–20]. Rotoloni et al. [17] proposed two lower-complexity solutions for digital video broadcasting standard DVB-T2 [21]: (i) an ML estimator that only exploits the time structure of a specific symbol for time synchronisation and (ii) a pseudo ML scheme that resorts to a suboptimal CFO estimator while still performing ML time synchronisation. A modified ML-based synchronisation method is also introduced in [18] which reduces the complexity; however, the authors of this paper only simulated their proposed method in AWGN channel and did not illustrate the performance of their method in fading channels. Wu and Serpedin [19] proposed a low-complexity symbol-timing estimator based on conditional ML principle. They approximated the likelihood function to obtain a low-complexity estimator and showed that the proposed estimator can be viewed as a generalisation of the square non-linearity estimator. Wang and Hu [20] proposed a closed-form ML-based acquisition algorithm for carrier and sampling frequency offsets in OFDM systems using two long training symbols in the preamble. They developed an approximate ML estimator for the sampling frequency offset by taking the second-order Taylor series expansion which reduces the complexity greatly.

Two ML-based methods are also introduced in [22, 23] for using in poor communication channels. Lin [22] proposed an ML method for frame timing and frequency offset estimation in an OFDM transceiver by taking advantage of cyclic prefixes. He approximated the log-likelihood function and developed a method that performs accurate frame timing estimation before the carrier offset is estimated. He used an estimator for the CFO which is very similar to that proposed previously [14] and proposed a new estimator for the frame timing instant that reduces the hardware and computational complexity. In addition, synchronisation performance is improved in his work by combining multiple symbol observations spanning a series of OFDM frames. Chin [23] proposed an ML-based joint synchronisation method that takes advantages of the distinctive correlation characteristics of OFDM signals. In this method, non-linear operations of the approximate solution are implemented using a lookup table to reduce the complexity such that a low-complexity algorithm [(ON)] can be obtained. He proposed a CFO estimator similar to that presented in [14] which utilises the delay spread of dispersive fading channels which is a factor that limits the performance of conventional methods.

All of the methods discussed so far, including ours, deal with the design of preambles; however, Chen et al. [24] deal with other aspects of synchronisation. They proposed a method for synchronisation in fading channels in which a new correlation function is proposed. This correlation function reduces the effect of inter-symbol interference regardless of the preamble used. They also developed a method to track the delay of the channel and showed that their method performs faster than the conventional ones.

There are also blind methods for timing estimation introduced in [25–27] which make use of cyclic prefixes for primary synchronisation. Another blind method was introduced by Sun et al. [28] that works based on frame detection of OFDM systems in frequency-selective fading channels. They proposed two simple frame detectors by applying log-likelihood ratio testing, and different approaches for considering statistical characteristics of channels.

In this paper, we propose a preamble structure which is designed and optimised for frequency-selective fading channels. Our method mostly deals with the frequency behaviour of the preamble and performs well in frequency-selective fading channels as well as in low SNR conditions. We aim to decrease the sensitivity of symbol synchronisation with respect to the distortions caused by frequency-selective fading channels.

### 2 Multi-tone preamble approach

#### 2.1 For frequency-selective channels

Synchronisation in frequency-selective fading channels is one of the most challenging issues in wireless communications, which is caused by the presence of anomalies and disturbances in the atmosphere. The main problem is that regardless of the synchronisation algorithm, the preamble signal fades at some frequencies, leading to the possibility of losing the incoming signal. The reason why the preamble is incapable of synchronising the transmitter and the receiver is that some or all frequency components of the preamble fade because of the unpredictable time-frequency behaviour of fading channels. Multi-tone preambles are developed to avoid this problem. Such preambles consist of two or more monotonies in different regions of the bandwidth. The more diverse the tone frequencies, the more the chance to avoid losing all frequency components in the frequency-selective channel. The most important merit of
multi-tone preambles is the diversity of tone frequency and dividing the whole preamble power between some narrow regions of bandwidth. In this way, the fading phenomenon which is very common in frequency-selective channels cannot eliminate all of the preamble components and synchronisation is possible even if some tones are omitted. Unfortunately, it is not possible to relate the position of notch-filters in the frequency-selective channel with the design of the preamble because their position is totally random.

A simple and common structure for a multi-tone preamble is represented in (1)

\[ p(t) = \sin(\omega_1 t) + \sin(\omega_2 t) + \cdots + \sin(\omega_T t) \]  

The synchronisation is done by sending this signal before the information signal to the receiver and finding the instant at which the cross-correlation of \( p(t) \) and the arriving signal reaches its maximum. If the peak is larger than a threshold, the presence of a meaningful signal is realised and the synchronisation is done by finding the time at which the peak has occurred.

Assume that the signal sent is \( p(t) \) and the arrival signal is \( r(t) \) in multipath fading channel we have

\[ r(t) = \sum_{n=-N}^{N} a_n e^{j\phi_n} p(t - \tau_n) \]  

In which \( a_n \) is the attenuation coefficient, \( e^{j\phi_n} \) is the random phase and \( \tau_n \) is the delay of the \( n \)th path in the multipath channel. This equation implies that there would be an intolerable distortion in the fading channel which is modelled as a Rayleigh or Rician channel [29]. Moreover, Gaussian noise is also present and because of all of these distortions \( p(t) \) could not be found easily. There might be a situation that a large number of notch-filters exist in the channel. In such a situation, none of the methods listed in the previous section are able to synchronise the receiver and the transmitter because the preamble undergoes severe deformations such that the peak of the cross-correlation of the received preamble and the original one does not reach the threshold.

To achieve synchronisation, we must use the auto-correlation of the multi-tone preamble which is a fluctuating function similar to Fig. 1. In the next sections, we elaborate on the properties of this function and improving the timing estimation.

3 Proposed method for designing

3.1 Multi-tone preamble

Assume that the frequency interval used to transmit the signal is \([a, b]\). On account of the fading phenomenon of frequency-selective fading channels, some regions of the introduced bandwidth may eliminate some components of the preamble. In fact, we can consider the channel as a time-varying filter which is a composition of a number of notch-filters randomly distributed in the entire interval \([a, b]\). We may choose the frequencies of each tone of the preamble in a way that each elimination region (notch-filter) at the frequency interval \([a, b]\) cannot eliminate more than one tone of the preamble. Towards this aim, one of the criteria for choosing the tone frequencies is that they should be placed far away from each other so that each notch-filter would only be able to disturb a single tone of the preamble.

The other criterion regarding the number of tones is that the tone frequencies should not be close to the allowed frequency limits because those regions usually undergo distortions associated with the bandpass filter that is used to allocate the band \([a, b]\) to the user.

Assuming both of these criteria have the same cost, tone frequencies should be chosen in a way that they would be as far as possible from not only each other but also from the end points of the interval \([a, b]\). For example, suppose that a multi-tone preamble is composed of two monotonies. The associated frequencies are represented by \( f_i \) and \( f_j \). Suppose that the lower-frequency limit \( a \) is 0 Hz. Considering the above criteria, tone frequencies must avoid the following set, shown here as three straight lines in \( f_i - f_j \) plane

\[ \{f_i; f_i = 0\} \cup \{f_i; f_j = 0\} \cup \{f_i; f_j = f_i\} \]

For optimal results, one should avoid choosing the above sets as well as all of their combinations, and also maximise the distance between the chosen point and these three lines on the \( f_i - f_j \) plane. Suppose that \( f_i \) is the frequency of first tone (\( T_1 \)) and \( f_j \) is the frequency of the second tone (\( T_2 \)); the multi-tone preamble is then \( P_1 = T_1 + T_2 \) which is equal to \( P_2 = T_2 + T_1 \). The undesirable sets attributed to \( P_1 \) form two lines in the \( f_i - f_j \) plane which are \( \{f_i; f_i = 0\} \cup \{f_i; f_j = f_i\} \) and undesirable sets attributed to \( P_2 \) form two lines which are \( \{f_j; f_i = 0\} \cup \{f_j; f_j = f_i\} \).

Owing to the equivalence of \( P_1 \) and \( P_2 \) we only consider \( P_1 \), therefore we only consider \( \{f_i; f_i = 0\} \cup \{f_i; f_j = f_i\} \) as the undesirable set. Considering this assumption, the optimum frequencies should be chosen to maximise the distance of the optimum point with respect to the set \( \{f_i; f_i = 0\} \cup \{f_i; f_j = f_i\} \). Assuming the deficiency caused by both of these lines is the same, the optimum region for choosing the tone frequencies lies along the bisector of the aforementioned lines. (1, 0), (1, 1) are direction vectors of the first two lines specified in (3). Therefore the frequency vector which lies along the bisector of those two undesirable lines is obtained as follows

\[ A = \frac{1}{\sqrt{2}} \left( \frac{1}{0}, \frac{1}{1} \right) = \left( 1 + \frac{1}{\sqrt{2}}, \frac{1}{\sqrt{2}} \right) \]  

Note that (1, 0) and (1, 1) are normalised because the direction vector of lines must have the same magnitude for calculation of the bisector. Throughout this paper, the vector \( A \) is dimensionless and only shows the direction of a line which is the optimum solution for the unknown vector of tone frequencies. Using vector \( A \), the ratio of the
two-tone frequencies is obtained as follows

\[ r = \frac{1 + \sqrt{2}}{\sqrt{2}} \]  

(5)

Using this ratio between the tone frequencies, the optimum normalised frequencies can be obtained based on the method discussed above. In the rest of this section, the criteria for obtaining the optimum normalised frequencies are generalised and the next sections of this paper will mainly be dedicated to finding the optimum basic frequency for de-normalisation of the frequency vector.

Assume the multi-tone preamble contains three tones, therefore the regions that should be avoided are as follows

\[ \forall f_x, f_y, f_z: f_x = 0, f_y = 0, f_z = 0 \]  

(6)

\[ f_x = f_y, f_x = f_z, f_x = f_y, f_x = f_z = f_y = f_z \]  

(7)

In Fig. 2, the sets that need to be avoided are shown as black meshed planes. According to Fig. 2, we may search for the optimum normalised frequencies in one of the six holes that are identical from the viewpoint of our method. If we use the hole on the bottom left in order to obtain the frequencies, we must avoid the following planes

\[ \forall f_x, f_y, f_z: f_x = 0, f_y = f_z, f_y = f_z \]  

(8)

Similar to the two-tone preamble, the intersection of three bisector planes of each two undesirable planes in the hole contains normalised optimum frequencies. The locus of intersection of the three bisector planes of each two undesirable planes in a hole lies along the average of the unit intersection vector of each two undesirable planes

\[ A = \frac{(1, 0, 0)}{|(1, 0, 0)|} + \frac{(1, 1, 0)}{|(1, 1, 0)|} + \frac{(1, 1, 1)}{|(1, 1, 1)|} \]  

(9)

\[ A = \left(1 + \frac{1}{\sqrt{2}} + \frac{1}{\sqrt{3}} + \frac{1}{\sqrt{3}} + \frac{1}{\sqrt{3}} + \frac{1}{\sqrt{3}}\right) \]  

(10)

Therefore the ratio of frequencies is obtained as follows

\[ \frac{f_x}{f_z} = 0.2527, \quad \frac{f_y}{f_z} = 0.5623 \]  

(11)

In a general manner, the locus of the tone frequencies of an \( n \)-tone preamble satisfying the aforementioned constraints lies along the following vector (see (12))

\[ A_n = \left(1 + \cdots + \frac{1}{\sqrt{2}} \right) + \left(1 + \cdots + \frac{1}{\sqrt{n}} \right), \quad \cdots \]  

(12)

\[ \left[1 + \frac{1}{\sqrt{n}} \right] \]

4 Metric for evaluation of

4.1 Multi-tone preamble performance

Consider Fig. 1; if we magnify the time axis about 1000 times a fluctuating function is observed. In fact, the envelope of this function is important in the sense that the global maximum can be found using merely this envelope. Unfortunately, in noisy channels it is possible that the position of the global maximum is changed because of noise and in this way synchronisation cannot be correctly done. As a matter of fact, noise brings about changes in regular variations of the cross-correlation of the received and ideal preamble and sometimes increases the largest local maximum in a way that the local maximum becomes the global maximum. Consider Fig. 3 as an example. In this figure, the cross-correlation of the received preamble and the template preamble is plotted for a distortion-less and AWGN channel. As it is clear, the peak time changed in the AWGN channel and this is because of the Gaussian noise and the channel’s unpredictable frequency behaviour. To avoid the synchronisation time error, the global maximum must be so large that the local maximum cannot reach it even if the SNR is very low or the instant of occurrence of the largest local maximum except the global maximum must be so close to the instant of occurrence of the global maximum that if the largest local maximum became bigger than the global maximum the inserted error would be very small. Therefore a good metric for evaluating the preamble

\[ a \quad \text{Auto-correlation of preamble} \]

\[ b \quad \text{Cross-correlation of received signal and template preamble after passing an AWGN channel} \]

\[ c \quad \text{Zoomed version of} \quad a \]

\[ d \quad \text{Zoomed version of} \quad b \]
can be defined as
\[
\text{timing metric} = k \frac{h}{w}
\]  
(13)

\(K\) is a constant and \(h\) and \(w\) are defined in Fig. 4. This metric is the best in terms of defining the impulsiveness; since it is increased when the difference between the global maximum and the largest local maximum is increased, and it will also increase in case where the instant of occurrence of the global maximum and the largest local maximum get close to each other.

In other words, according to Fig. 4, the amount of timing metric defined by (13) is high in two cases. First in the case where the distance between the closest local maximum and the global maximum is very small (\(w\) is small) and second in the case where the global maximum is remarkably larger than the largest local maximum (\(h\) is large). In both cases, the timing error will be small in low SNR’s since in the first case noise may be added to the local maxima and make them larger than the true peak, but because of a small \(w\), the timing error will be very small. As for the second case, the power of noise added to the local maxima has to be very high to enable them reach the level of true peak. Therefore a large timing metric with the specified definition decreases the amount of timing error as well as the probability of error. Using this metric, it is possible to optimise the tone frequencies which are used in the structure of the preamble. In the next section, we obtain this goal. Note that frequency de-normalisation is done to approach the tone frequencies that maximise the timing metric, thus enhancing the robustness of the preamble against additive noise while the ratio of frequencies is chosen in a way to obtain the most evenly spaced frequencies in the bandwidth which solely decreases the distortion caused by notch-filters in frequency-selective fading channel.

## 5 De-normalisation of tone frequencies

Using the frequency ratio vector obtained in (12), it is possible to define the de-normalised tone frequencies as the product of vector \(A\) and a basic frequency (BasicFreq) as follows
\[
A_{de} = \text{BasicFreq} \cdot A
\]  
(14)

In which \(A_{de}\) is the vector of de-normalised frequencies and BasicFreq is the de-normalisation factor. Using the timing metric defined in (13), we find the basic frequency for a specific number of tones which provides an impulsive-shape auto-correlation function. Consider an \(n\)-tone preamble as follows
\[
p(t) = \sin (2 \pi f_1 t) + \sin (2 \pi f_2 t) + \cdots + \sin (2 \pi f_n t)
\]  
(15)

The basic frequency is obtained by maximising the timing metric. Assume that the channel which is allocated to the \(i\)th user is \([a_i, b_i]\) Hz. Therefore the available frequency interval for optimisation of BasicFreq is as follows
\[
\left\{ \left[ 1 + \cdots + \frac{1}{\sqrt{n - 1}} + \frac{1}{\sqrt{n}} \right] \times \sqrt{n} < a_i \right\} 
\]  
\cap \{ \text{BasicFreq} < b_i \}
\]  
(16)

The entire available frequency interval is then swept for finding the basic frequency at which the timing metric reaches its maximum value. Using this basic frequency, the \(i\)th preamble is uniquely defined for the \(i\)th user. Suppose that the available frequency interval for a user is limited to [300–3100] Hz for digital voice transmission. In Fig. 5, the optimum basic frequency for different number of tones is shown.

Using the reference frequencies that are obtained for preambles with different number of tones, the frequency of each tone is obtained. Assume an \(n\)-tone preamble is produced using the obtained frequencies and the preamble’s duration is 1 s. The auto-correlation of the preamble is shown in Fig. 6 for different number of tones. As it is clear

\[\text{Fig. 4 Auto-correlation of the preamble and its zoomed version about time axis}\]

\[\text{Fig. 5 Timing metric against basic frequency for different number of tones}\]
Acorr = \text{auto-correlation of the preamble could be written as follows}

\begin{equation}
\text{Acorr} = \text{Xcorr}(T_1 + T_2 + \cdots + T_n, T_1 + T_2 + \cdots + T_n) \tag{17}
\end{equation}

\begin{equation}
= \sum_{i=1}^{n-1} \sum_{j=1,j\neq i}^{n-1} \text{Xcorr}(T_i, T_j) + \sum_{i=1}^{n} \text{Xcorr}(T_i, T_i) \tag{18}
\end{equation}

The first term in (18) is the sum of the cross-correlation of each of the two tones. This term diminishes when the tones are not harmonics of a basic frequency; because in this way the tones become less correlated and the auto-correlation of the preamble gets smaller at all instants except the main peak. This, in turn, happens because the main peak is produced by the second term of (18) which is the sum of the auto-correlation of the tones. Therefore the amount of timing metric increases dramatically. Considering all of these merits using (12) is reasonable.

From another viewpoint, the designed preamble is a pseudo-random signal because of the fact that its auto-correlation is very sharp like the auto-correlation of a noise signal. The preamble could be designed using the pseudo-randomness point of view; but such a preamble would not necessarily be able to escape notch-filters that are prevalent in frequency-selective channels; since it is not designed considering the frequency behaviour. Using this approach, it is possible to produce a pseudo-random preamble that its frequency components are distributed in a way that their distance is maximum.

To put in a nutshell, maximising the distance of the frequency components decreases the probability of losing more than one tone at the same time, the phenomenon caused by notch-filters in frequency-selective fading channels. Moreover, using (12) the impulsiveness of the auto-correlation of the preamble will be increased and this way the probability of missing the correct peak time will be readily decreased.

6 Simulations and numerical results

Computer simulations are done using Monte Carlo method. The channel used for the simulation is a 3GPP channel model in series with an AWGN channel. The 3GPP channel model is (3GPP TR 25.943 V6.0.0) [30] which is implemented in MATLAB R2012, simulating the typical urban channel model and identified as 3GPPTUX in MATLAB. Performances are then evaluated using mean squared error of the difference between the estimated instant of preamble arrival and its real value. Our method is then compared with some of the effective methods previously introduced. In Figs. 7 and 8, performance of different methods is compared.

Fig. 7 shows preambles introduced in [7, 8] are very good in the AWGN channel. In Fig. 8, it is shown that our method is more robust in high SNR’s of the 3GPP channel model. The reason why our method outperforms [7, 8] in the frequency-selective fading channel is that the frequency components of the preamble are distributed in the available frequency interval such that the probability of losing more...
than one of them is minimised because of the constraints introduced in previous sections. However, Gul’s and Kang’s method are better than ours in very low SNR’s even in the frequency-selective channel. Our method outperforms other methods when frequency-selectivity of the channel causes the major and Gaussian noise causes the minor distortion on the signal.

In this section, the performance of our method in a very common channel model is presented. Another scenario could be considered in which there are a lot of notch-filters that are able to omit multiple tones of preamble. In such a situation, the only solution is to increase the number of tones. As the number of tones increases, notch-filters will omit a lower percentage of the preamble power and synchronisation would be possible.

7 References


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Q1 Please check the email id of the corresponding author.
Q2 Ref. [25] has been changed to Ref. [9] and references are renumbered accordingly.
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