Channel Estimation Error Model for SRS in LTE

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Master thesis by Pontus Arvidson

Abstract

In 3GPP long term evolution (LTE), sounding is used to gain a wideband estimate of the uplink channel. This channel estimate may then be used for several radio resource management related applications such as frequency selective scheduling and beam forming. Code division multiplexing (CDM) enables several users to broadcast sounding reference signals (SRS) simultaneously on the same time and frequency resource. As the multiplexed users may interfere with one another there is a trade-off between having users broadcast SRS as often as possible to get a frequent channel estimate and getting higher quality estimates with a lower periodicity. To assess this trade-off one must have a good understanding of what causes the errors in the channel estimate so that the sounding resource may be used as efficiently as possible.

This thesis proposes a method to model the channel estimation error with sounding for use in a system simulator environment. The method consists of estimating a standard deviation with a per-resource-block resolution of the channel estimates as a function of received signal powers of interfering users as well as the target user and background noise. This estimated estimation error may then, in the system simulator, be applied to a known ideal channel estimate as noise. The main limiting source of error is shown to be interference, both from sounding users in the same cell and in others as well as some effects of limited frequency resolution. Simulation results indicate that a cleverly designed sounding resource handler is needed to fully utilize the possible gains of sounding.

Contents

1 Abbreviations

CDD	Code Division Duplex
CDM	Code Division Multiplexing
CP	Cyclic Prefix
DMRS	Demodulation Reference Symbols
FDD	Frequency Division Duplex
FDM	Frequency Division Multiplexing
HSPA	High Speed Packet Access
LTE	Long Term Evolution
MMSE	Minimum Mean Square Error
MSE	Mean Square Error
OFDM	Orthogonal Frequency Division Multiplexing
PRB	Physical Resource Block
PUSCH	Physical Uplink Shared Channel
SC-FDMA	Single Carrier Frequency Division Multiple Access
SRS	Sounding Reference Signal
RB	Resource Block
RRM	Radio Resource Management
RX	Reception
TDD	Time Division Duplex
TTI	Transmission Time Interval
TX	Transmission
UE	User Equipment
WCDMA	Wideband Code Division Multiple Access

2 Introduction

In the LTE uplink, sounding reference signals (SRS) are used to estimate the channel between user equipment (UE) and base station. This channel estimate can then be used for several radio resource management functions such as frequency selective scheduling, link adaptation, beam forming and more.

Sounding is done by periodically having the UE broadcast a wideband reference signal to the base station from which the base station can calculate an estimate of the channel. The sounding reference signals are generated from cyclic shifts in so called Zadoff-Chu sequences. These Zadoff-Chu sequences have zero autocorrelation, which means that different sounding sequences generated from the same base sequence will be orthogonal. This allows users transmit simultaneously on the same time and frequency resource and is known as code division multiplexing (CDM). The SRS may also be separated using frequency division multiplexing (FDM). This is done by having the UE broadcast SRS on every other subcarrier in the SC-FDMA (Single Carrier Frequency Division Multiple Access) scheme used in the LTE uplink. The sounding can also be set to hop between different smaller frequency bands. Because of the orthogonality of the sequences and the frequency division, several UEs will be able to broadcast SRS simultaneously. Simultaneous sounding enables more users to broadcast SRS with a higher periodicity, and in this way get a more frequent estimate of the channel.

When implemented for real use, the channel estimate will of course never be identical to the true channel, instead it will be hindered by a number of different factors. Some of these are interference due to lost orthogonality, which may occur on a too time dispersive channel, interference from other cells or additive noise. The more users that are multiplexed on the same sounding resource, the more the interference and noise will increase the difficulty of obtaining a good channel estimate. Thus, there is a trade-off between multiplexing many users for a more frequent channel estimate and getting a higher quality estimate but not as often.

The goal of this thesis will be to create a model of the error in the aforementioned channel estimate and to examine how this error correlates in time and frequency and also how it depends on other factors such as the number of broadcasting UEs, signal power or movement speed. This model will then be used to get a more realistic evaluation of performance of the various radio resource management algorithms mentioned previously.

The model shall be created based on training data from a link simulator. The link simulator simulates one cell with one base station and a number of connected users. All signals are modeled in detail. As the link simulator provides such detailed results, it also proves impractical to increase the number of cells or users to a level where it can perform simulations where the number of users may be in the order of several hundreds. It also does not provide information for evaluation of performance from a system point of view.

Hence a simpler parametric model of the error must be created for use in a system simulator which simulates the entire LTE network.

3 An introduction to LTE

With an ever growing demand for mobile high speed communication solutions, LTE shows to be one of the most promising systems for the next generation of mobile communication with both mobile broadband and voice services. The LTE standard is set by the third generation partnership project (3GPP).

3.1 LTE design goals

Among the design targets for the first release of the LTE standard are a downlink bit rate of 100 Mbit/s and a bit rate of 50 Mbit/s for the uplink with a 20 MHz spectrum allocation. Smaller spectrum allocation will of course lead to lower bit rates and the general bit rate can be expressed as 5 bits/s/Hz for the downlink and 2.5 bits/s/Hz for the uplink. The LTE standard also supports both FDD (Frequency Division Duplex), where the uplink and downlink channel are separated in frequency, and TDD (Time Division Duplex), where uplink and downlink share the same frequency channel but are separated in time.

Another important design goal is mobility. While the highest performance will be achieved with UE movement speeds between 0 and 15 km/h, a high performance can be achieved up to movement speeds up to 120 km/h. The connection shall also be maintained for speeds exceeding 120 km/h.

The voice quality for phone calls shall be at least equal to that of WCDMA/HSPA.

Full user throughput shall be achieved in cells with a radius smaller than 5 km in non interference limited scenarios, while a slight degradation in user throughput may be accepted in cells with a radius up to 30 km. The requirements for mobility must however be met even in these larger cells.

The LTE standard shall also fully support handover to the older cellular systems GSM and WCDMA.

Finally, LTE is designed to have a large spectrum flexibility for allocation in areas with an already crowded spectrum.

3.2 LTE transmission schemes

LTE uses an OFDM transmission scheme for the downlink and SC-FDMA for the uplink. The OFDM (Orthogonal Frequency Division Multiplexing) scheme uses a rather large number of narrowband subcarriers compared to many other transmission schemes. The demodulation of each subcarrier corresponds to convolution with a rectangular wave shape which will generate side lobes of each subcarrier in the frequency domain. By cleverly spacing the different sub-carriers, so that the side lobes do not interfere with other subcarriers, the OFDM orthogonality is achieved. This is done by choosing the subcarrier frequency domain spacing as $\Delta f = 1/T_u$, where T_u is the modulation symbol time. Figure 1 shows how the frequency orthogonality prevents subcarriers from colliding with each other.

In LTE this frequency spacing is chosen as 15 kHz and a larger bandwidth allocation thus equals a larger number of subcarriers.

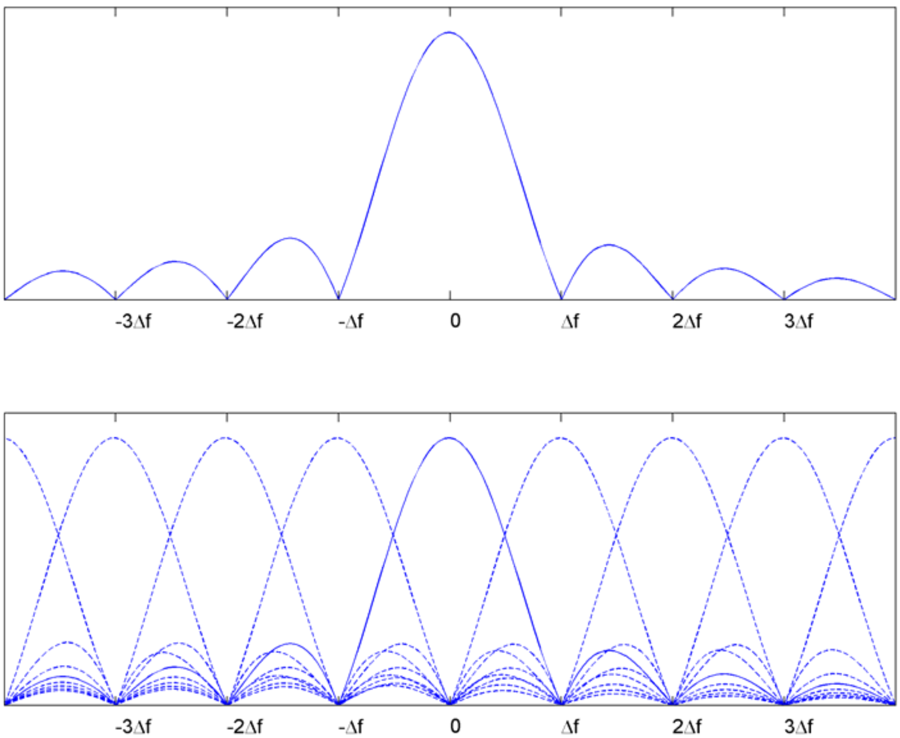

Figure 1: The above figure illustrates the OFDM frequency orthogonality due to the special subcarrier spacing used. The above plot shows the spectrum of one carrier, note how it is always equal to zero at multiples of the chosen frequency spacing. The lower plot shows the same carrier along with neighboring ones.

Even though the OFDM scheme has many advantages over other transmission schemes it proves disadvantageous to use in the uplink due to large variations in transmission power. This is due to the fact that OFDM is a multicarrier transmission scheme. Since the UE will typically have smaller resources in terms of transmission power it has larger demands on power efficiency than the base station and a less power demanding transmission scheme is to prefer for the uplink. For the LTE system, the SC-FDMA (Single Carrier Frequency Division Multiple Access) has been chosen as the uplink transmission scheme, this is because it has many of the same advantages as OFDM but has a better peak to average power ratio. The SC-FDMA uplink has a similar frame structure and the same subcarrier spacing as the downlink. The main difference is that only contiguous frequency resources are allowed to be scheduled with the SC-FDMA transmission scheme.

Both the uplink and downlink support different modulation schemes such as QPSK and 64-QAM.

3.3 Cyclic prefix

On a time dispersive channel there is always a risk of inter-symbol interference. That is, one symbol “ringing” into the next time consecutive symbol. Not quite as obvious is the risk of inter-subcarrier interference which also may occur under these conditions due to some of the frequency orthogonality being lost. Figure 2 shows how the spectrum of subcarriers can be corrupted when the time dispersion causes a previous symbol to ring the current one. This example is a rather exaggerated one but still goes to prove that inter-subcarrier interference may be a non-negligible consequence of long time dispersion.

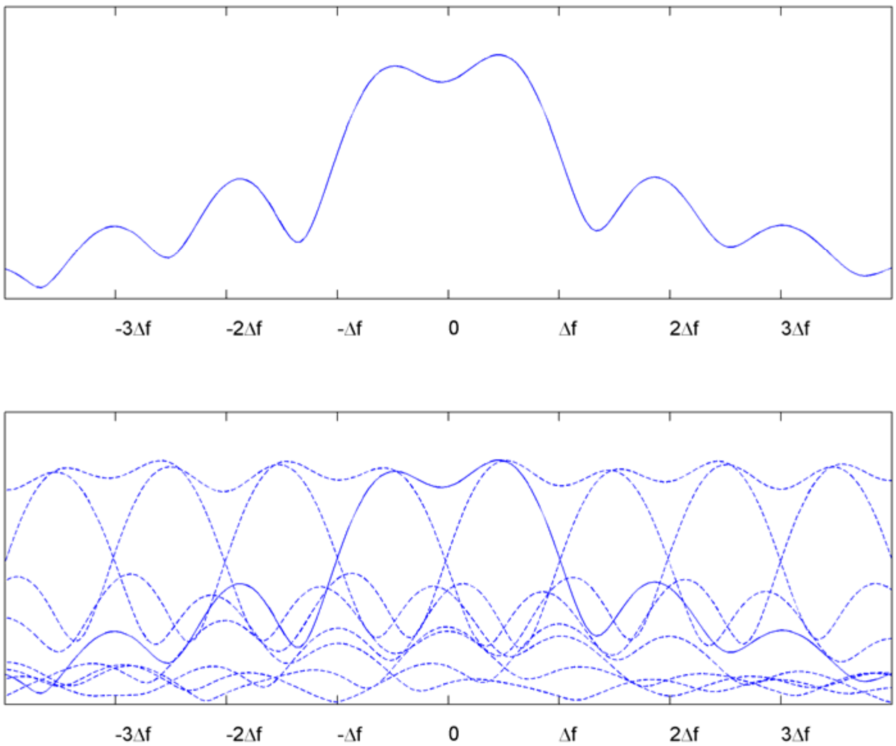

Figure 2: This figure illustrates how the spectrum of a subcarrier is affected by an extremely time dispersive channel and how the orthogonality between subcarriers is lost under these circumstances.

Both of these problems are addressed by adding a cyclic prefix to each transmitted symbol. The cyclic prefix insertion is done by copying the last part of each OFDM symbol and inserting in the beginning of that same symbol. This will increase the symbol time from T_u to $T_u + T_{CP}$, where T_{CP} is the length of the cyclic prefix. The correlation in the receiver is still done with $T_u = 1/\Delta f$. As long as the time dispersion is shorter than the inserted cyclic prefix one will avoid inter-symbol interference as well as maintain the orthogonality between subcarriers.

In cases with a very time dispersive channel, such as very large cell size, an extended cyclic prefix may be used.

While cyclic prefix does introduce a larger overhead in terms symbol length, thus decreasing the number of symbols that may be transmitted in a given time period, the gain from decreasing interference vastly surpasses this drawback. The cyclic prefix should however be chosen as short as possible to minimize the relative cyclic prefix overhead, $T_{CP}/(T_u + T_{CP})$.

3.4 The resource grid

The transmitted signal can be described using a resource grid with time, measured in number of OFDM/SC-FDMA symbols, on one axis and frequency, measured in number of subcarriers, on the other.

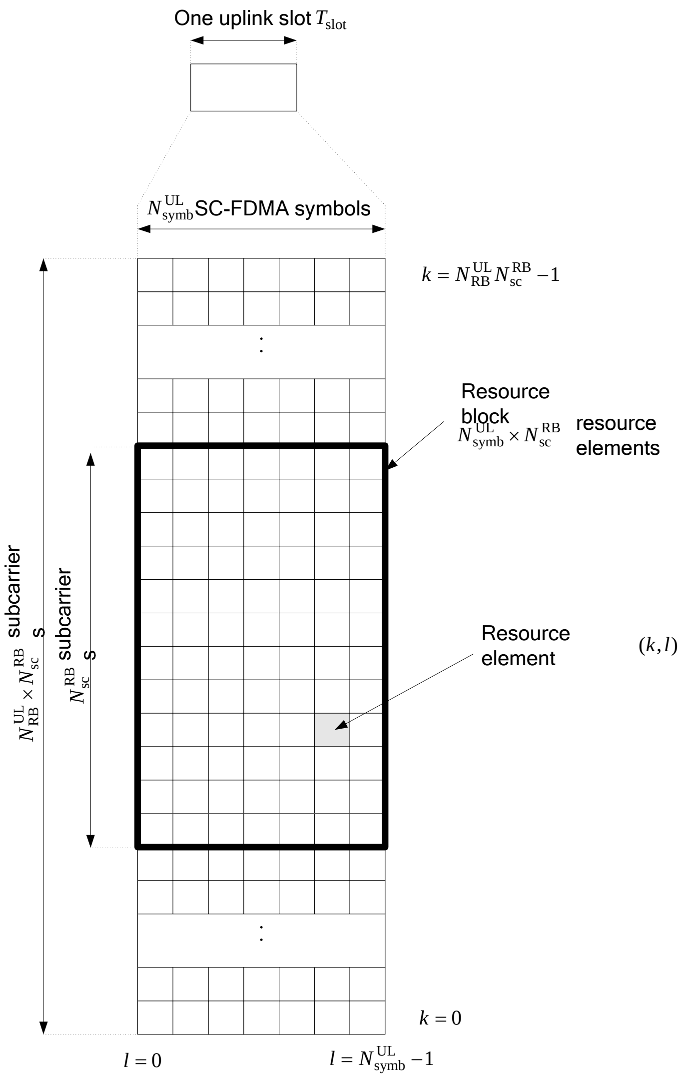

Figure 3: The LTE uplink resource grid

The smallest unit that can be scheduled is two time consecutive resource blocks. One resource block consists of N_{sc}^{RB} subcarriers and N_{symb} symbols. N_{sc}^{RB} and N_{symb} are given by the table below.

Configuration	N_{sc}^{RB} (Number of subcarriers per resource block)	N_{symb} (Number of symbols per resource block)
Normal cyclic prefix	12	7
Extended cyclic prefix	12	6

The smallest possible physical unit, one subcarrier times one symbol, is referred to as a resource element and a resource block thus consists of $N_{sc}^{RB} \times N_{symb}$ resource elements. Accordingly, one resource block has a bandwidth of 180 kHz and has a duration of what is referred to as one slot in the time domain, which translates to 0.5 ms. The amount of data that is transmitted in each resource element depends on which modulation is used and may vary between 1 bit (BPSK) and 6 bits (64QAM).

As a resource block is a unit in the physical layer it may also be referred to as a physical resource block (PRB).

4 Sounding

4.1 Basic idea of sounding

The LTE standard already contains demodulation reference symbols (DMRS) which are also used to get an estimate of the channel in order to demodulate the received symbols. However, as the DMRS are only transmitted together with data on the PUSCH (Physical Uplink Shared Channel) resource, they can only be used to obtain an estimate of the channel in frequencies currently being used by the UE for data transmissions. No information regarding the channel in other frequency bands may be gained from these.

As the base station may also utilize information about the uplink for other uses than demodulation it proves useful to have more knowledge about the uplink channel than what may be obtained from the DMRS. Sounding allows the UE to transmit reference signals to the base station over almost any arbitrary frequency from which an estimate of the channel may be obtained. This estimate may then be used for a large number of different applications such as link adaptation or frequency selective scheduling.

By means of both FDM and CDM, several UEs may broadcast SRS simultaneously. This effectively increases the number of available sounding resources and thus allows for a more frequent channel estimate which quite obviously is preferred with a time varying channel.

4.2 The SRS sequences

The sounding reference signals are generated from cyclic shifts in a common Zadoff-Chu sequence. These sequences are described in more detail in [9] and [10]. The Zadoff-Chu Base sequence is given by Equation 1.

$$a_u(n) = e^{-j\frac{un(n+1)}{N_{ZC}}} \quad ; \quad 0 \le n \le N_{ZC} - 1$$

Equation 1

Where n is the sample index, u is the sequence index and N_{ZC} is the length of the sequence. The sequence number u can be chosen between 1 and the largest prime number smaller than N_{ZC}. This means that for a prime length sequence there are N_{ZC} different unique Zadoff-Chu sequences that can be generated. The sequences will also be periodic with a period equal to the largest prime relative to the sequence length.

The Zadoff-Chu sequences are so called CAZAC (constant amplitude zero autocorrelation) sequences and these properties make them very interesting for use in radio communications. For sounding, it is especially the zero autocorrelation property that is useful. Since the sequences do not correlate over time, different sequences generated from cyclic shifts in the same base sequence will be orthogonal and can thus be separated from one another when transmitted simultaneously. The constant amplitude property is also very useful for radio reference signals in general since limited power variations allow for higher efficiency in power amplifiers. Each SRS sequence for user *m* can be calculated with Equation 2.

$$x_m(n) = e^{j\alpha_m n} a_u\left(\mathrm{mod}(n, N_{ZC})\right)$$

Equation 2

Where α_m is the cyclic shift, which in turn is given by Equation 3.

$$\alpha_m = 2\pi \frac{\mathrm{n}_{\mathrm{SRS}}^{\mathrm{cs}}}{8}$$

Equation 3

$\mathrm{n}_{\mathrm{SRS}}^{\mathrm{cs}}$ is the number of the cyclic shift assigned to user *m* and is an integer between 0 and 7.

Another interesting property of the ZC sequences is that the Fourier transform of one prime length ZC sequence is a ZC sequence as well, although scaled and complex conjugated. This means that a ZC sequence generated in the frequency domain will also have the same useful properties in the time domain.

Zadoff-Chu sequences generated from different base sequences will not be orthogonal but instead have a constant cross correlation according to the following expression:

$$r_{ZC_1\,ZC_2}(k) = 1/\sqrt{N_{ZC}}$$

Equation 4

4.3 Mapping of SRS to physical resources

The sounding sequences are generated in the frequency domain. Due to the characteristics of the Zadoff-Chu sequences described in previous sections, their useful properties are still maintained. Sounding is only transmitted on every other subcarrier and always in the last symbol of the resource block it is transmitted within. This last symbol is dedicated to only be used for sounding transmissions and data transmission within this symbol is not allowed. This means that as long as the LTE system is assumed to be synchronized within one cyclic prefix length, sounding will never collide with data transmissions.

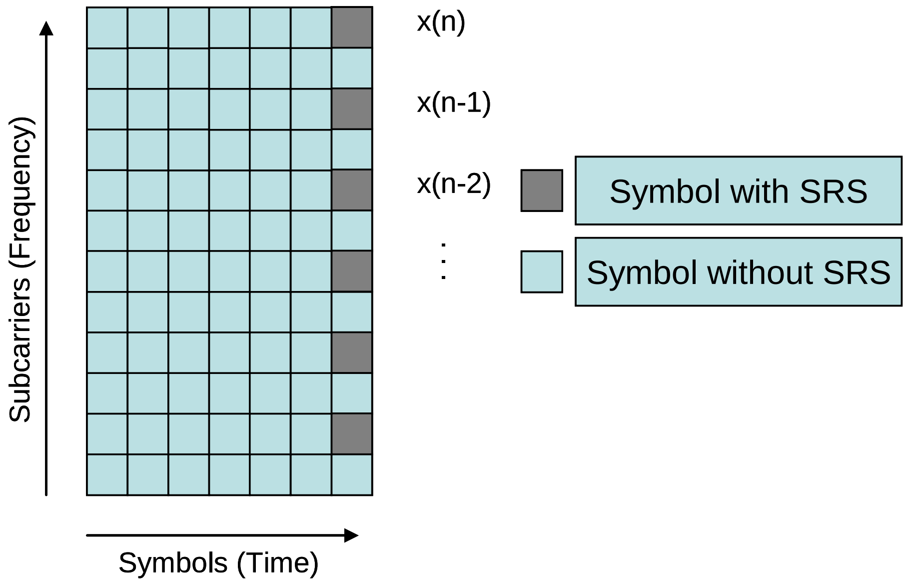

Figure 4: The above figure shows how the SRS are allocated within a resource block

4.4 Sounding periodicity

A UE may be configured to broadcast SRS as often as every second subframe (2 ms) or as rarely as every 16th frame at most (160 ms). The SRS always occupies the last symbol of the subframe it is being sent within. This is done to have a channel estimate as recent as possible in following subframes.

4.5 User multiplexing

Sounding UEs are mainly multiplexed in two different ways, either by means of code or frequency.

The frequency division is primarily achieved by allocating the SRS to a frequency comb and transmitting only on even or odd subcarriers. This way, two users may transmit SRS in the same frequency band and still be orthogonal in frequency. As the LTE system only works with a per-resource-block granularity when it comes to scheduling and similar applications, the channel estimate will be averaged over each resource block and it makes little or no difference whether a user has been sounding on one frequency comb or the other. The frequency division may also be done by simply assigning different frequency bands to sound for different UEs.

The code division is done by assigning the SRS shifted versions of a common Zadoff-Chu base sequence, as explained in more detail in 4.2, and exploiting the fact that these resulting sequences will have no mutual correlation, and thus will not affect each other.

One further way of separating users is called frequency hopping. Frequency hopping is done by, instead of having the UE sound the entire frequency band of interest in one instant, having the UE send a more narrowband SRS that is hopping in the frequency domain so that a measurement of the entire frequency band may be obtained over several periods. This scanning of the frequency band may be useful when the UE has limited resources in terms of transmission power.

This thesis will focus on the effects of code multiplexing UEs as the sounding capacity seems to mainly be limited by the number of simultaneous CDM users, multiplexed on the same frequency comb, rather than the total number of users. These phenomena will be discussed more in detail in later chapters.

4.6 Channel estimation

Which method to use for estimating the channel based on the received SRS is not specified in the 3GPP standard. Among commonly used methods for channel estimation are MMSE and ML estimates.

For this thesis the channel estimator will be assumed to be a regular MMSE estimator. The MMSE estimator strives to find an estimate which is optimal in the minimum mean square error sense.

Here follows a simple example of channel estimation for a channel on the following form:

$$y(n) = h(n) * x(n) + w(n)$$

Equation 5

Where h(n) is the impulse response of the channel, w(n) is Gaussian distributed additive noise with zero mean and "*" denotes linear convolution. On vector form the channel may be described as done in Equation 6.

$$y(n) = \begin{pmatrix} h(1) & h(2) & \dots & h(N) \end{pmatrix} \cdot \begin{pmatrix} x(n) \\ x(n-1) \\ \vdots \\ x(n-N+1) \end{pmatrix} + w(n)$$

Equation 6

The same equation may also be described by a shorter expression as shown in Equation 7.

$$\mathbf{y} = \mathbf{h} \cdot \mathbf{x} + \mathbf{w}$$

Equation 7

To create a matched filter, both sides are multiplied with $x(n)^T$ and the expected value is taken. This results in an expression described by Equation 8.

$$E[\mathbf{y} \cdot \mathbf{x}^{\mathbf{T}}] = E[\mathbf{h} \cdot \mathbf{x} \cdot \mathbf{x}^{\mathbf{T}} + \mathbf{w} \cdot \mathbf{x}^{\mathbf{T}}] = \mathbf{h} \cdot E[\mathbf{x} \cdot \mathbf{x}^{\mathbf{T}}] + E[\mathbf{w} \cdot \mathbf{x}^{\mathbf{T}}]$$

Equation 8

Since w is assumed to be zero mean and independent of x, the term $E[\mathbf{w} \cdot \mathbf{x}^{\mathbf{T}}]$ is equal to zero.

The resulting channel estimate may then be found as shown in Equation 9.

$$\hat{\mathbf{h}} = E[\mathbf{y} \cdot \mathbf{x}^{\mathbf{T}}] \cdot E[\mathbf{x} \cdot \mathbf{x}^{\mathbf{T}}]^{-1}$$

Equation 9

Exploiting the fact that the matrices in the above expression are equal to the autocorrelation matrix of x and the cross-correlation vector of x and y the expression may be re-written even further as follows:

$$\hat{\mathbf{h}} = \begin{bmatrix} r_{xy}(0) & \cdots & r_{xy}(N) \end{bmatrix} \cdot \begin{bmatrix} r_{xx}(0) & r_{xx}(1) & \dots & r_{xx}(N) \\ r_{xx}(1) & r_{xx}(0) & & \\ \vdots & & \ddots & \\ r_{xx}(N) & & & r_{xx}(0) \end{bmatrix}^{-1}$$

Equation 10

For the special case where x is known to have zero auto-correlation, as with the Zadoff-Chu sequences used for sounding, $E[\mathbf{x} \cdot \mathbf{x}^{\mathbf{T}}]$ is known to be equal to the identity matrix, $\mathbf{I}$, and $\hat{\mathbf{h}}$ may thus be found as shown in Equation 11.

$$\hat{\mathbf{h}} = E\left[\mathbf{y} \cdot \mathbf{x}^{\mathrm{T}}\right]$$

Equation 11

Due to the means by which power control is performed in LTE, the power of the transmitted signals is not perfectly known to the base station when receiving the reference signals. This results in a modified version of Equation 7 shown in Equation 12, where a is the amplitude of the transmitted signal.

$$\mathbf{y} = \mathbf{h} \cdot a \cdot \mathbf{x} + \mathbf{w}$$

Equation 12

As the matched filtering is performed assuming a unitary gain of the reference signal, the resulting channel estimate will also be scaled by the amplitude of the transmitted signal. This is shown in Equation 13.

$$h = h_{unscaled} * a$$

Equation 13

5 Modeling

5.1 Training sequences

The sequences used for studying the estimation error properties as well as for model training are generated from a link simulator. This link simulator simulates the physical layer of one cell of an LTE network with one base station and several UEs. Each set of training sequences consists of two complex valued matrices for each user indexed by time and subcarrier. The first of these matrices contains an ideal channel estimate and the second contains the actual estimator output. The estimation error for a user may then be found as the difference between the two matrices.

5.1.1 Choice of simulation scenarios

Simulation scenarios used to evaluate the behavior of the estimation error include different numbers of multiplexed users, different channel types, varying signal powers and different UE velocities.

5.2 Estimation error behavior hypotheses

To properly model the behavior and characteristics of the estimation error it is important to gain an understanding of which factors may affect the resulting channel estimate. In this chapter hypotheses about the error behavior will be discussed to determine which of these factors are relevant for modeling.

Experiments in link simulator performed in order to evaluate each of the different aspects of the estimation error behavior are also described for each subchapter. Vector valued parameter values denote parameter value being changed between iterations. All transmission powers described are given in dB relative to the thermal noise level. The different channel types referred to are specified in [7]. The experiments to evaluate the estimation error behavior have mainly been performed on an ETU (Extended Typical Urban) channel, which is a very time dispersive channel, as well as additional experiments on the somewhat less time dispersive EPA (Extended Pedestrian A) channel.

The relative values of the MSE, measured in decibels, in the figure are calculated relative to the ideal channel estimator output in accordance with Equation 14 and Equation 15. The expected value in Equation 14 is for these deterministic cases calculated as an arithmetic mean.

$$MSE = E\left[(h - \hat{h})^2\right]$$

Equation 14

$$MSE_{relative}\left[dB\right] = 10\log_{10}\left(\frac{MSE}{\bar{h}^2}\right)$$

Equation 15

5.2.1 Intra-cell Interference – CDM users

The effects of CDM user interference within a cell is highly significant on a time dispersive channel as the orthogonality between sounding users may no longer be assumed to be maintained. Simulations were run to illustrate the effects of varying signal transmission power as well which cyclic shift is assigned to the users.

The first experiment performed is designed to illustrate how the estimation error of one sounding user is affected by the signal power of one code multiplexed user sounding the same bandwidth. Simulation parameters are specified in Table 1 and the results can be seen in Figure 5.

Intracell interference experiment 1 setup	
Channel type	etu5
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	500
Number of UEs	2
Cyclic shift of UE1	0
Cyclic shift of UE2	4
Signal TX power UE1 [dB]	[-16.00, -12.00, -8.00, -4.00, 0.00, 4.00, 8.00, 12.00,16.00, 20.00, 24.00, 30.00]

Signal TX power UE2 [dB]	16

Table 1

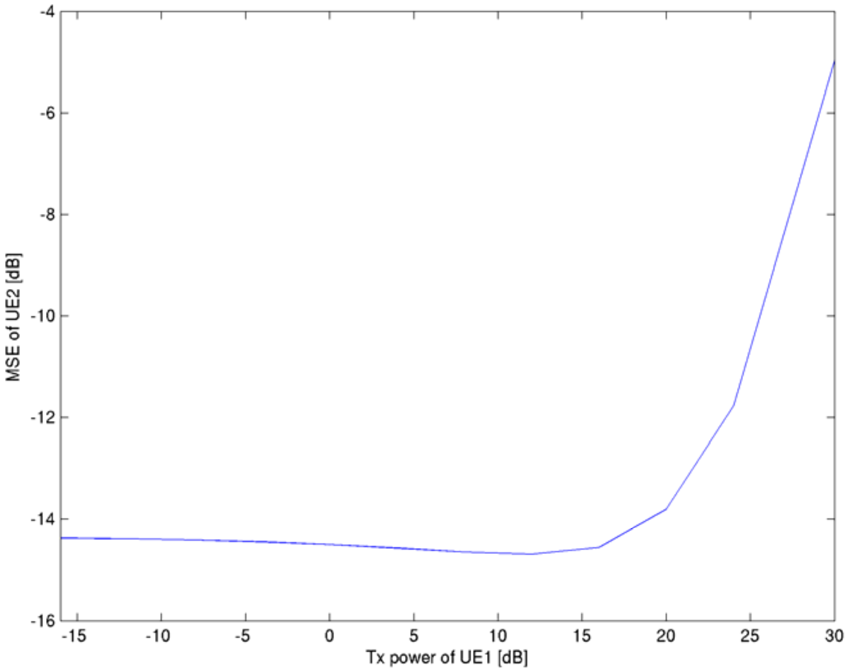

Figure 5: Intracell experiment 1, this figure shows how the MSE of one user varies with the signal power of one code multiplexed user. User 2 has constant transmission power.

Intra-cell experiments 2 and 3 are performed to illustrate how the size of the estimation error of a sounding user depends on which cyclic shift is assigned to an interfering user. Experiments 2 and 3 are performed with different channel types to further illustrate how the intra-cell interference also depends on the time dispersion of the channel.

Intracell interference experiment 2 setup	
Channel type	etu5
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	500
Number of UEs	2
Cyclic shift of UE1	0
Cyclic shift of UE2	[1, 2, 3, 4, 5, 6, 7]
Signal TX power UE1 [dB]	16
Signal TX power UE2 [dB]	16

Table 2

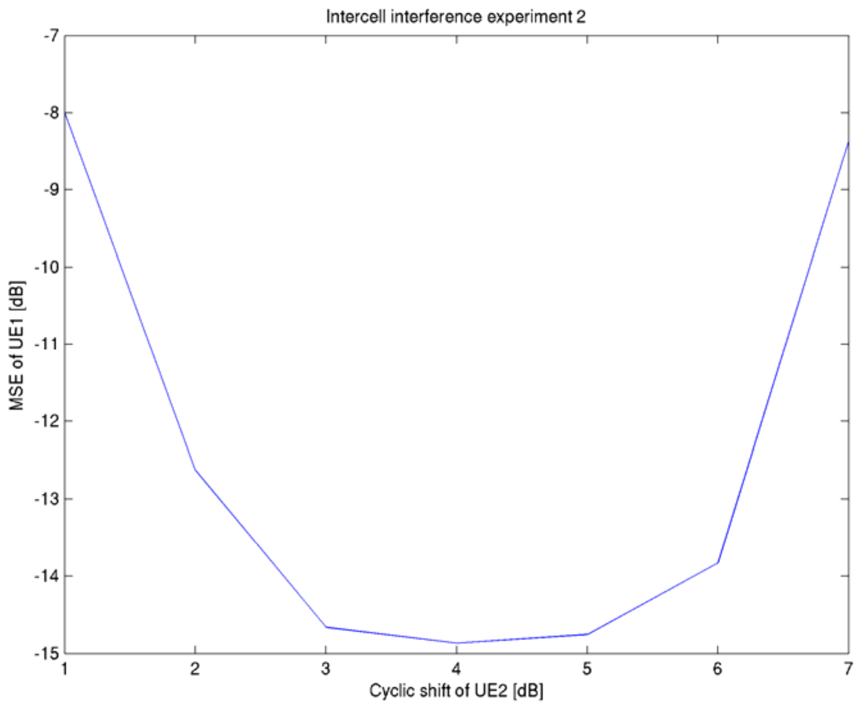

Figure 6: Intracell interference experiment 2, the figure above shows how the MSE of one UE varies with the cyclic shift of one code multiplexed UE.

Intracell interference experiment 3 setup	
Channel type	epa5
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	500
Number of UEs	2
Cyclic shift of UE1	0
Cyclic shift of UE2	[1, 2, 3, 4, 5, 6, 7]
Signal TX power UE1 [dB]	16
Signal TX power UE2 [dB]	16

Table 3

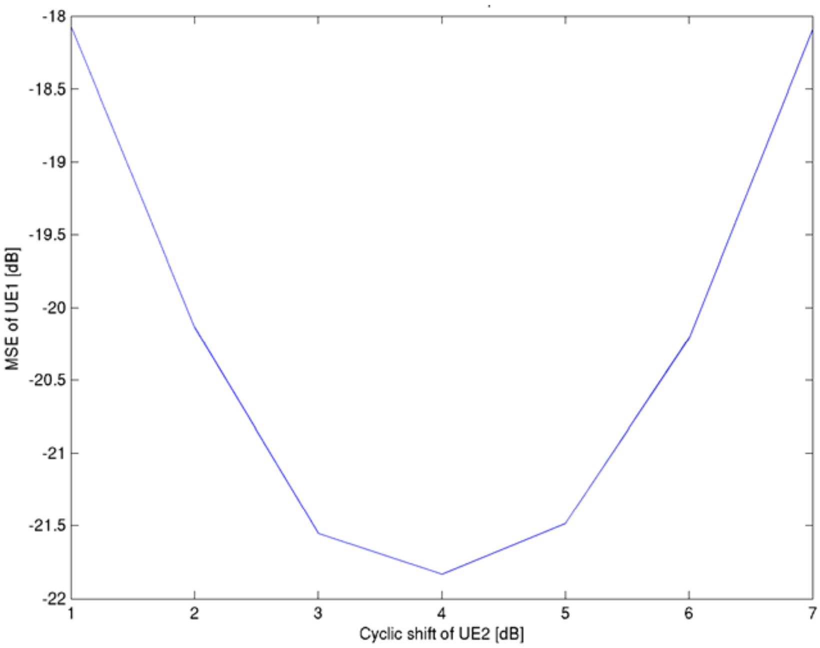

Figure 7: Intracell experiment 3, the figure above shows how the MSE of one UE varies with the cyclic shift of one code multiplexed UE. Similar to experiment 2 but with a less time dispersive channel.

The results of these experiments clearly show that intra-cell interference is a very significant factor for the size and characteristics of the estimation error. The estimation error of one user is also shown to be strongly dependent on which cyclic shift is assigned to the interfering user. The interference from a user with neighboring cyclic shift is for instance much more significant than the interference of a UE with a relative cyclic shift equal to four, which is the largest possible distance as the sequences are periodic and eight different shifts exist. A comparison between the results of intra-cell experiments 2 and 3 also clearly shows that the interference between same-cell sounding users is much more significant on a more time dispersive channel.

One way of describing the effect one sounding UE has on another seems to be with an unorthogonality factor. This factor depends on the cyclic shift of the sounding UEs as well the chosen channel model. This unorthogonality factor may act as a scaling factor to describe which UE affects which. A small distance in cyclic shifts will thus result in a larger unorthogonality factor, and a larger distance results in a smaller factor.

This unorthogonality factor may, if estimated accurately, be used to describe the distribution of error contributors in the final model.

5.2.2 Intra-cell interference - FDM users

To study the effects of interference from a sounding UE transmitting sounding on the other frequency comb an experiment where the MSE of one user was studied when varying the signal power of a FDM user.

FDM user interference experiment setup	
Channel type	etu5
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	500
Number of UEs	2
Cyclic shift of UE1	0
Cyclic shift of UE2	0
Transmission comb of UE1	0
Transmission comb of UE2	1
Signal TX power UE1 [dB]	16
Signal TX power UE2 [dB]	[-16.00, -12.00, -8.00, -4.00, 0.00, 4.00, 8.00, 12.00,16.00, 20.00, 24.00, 30.00]

Table 4

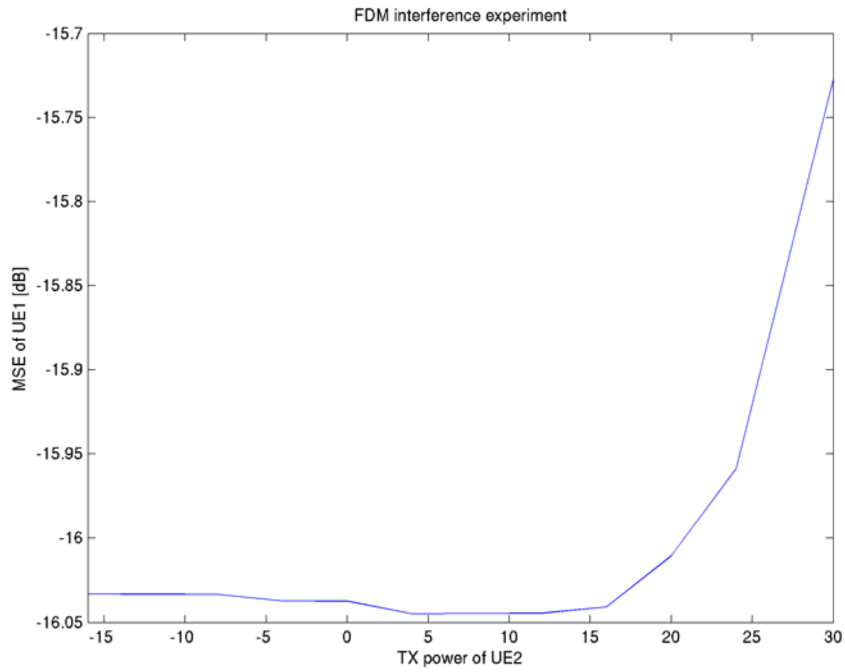

Figure 8: Plot showing variations in MSE with varied signal power of one FDM user.

As shown in Figure 8 the interference between users sounding on different frequency combs is notable only for extreme cases of different signal powers and is still very small compared to that of code multiplexed users, thus the interference between FDM users may not necessarily have to be taken into account when modeling the estimation error.

5.2.3 Inter-cell interference

The experiments presented in this section are performed to illustrate the effects of inter-cell interference. The MSE of one user is studied while the signal power of an interfering user in a neighboring cell is varied. Two separate experiments were performed with different sounding bandwidths to illustrate how the sounding bandwidth affects the results.

Since it was concluded in 5.2.2 that the interference between users on different transmission combs may be neglected, it is assumed that the same is valid even for users in different cells. Therefore, only users transmitting on the same transmission comb are considered ion this chapter.

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Inter-cell interference experiment 1 setup	
Channel type	etu5
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	500
Number of UEs	2
Cyclic shift of UE1	0
Cyclic shift of UE2	0
Transmission comb of UE1	0
Transmission comb of UE2	0
Signal TX power UE1 [dB]	16
Signal TX power UE2 [dB]	[-16.00, -12.00, -8.00, -4.00, 0.00, 4.00, 8.00, 12.00,16.00, 20.00, 24.00, 30.00]
Base sequence of UE1	0
Base sequence of UE2	1

Table 5

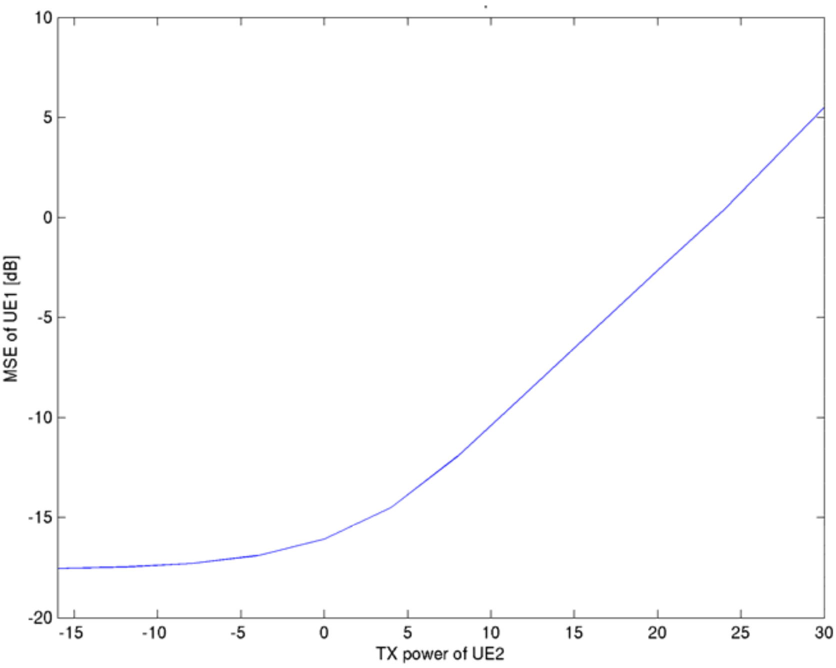

Figure 9: Intercell experiment 1, this figure shows variations in MSE due to one interfering user in a neighboring cell. User 1 has constant transmission power.

Inter-cell interference experiment 2 setup	
Channel type	etu5
Sounding bandwidth	16 PRBs
Simulation duration (Number of sounding instances)	500
Number of UEs	2
Cyclic shift of UE1	0
Cyclic shift of UE2	0
Transmission comb of UE1	0
Transmission comb of UE2	0
Signal TX power UE1 [dB]	16
Signal TX power UE2 [dB]	[-16.00, -12.00, -8.00, -4.00, 0.00, 4.00, 8.00, 12.00,16.00, 20.00, 24.00, 30.00]
Base sequence of UE1	0
Base sequence of UE2	1

Table 6

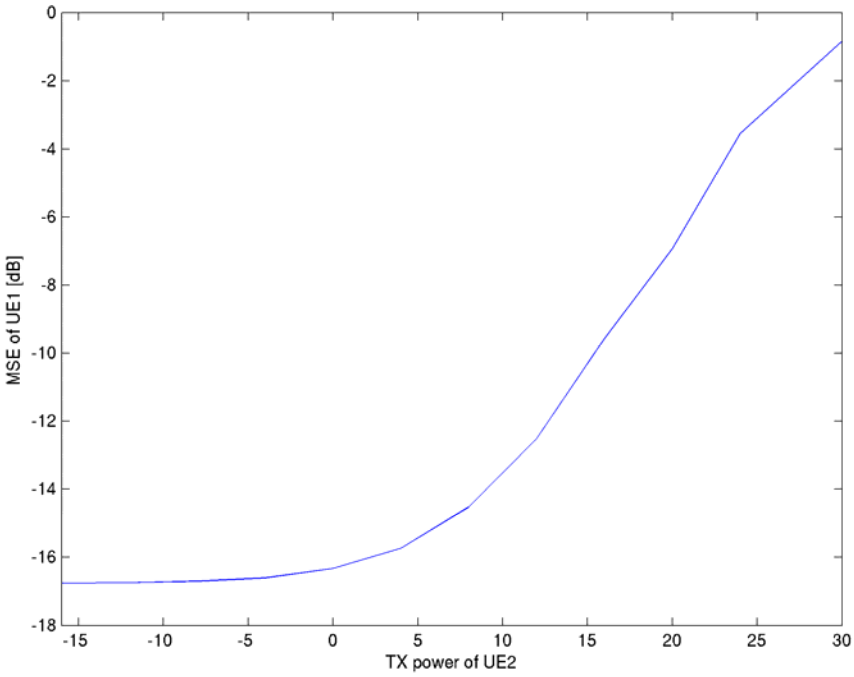

Figure 10: Intercell interference experiment 2, similar setup to that of intercell interference experiment 1, but with different sounding bandwidth.

As the link simulator only simulates one cell, the behavior of having a user in a neighboring cell was recreated by assigning UE2 a different Zadoff-Chu base sequence.

The interference from sounding users in other cells seems to affect the sounding channel estimate in a fashion very similar to that of users in the same cell. There are, however, a few notable differences.

The cyclic shift of users in neighboring cells will not affect the orthogonality as different cells use different Zadoff-Chu base sequences. Instead, as the cross-correlation of Zadoff-Chu sequences with different base sequences vary with the length of the sequence, the impact of the inter-cell interference will depend on the sounding bandwidth.

Comparing Figure 9 and Figure 10 one can see that the impact of the intercell interference is approximately 6 dB smaller when using a sounding bandwidth of 16 PRBs rather than 4. This relates well to what can be expected according to Equation 4 as shown in Equation 16.

$$20\log_{10}\left(\frac{1}{\sqrt{N_{ZC1}}} \Big/ \frac{1}{\sqrt{N_{ZC2}}} \right) = 20\log_{10}\left(\sqrt{\frac{89}{23}} \right) \approx 6\ dB$$

Equation 16

5.2.4 UE velocity / Doppler shift

To evaluate whether the movement speed of a user is relevant for modeling the estimation error, simulations were run where the movement speed of 4 CDM users was increased while studying the resulting MSE.

UE velocity experiment setup	
Channel type	ETU
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	1000
Number of UEs	4
Cyclic shift of UE1	0
Cyclic shift of UE2	2
Cyclic shift of UE3	4
Cyclic shift of UE4	6
Signal TX power UE1 [dB]	16
Signal TX power UE2 [dB]	16
Signal TX power UE3 [dB]	16
Signal TX power UE4 [dB]	16
Doppler shift of all UEs [Hz]	[5, 10, 20, 50, 100, 150, 200, 250, 300]

Table 7

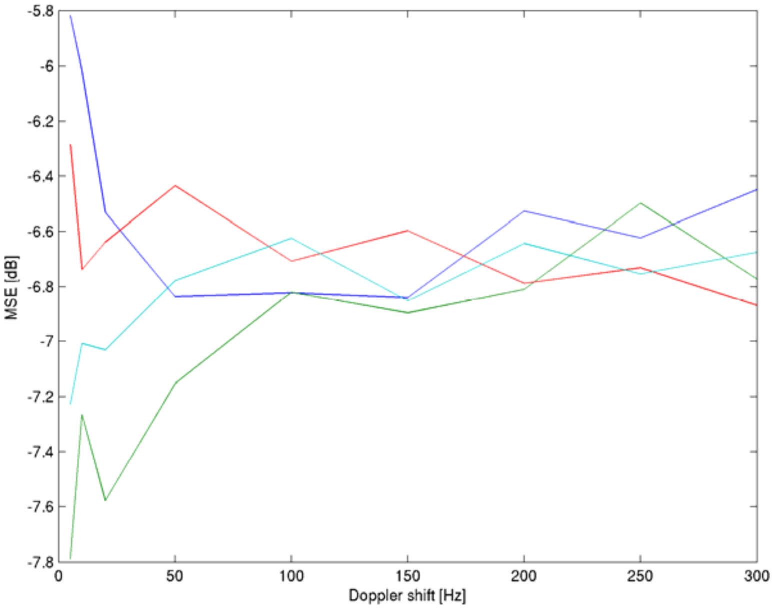

Figure 11: MSE for 4 CDM UEs with increasing Doppler shift

As Figure 11 implies, the quality of the resulting SRS channel estimate does not seem to decrease noticeably when the movement speed of the UE is increased. High speed will however still affect the usefulness of the resulting channel estimate as the estimate will, even with a high sounding periodicity, quickly grow obsolete with a rapidly changing channel.

Figure 11 shows variations in MSE for four code multiplexed when the movement speed of the users is varied. It can be concluded that the effect on the instantaneous channel estimate is very small, if any at all. The actual parameter being iterated over is actually Doppler shift, however, as the Doppler shift depends on the movement speed of the UE, increasing the Doppler shift has the exact same effect as increasing the movement speed.

The large variations of the MSE on the slow moving channels, to the left side in the figure, can be attributed to the slower channels not giving as representative simulation results as the fast one with an equally long simulation time. This occurs as the probability of one user having a very strong or very weak fading channel throughout the entire simulation is increased on a slow moving channel.

5.2.5 Low frequency resolution

With a channel estimator estimating the channel as a FIR (Finite Impulse Response) channel there is a risk of underestimating the number of filter taps needed to properly describe the channel behavior. A too short impulse response estimate will thus lead to a decreased ability to describe the behavior of a very frequency selective channel. Because of this one can see a low pass effect in the frequency domain of the channel estimate.

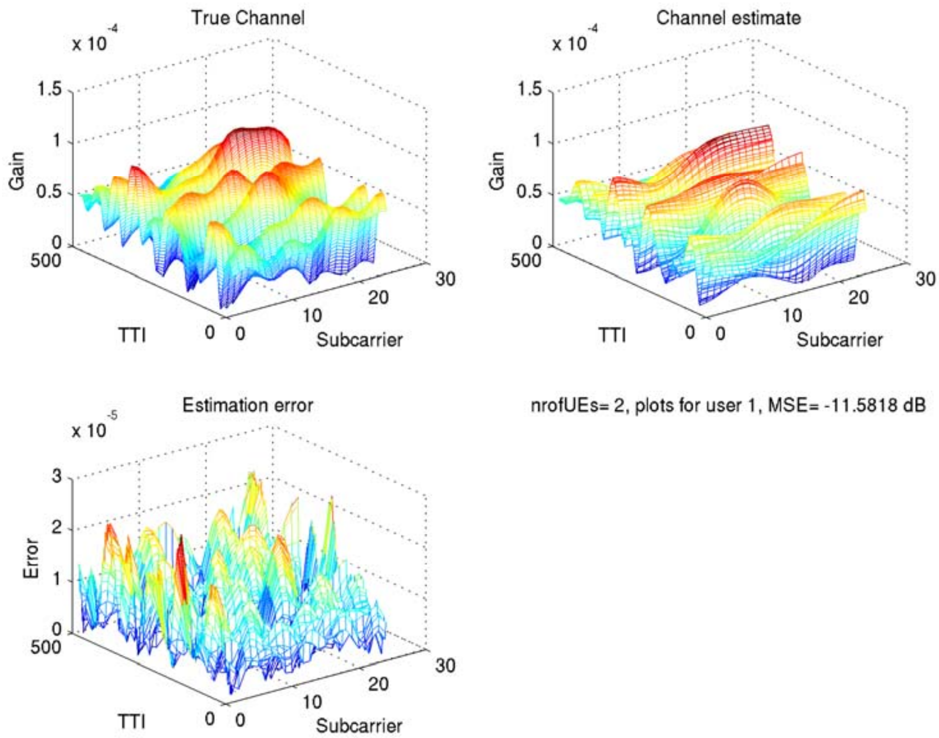

Figure 12: The above figure shows loss of frequency resolution in channel estimate

As the sounding channel estimates are used by the scheduler with a per-resource-block resolution, the effects of this issue are decreased by averaging the channel over each resource block. The effect is still highly significant for the purpose of modeling the estimation error. Figure 11 shows the true channel, the estimator out put as well as the estimation error before the per-resource-block averaging. Figure 12 shows the same plots but after the averaging is done.

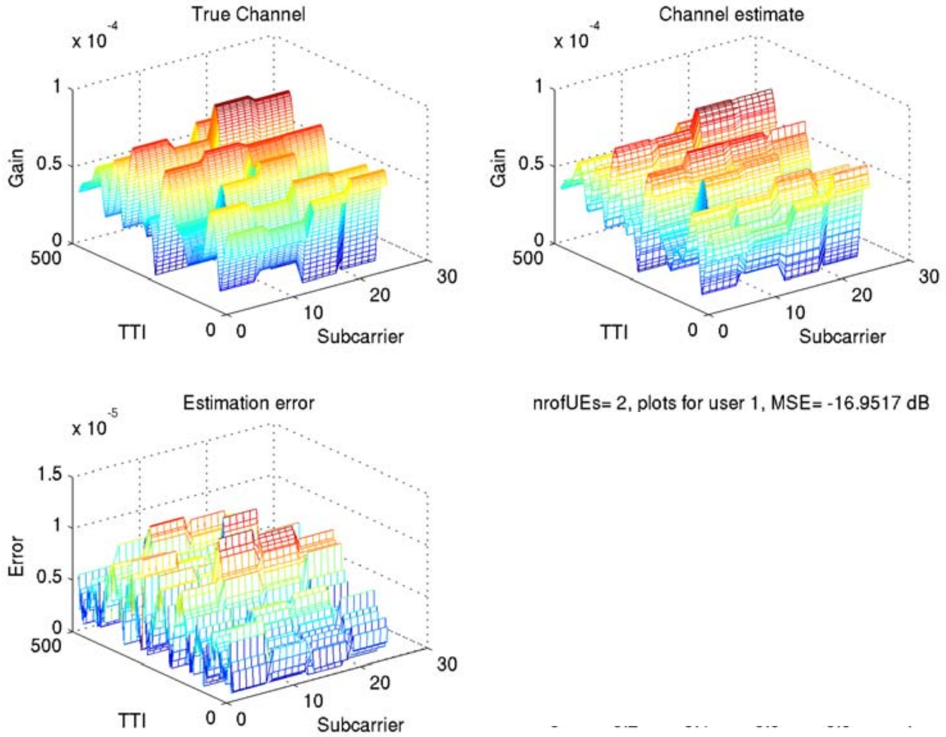

Figure 13: The same channel as in the previous figure but with averaging over each PRB

This averaging per resource block also seems to slightly decorrelate the error in the frequency domain. This is interesting for modeling purposes as it means the estimation error may more easily be modeled as additive white noise.

5.2.6 Time dependency

Error time dependency experiment setup	
Channel type	etu5
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	500
Number of UEs	4
Cyclic shift of UE1	0
Cyclic shift of UE2	2
Cyclic shift of UE3	4
Cyclic shift of UE4	6
Signal TX power UE1 [dB]	30
Signal TX power UE2 [dB]	30
Signal TX power UE1 [dB]	30
Signal TX power UE2 [dB]	30

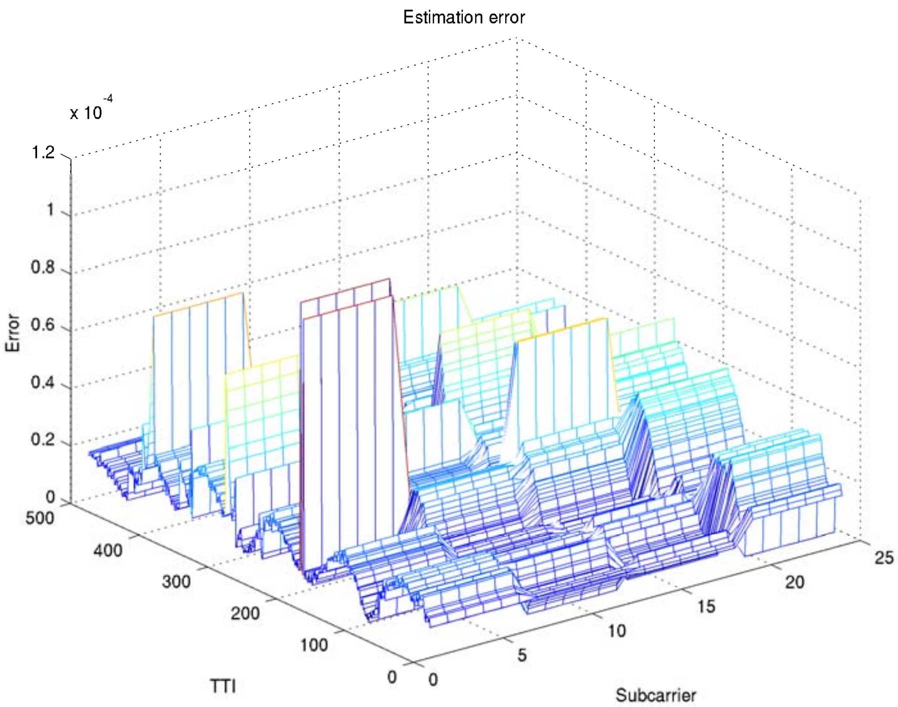

Figure 14: This figure shows the estimation error in a high interference scenario

It can be seen in Figure 14 that the estimation error does display some time correlational properties in a scenario where interference from other sounding UEs is the main source of error.

As the error contribution from interfering users may correlate between sounding instances, additive noise white may prove insufficient to completely recreate all aspects of the behavior of the estimation errors.

5.2.7 Conclusions regarding estimation error behavior

The model should mainly be able to recreate the estimation error contribution due to interference from other users, both in the same and in other cells. To properly be able to do this the model must take into account different phenomena that arise due to the means by which the users are code multiplexed.

The effects from lost frequency resolution due to the time dispersion of the channel being too long to be properly described with the available number of filter taps is also a factor important to include in the model.

Decreases in estimate quality due to user movement speed and interference from users sounding on the other frequency comb will be neglected as their effects are very small compared to those previously mentioned.

As many of the characteristics of the estimation error are strongly depending on time dispersion, the model shall be trained with different sets of parameters for different channel types.

The project was started with an initial hypothesis of overhearing between sounding users. This overhearing should occur due to lost orthogonality between sounding UEs and should thus be much more significant between UEs with neighboring cyclic shifts in their SRS sequences. There should thus exist a correlation between the signal strength of the received SRS signal, and thus also the channel estimate, of a strong sounding user and the estimation error of a user with a neighboring cyclic shift and a weak signal.

However, as the existence of such correlation has not been completely verified. The received signal power of a user with a close cyclic shift does strongly affect the size of the estimation error but it does not necessarily seem to create a deterministic error similar to the received interfering signal, and a statistical model is therefore chosen to describe the error rather than a deterministic one.

Initially, the estimated estimation error shall be added to the known ideal channel in the system simulator as white noise. Some time filtering to this added noise may be used to further mimic the behavior of the actual estimation error. This will, however, greatly increase the complexity of implementing the model, as sounding periodicity and user movement speed may differ and different amounts of filtering will then be needed.

5.3 Model

This model description is made using a general MMSE to derive the error sources in the channel estimate. The MMSE estimator is described more closely in chapter 4.6. The model is described for N users sounding the same bandwidth and on the same frequency comb. All the users do not necessarily need to be in the same cell.

It is assumed that the received signal can be described on the form described by Equation 17.

$$\mathbf{y} = \mathbf{h}_1 \cdot \mathbf{x}_1 + \sum_{i=2}^{N} \mathbf{h}_i \cdot \mathbf{x}_i + \mathbf{w}$$

Equation 17

Where $\mathbf{y}$ is the received signal, $\mathbf{x}_1$ is the transmitted signal of the UE whose channel is to be estimated and the sum of $\mathbf{x}_i$ are the transmitted signals from N-1 interfering sounding users. $\mathbf{h}_1$ and $\mathbf{h}_i$ are their respective fading channels and $\mathbf{w}$ is additive noise. $\mathbf{h}_1$ is then the channel to be estimated.

It has been concluded in 5.2.5 that the time dispersion of the channel may be larger than the available number of filter taps. This leads to that $\mathbf{h}_1$ may be rewritten as shown in Equation 18.

$$\mathbf{h}_1 = \mathbf{h}_1' + \mathbf{h}_1''$$

Equation 18

Where $\mathbf{h}_1'$ is the part of the channel impulse response which can be described with the available number of filter taps and $\mathbf{h}_1''$ is the part which will be truncated. This will yield the following expression for the received signal from which the channel will be estimated.

$$\mathbf{y} = (\mathbf{h}_1' + \mathbf{h}_1'') \cdot \mathbf{x}_1 + \sum_{i=2}^{N} \mathbf{h}_i \cdot \mathbf{x}_i + \mathbf{w}$$

Equation 19

In the same way that was described in 4.6, a matched filter is created by multiplying with $\mathbf{x}_1^{\mathbf{T}}$ and the expected values are calculated.

$$E\left[\mathbf{y}\mathbf{x}_1^{\mathbf{T}}\right] = E\left[(\mathbf{h}_1' + \mathbf{h}_1'') \cdot \mathbf{x}_1\mathbf{x}_1^{\mathbf{T}} + \sum_{i=2}^{N}(\mathbf{h}_i \cdot \mathbf{x}_i)\mathbf{x}_1^{\mathbf{T}} + \mathbf{w}\mathbf{x}_1^{\mathbf{T}}\right] =$$

$$(\mathbf{h}_1' + \mathbf{h}_1'') \cdot E\left[\mathbf{x}_1\mathbf{x}_1^{\mathbf{T}}\right] + \sum_{i=2}^{N} E\left[(\mathbf{h}_i \cdot \mathbf{x}_i)\mathbf{x}_1^{\mathbf{T}}\right] + E\left[\mathbf{w}\mathbf{x}_1^{\mathbf{T}}\right]$$

Equation 20

Exploiting the fact that $E\left[\mathbf{x}_1\mathbf{x}_1^{\mathbf{T}}\right]$ is by definition equal to the identity matrix, **I**, due to the correlation properties of the Zadoff-Chu sequences, the optimal channel estimate will be given by $E\left[\mathbf{y}\mathbf{x}_1^{\mathbf{T}}\right]$.

$$\hat{\mathbf{h}}_1' = \mathbf{h}_1' + \mathbf{h}_1'' + \sum_{i=2}^{N} E\left[(\mathbf{h}_i \cdot \mathbf{x}_i)\mathbf{x}_1^{\mathbf{T}}\right] + E\left[\mathbf{w}\mathbf{x}_1^{\mathbf{T}}\right] = \mathbf{h}_1' + \varepsilon$$

Equation 21

Where ε is the channel estimation error which may then be described as done in Equation 22.

$$\varepsilon = \mathbf{h}_1'' + \sum_{i=2}^{N} E\left[(\mathbf{h}_i \cdot \mathbf{x}_i)\mathbf{x}_1^{\mathbf{T}}\right] + E\left[\mathbf{w}\mathbf{x}_1^{\mathbf{T}}\right]$$

Equation 22

The first term in the above expression is thus the error contribution that comes from truncating the impulse response. The second term should be equal to zero, but, as stated in previous parts of this report, the orthogonality may not always be maintained and this term thus represents the contributing error from the interference of code multiplexed users. The third term should also theoretically be equal to zero as the background noise is generally assumed to not correlate with transmitted signals. However, in real life, and especially for shorter sequences, it is not necessarily a valid assumption that the background noise will in no way affect the size of the estimation error, and a third term will therefore still be considered for the final model.

As the channel estimator is unaware of the actual power of the transmitted signals, the terms describing the channel gain above are actually proportionally scaled by the amplitude of the transmitted signals. This is described more closely in 4.6 and specifically in Equation 13. This will lead to the term $\mathbf{h}_1''$ being proportional to the received signal from user 1 as well as each term in the sum of interfering users being proportional to the received signal stemming from each user 2 through N.

Assuming that the noise may not be completely orthogonal to the used signals, and is thus not negligible in the resulting channel estimate, and that the error in the estimate can be sufficiently described as a zero mean additive noise process, the standard deviation or root mean square error for user 1 may be described as done in Equation 23.

$$\hat{\sigma}_1 = \alpha \cdot |h_1| + \sum_{i=2}^{N} \beta_i |h_i| + \gamma$$

Equation 23

The factor α will then correspond to the error contribution due to the truncating of the impulse response. The factors β_i will correspond to the unorthogonality factors mentioned in 5.2.1 and 5.2.3 and thus relate to the error contribution from interfering sounding users. The term γ corresponds to the part of the estimation error that occurs due to background noise.

The above expression does suffice to describe interference from both users in the same cell and users in neighboring cells as the interference is added in the same way but with different values of the unorthogonality factors, β_i.

5.4 Model training

Since the values of the different parameters depend strongly on time dispersion, different sets of parameters will be trained for different channel types. These channel types are EPA (Extended Pedestrian A) and the more time dispersive ETU (Extended Typical Urban) specified in [7] as well as an AWGN channel.

The training sequences used have a length of 500 frames, which in this case translates to 2500 sounding instances. This as each frame consists of ten subframes and the SRS periodicity is chosen to 2. The SRS bandwidth used for model training is 4 resource blocks. Because the addition of interference from different users has been shown to be linear, each set of training sequences is generated with only two users. The user whose error is to be studied is given cyclic shift 0. Training sequences are then generated with user 2 having a relative cyclic shift of 1 through 7 as well as one set of sequences with user 2 in an adjacent cell. The setups used for model training are described more closely in Table 8.

Training sequence setup	
Channel type	[etu5, epa5, awgn]
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	2500
Number of UEs	2
Cyclic shift of UE1	0
Cyclic shift of UE2	[1, 2, 3, 4, 5, 6, 7, 0]
Transmission comb of UE1	0
Transmission comb of UE2	0
ZC Base sequence of UE1	0
ZC Base sequence of UE2	[0, 0, 0, 0, 0, 0, 0, 1]
Signal TX power UE1 [dB]	[-16.00, 0.00, 16.00, 30.00,-16.00, 0.00, 16.00, 30.00,-16.00, 0.00, 16.00, 30.00,-16.00, 0.00, 16.00, 30.00]
Signal TX power UE2 [dB]	[-16.00,-16.00,-16.00,-16.00, 0.00,0.00,0.00,0.00, 16.00,16.00,16.00,16.00, 30.00, 30.00, 30.00, 30.00]

Table 8

To train the model parameters, the error, modeled as described in the previous chapter, must be written on matrix form. This results in the following equation.

$$\mathbf{E} = \mathbf{S} \cdot \mathbf{p}$$

Equation 24

For training the model, the training sequences are first described as vectors with the measurements from each resource block after another. These vectors, containing the absolute values of the received signal amplitudes of all users, together with a column of ones form the matrix, **S**. With the chosen training sequence setup this matrix will have dimensions 80 000 x 10. K referred to the equations below is thus equal to 80 000. The matrix **S** is shown in Equation 25.

$$\mathbf{S} = \begin{bmatrix} S_1(0) & \dots & S_8(0) & S_9(0) & 1 \\ S_1(1) & \dots & S_8(1) & S_9(1) & . \\ . & . & . & . & . \\ S_1(K-1) & \dots & S_8(K-1) & S_9(K-1) & 1 \end{bmatrix}$$

Equation 25

S_1 is a vector containing absolute values of the received signal of the user whose error will be studied. The vectors S_2 to S_8 are vectors containing absolute values of the signals of code multiplexed users in the same cell with relative cyclic shifts 1 through 7. S_9 contains the signal of one user sounding in a neighboring cell, with a different Zadoff-Chu base sequence. The last column of ones corresponds to the effect of thermal noise on the channel.

The vector, **E**, containing the absolute values of the error of user 1 is created as the right hand side of the initial equation.

$$\mathbf{E} = \begin{bmatrix} E(0) & \dots & E(K-1) \end{bmatrix}^{\mathbf{T}}$$

Equation 26

The parameter vector, **p**, consists of the parameters that are to be trained.

$$\mathbf{p} = \begin{bmatrix} \alpha & \beta_1 & \beta_2 & \beta_3 & \beta_4 & \beta_5 & \beta_6 & \beta_7 & \delta & \gamma \end{bmatrix}^{\mathbf{T}}$$

Equation 27

The elements in this vector are, α , β_1 to β_7 , δ and γ . α corresponds to the error contribution due to truncating the impulse response. β_1 to β_7 correspond to error contribution from code multiplexed users with relative cyclic shift one through seven. δ corresponds to error due to interference from neighboring cells and γ is used to take into account the effect of background noise.

The parameter values may now be calculated by solving the resulting overdetermined system of equations as described by Equation 28.

$$\mathbf{p} = (\mathbf{S}^{\mathbf{T}}\mathbf{S})^{-1}\,\mathbf{S}^{\mathbf{T}}\mathbf{E}$$

Equation 28

5.4.1 Inter-cell interference parameter adjustment

Due to the cross-correlation of Zadoff-Chu sequences generated from different base sequences shown in Equation 4 in section 4.2, the size of the inter-cell interference will also depend on which sounding bandwidth is used.

As the effect of the inter-cell interference will depend on the sounding bandwidth, the parameter or weighting factor corresponding is made independent of the bandwidth by multiplying with the square root of the sequence length, also in accordance with Equation 4.

The sequence length is calculated as the largest prime relative to the number of subcarriers used. For instance, sounding on 4 resource blocks will thus yield a sounding base sequence length of 23 as 6 subcarriers are used for each resource block.

When the the error model is applied, the sequence length effects will then be applied by dividing the bandwidth-independent parameter value with the square root of the sequence length currently used.

5.5 Model implementation

The modeled estimation error will be calculated separately for each resource block. The received signals, at the receiver antenna in question, of all simultaneous sounding users, on the same frequency comb, are gathered. The absolute values of these received signals are then weighted together, as described in previous chapters, and the modeled standard deviation of the channel estimate is calculated as done in Equation 29. Where h_n is the true uplink channel between UE n and the base station receiver in question. User 0 is the user whose estimation error is to be calculated, users 1 to N are interfering users in the same cell and users N+1 to M are interfering users in other cells. It should be noted that all the terms labeled h_i are scaled as stated in Equation 13.

$$\hat{\sigma}_0 = \alpha \cdot |h_0| + \sum_{i=1}^{N} \beta_i |h_i| + \frac{1}{\sqrt{N_{ZC}}} \delta \sum_{i=N+1}^{M} |h_i| + \gamma$$

Equation 29

This calculated estimated error is then applied to the known true channel by means of a random number draw from a zero mean Gaussian distributed process with standard deviation $\hat{\sigma}_0$.

$$\hat{\varepsilon} \in N(0, \hat{\sigma}^2)$$

Equation 30

Finally, the “realistic” channel estimate is created by adding the calculated estimation error to the known ideal channel estimate as shown in Equation 31.

$$\hat{h} = h + \hat{\varepsilon}$$

Equation 31

5.6 Necessary assumptions and model limitations

The model is only valid for the case where all users sound the same bandwidth. This is due to the fact that the special correlation properties of the Zadoff-Chu sequences are only valid for sequences of the same length. With different length sequences the orthogonality will not be maintained and the resulting interference will thus be much larger. This is, however, an already known phenomenon and configurations involving different but overlapping bandwidths are thus not likely to occur. This is therefore not consider a major drawback of the current model implementation.

The LTE system is also assumed to be completely synchronized, which is true in a simulator environment but not necessarily in real life implementations. This means that there may be a risk of sounding colliding with data transmissions from other cells, which is a phenomenon not taken into account by the error model.

5.7 Model Performance

In this chapter some experiments verifying the performance of the error model are presented. The resulting plots are shown for signal powers ranging from -5 to +20 dB relative to the noise floor, as this is the range was deemed most relevant for evaluating the model performance. Absolute values of the mean square error are calculated as shown in Equation 14. The decibel values of the MSE are calculated relative to the ideal estimator output according to the expression shown in and Equation 15.

For all performance tests except the one labeled “2UE performance test 2” The actual MSE as well as the modeled one are studied while increasing the transmission power of all UEs. For “2UE performance test 2” the transmission power of UE1 remains the same while the transmission power of the interfering UE is varied.

2UE model performance test 1	
Channel type	etu5
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	100
Number of UEs	2
Cyclic shift of UE1	0
Cyclic shift of UE2	4
Signal TX power UE1 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE2 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]

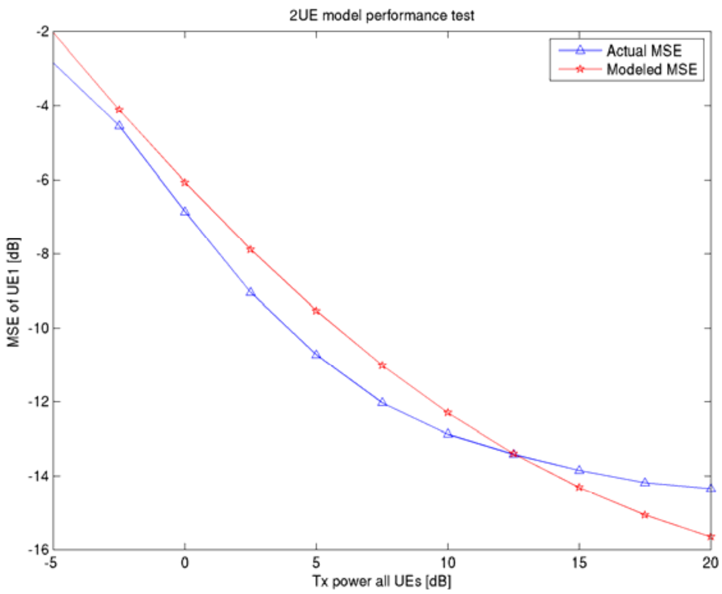

Figure 15: 2UE model performance test 1, relative MSE of channel estimate

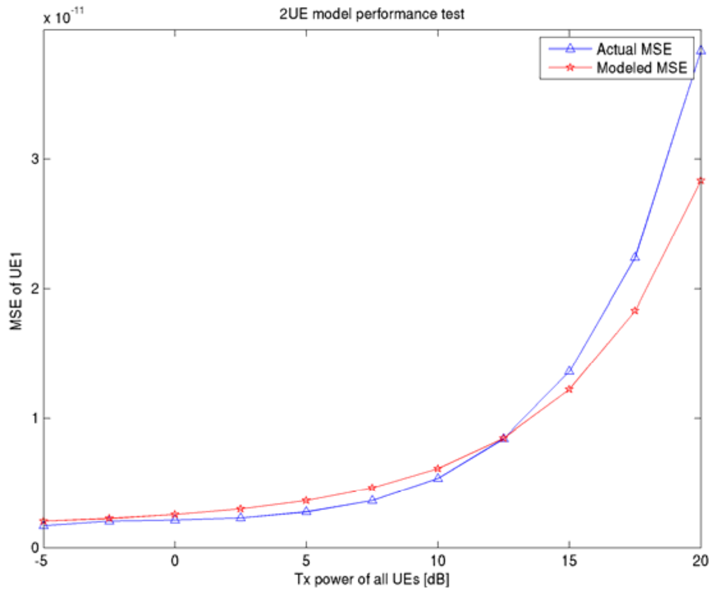

Figure 16: 2UE model performance test 1

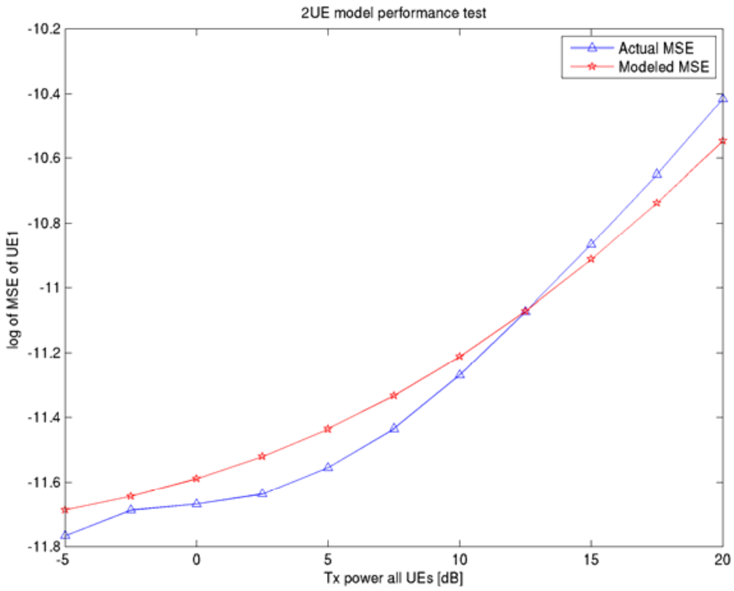

Figure 17:: 2UE model performance test 1

2UE model performance test 2	
Channel type	etu5
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	100
Number of UEs	2
Cyclic shift of UE1	0
Cyclic shift of UE2	4
Signal TX power UE1 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE2 [dB]	16

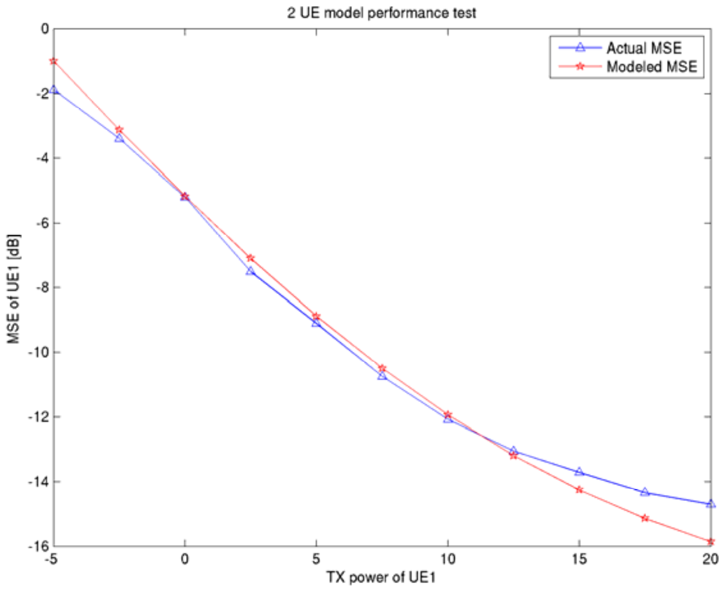

Figure 18: 2UE model performance test 2, relative MSE of channel estimate

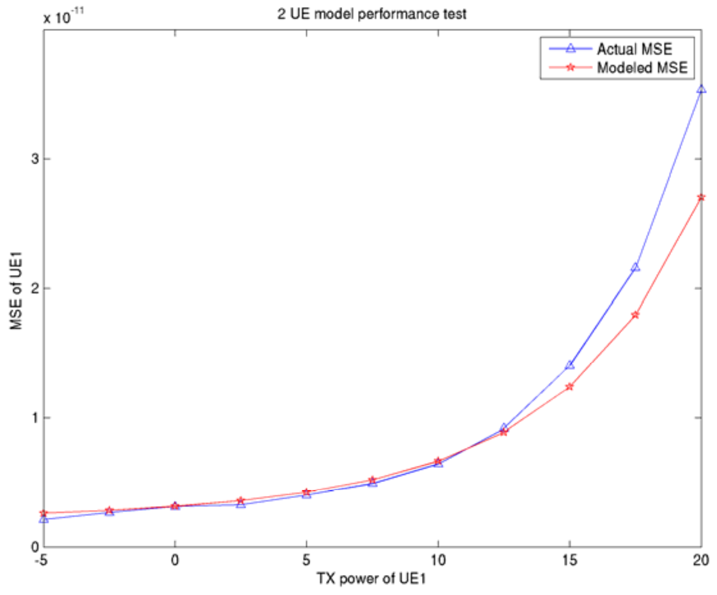

Figure 19: 2UE model performance test 2

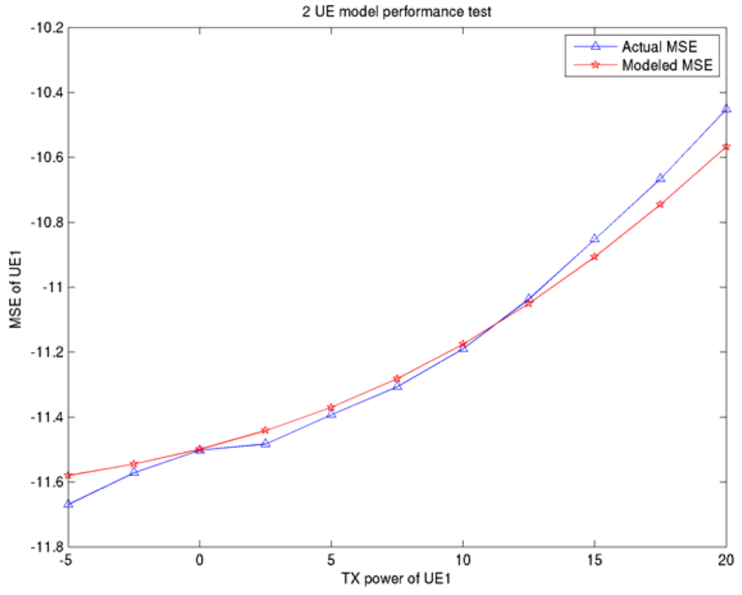

Figure 20: 2UE model performance test 2

Prepared (also subject responsible if other)		No.		
EAB/FJL/SF Pontus Arvidson		EAB/FJL-10:0537 Uen		
Approved	Checked	Date	Rev	Reference
		2010-12-01	A	

4UE model performance test	
Channel type	etu5
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	100
Number of UEs	4
Cyclic shift of UE1	0
Cyclic shift of UE2	2
Cyclic shift of UE3	4
Cyclic shift of UE4	6
Signal TX power UE1 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE2 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE3 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE4 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]

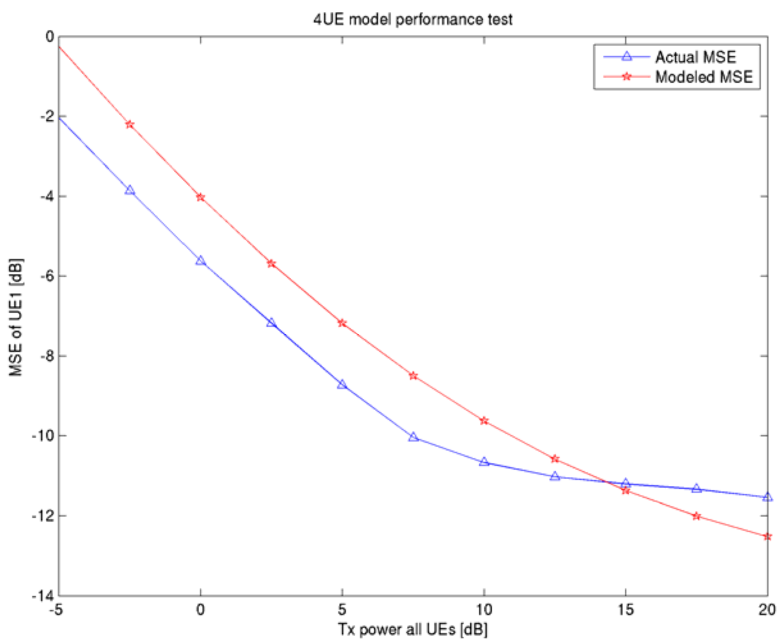

Figure 21: 4UE model performance test, relative MSE of channel estimate

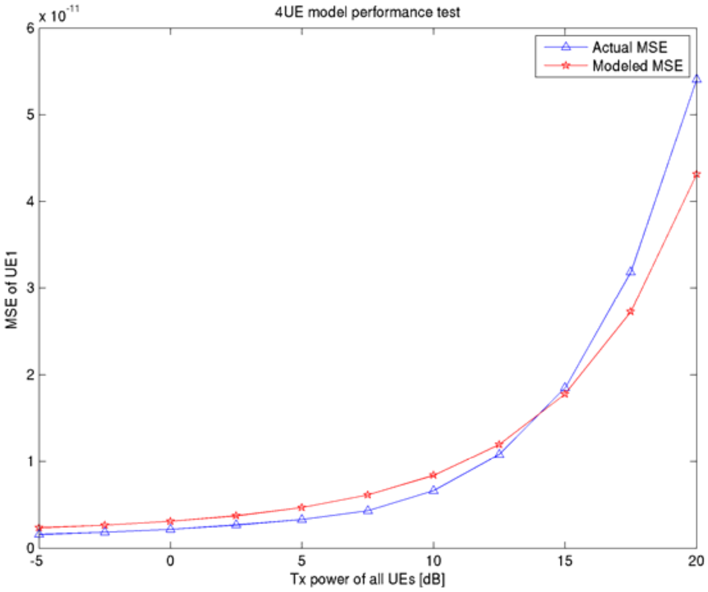

Figure 22: 4UE model performance test

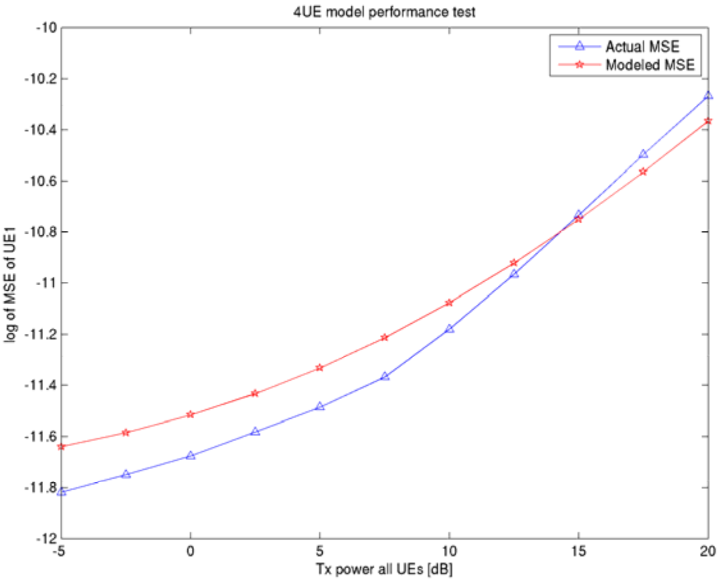

Figure 23: 4UE model performance test

8UE model performance test	
Channel type	etu5
Sounding bandwidth	4 PRBs
Simulation duration (Number of sounding instances)	100
Number of UEs	8
Cyclic shift of UE1	0
Cyclic shift of UE2	2
Cyclic shift of UE3	4
Cyclic shift of UE4	6
Cyclic shift of UE5	1
Cyclic shift of UE6	3
Cyclic shift of UE7	5
Cyclic shift of UE8	7
Signal TX power UE1 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE2 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE3 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE4 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE5 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE6 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE7 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]
Signal TX power UE8 [dB]	[-5, -2.5, 0, 2.5, 5, 7.5, 10, 12.5, 15, 17.5, 20]

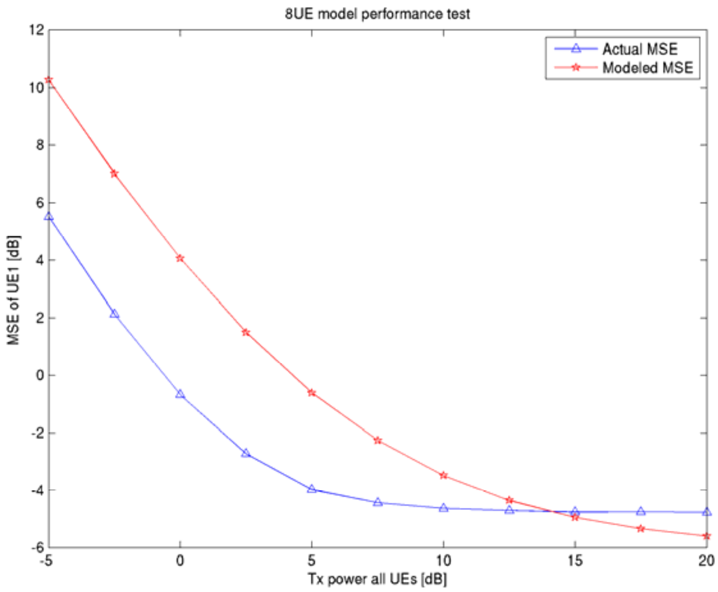

Figure 24: 8UE model performance test, relative MSE of channel estimate

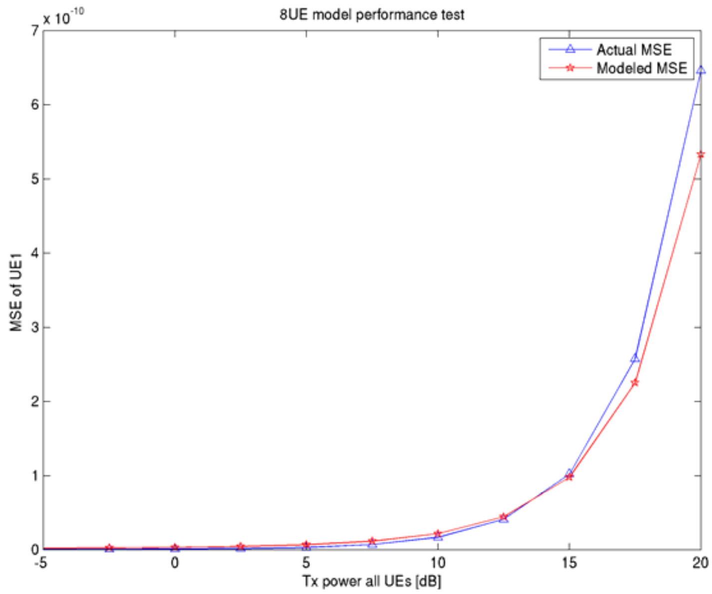

Figure 25: 8UE model performance test

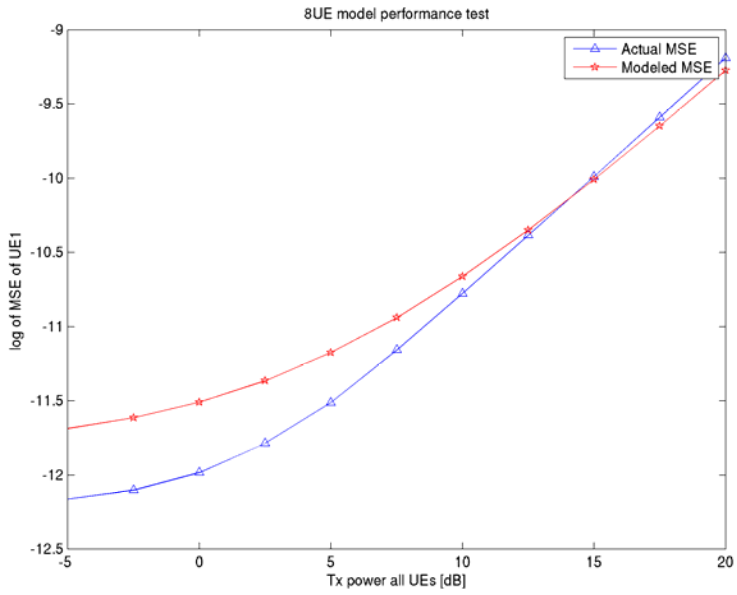

Figure 26: 8UE model performance test

From studying the results in Figure 15 through Figure 26 it may be concluded that the error model does a sufficient job of recreating the behavior of the actual estimation error. The results also imply that some non-linear behavior in the actual estimation exists which cannot be recreated using a linear model.

6 System Simulation Setup

System simulations with error model implemented were run with a “Proportional Fair in Time and Frequency” (PFTF) scheduler. This means that the sounding channel estimates are utilized by the scheduler and hence the quality of these will have a large effect on the uplink throughput.

A primitive sounding resource handler was also created for these simulations. This resource handler exists on a per-cell level and will, as long as free resources are available, give one sounding resource to each user connecting to the cell. The resource handler strives to minimize interference between sounding users by first adding the user to the time offset with fewest other sounding users allocated. Secondly the connecting user will be assigned to the one of the two frequency combs with fewest users on it. Lastly, a cyclic shift is chosen to maximize the relative cyclic shift distance to other already allocated code multiplexed sounding users. Once all resources are allocated, connecting users will not be given a sounding resource and will not transmit sounding. The resource of a user who leaves the cell will be returned and the resource may then be assigned to another user.

The current resource handler also only supports the case where all users transmit sounding over the entire frequency band in use.

6.1 User model

Only one type of user has been used for the system simulations used to test the error model implementation. These users will upon the start of the simulation send a HTTP request to upload a file big enough to not let the users finish uploading before the end of the simulation. This way a steady stream of packets is generated so that the buffer of each UE will almost always have some data to schedule.

The bit rate is calculated using the amount of data received on the TCP level when the simulation is finished. The cell uplink throughput is defined as the sum of the throughput for all users within the cell.

All users are generated at the beginning of the simulation and persist throughout the entire simulation. The users are generated at random starting positions and maintain a movement speed of 3 km/h in a random direction for the whole simulation time. Therefore, it is not guaranteed that all cells have the same identical number of users and that the number of users in each cell remains constant over time.

6.2 Simulation scenarios

All simulations use a radio environment similar to 3GPP case 1 described in [8] with a network that consists of 7 sites with 3 cells each. Wraparound is used to give realistic interference even in cells on the edge of the simulation area. The cell sounding periodicity is set to 1 ms, which means that the last symbol of every other PRB is dedicated for sounding unless otherwise stated, this gives a maximum UE sounding periodicity of 2 ms.

Each simulation is run from 5 different random seeds. The results presented are from combined statistics of all these seeds. The system setup is described in Table 9.

Simulation time	10.5 sec
Deployment	7 sites, 3 cells/site
Bandwidth	20MHz
Environment	3GPP Typical Urban
System	3GPP Case 1

Table 9

Different sounding configurations were used to illustrate how sounding affects the results of the simulations. These configurations are described more closely in Table 10.

Table 10: Sounding configurations

Configuration	SRS periodicity [ms]	Comments
2 ms	2	N/A
5 ms	5	N/A
10 ms	10	N/A
No sounding	N/A	Sounding turned off, allows for 7% more data to be transmitted as reference symbol overhead is smaller when data can be transmitted on symbols otherwise reserved for sounding.
2 ms (Limited resource)	2	Allows only 4 users to be code multiplexed on each frequency comb. Should decrease interference but number of users that may transmit sounding is decreased as well.
2 ms (Ideal)	2	Sounding estimate will always equal the true channel. This setup also includes unlimited sounding resources which means all users are given a sounding resource.

6.2.1 Uplink throughput

These simulations were run with a varying number of uploading users to study the effects on cell throughput with many users and different sounding setups used.

The different sounding setups include different sounding periodicities as well as ideal sounding and no sounding. These setups are described in Table 10.

6.2.2 UE movement speed

These simulations were run with an average of 4 UEs per cell and varying movement speeds. The goal of these simulations is to study if any specific sounding configuration is to prefer over others at a specific UE velocity.

Other parameters are the same as in the Uplink Throughput simulations described in 6.2.1.

7 System Simulation Results

7.1 Uplink Throughput

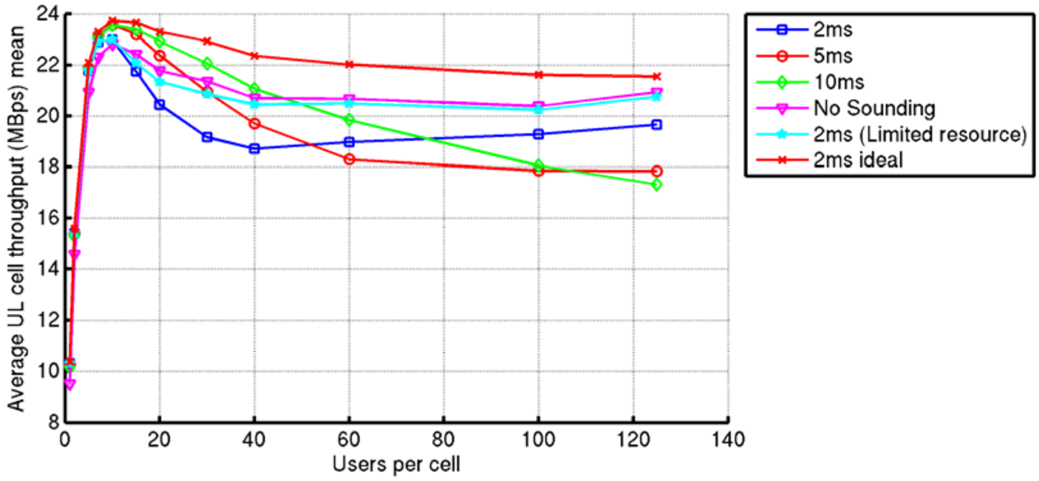

Figure 27: Average cell throughput

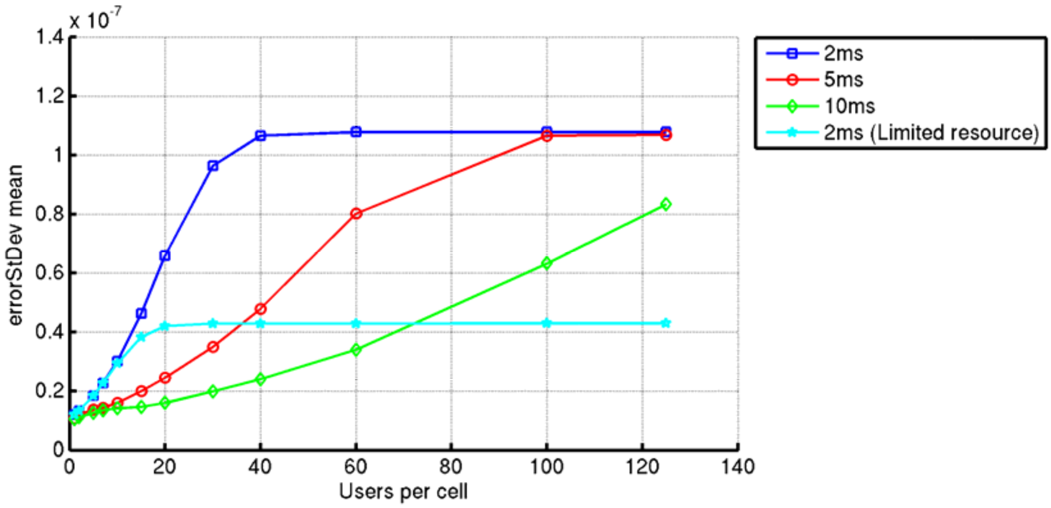

Figure 28: Average standard deviation of modeled estimation error

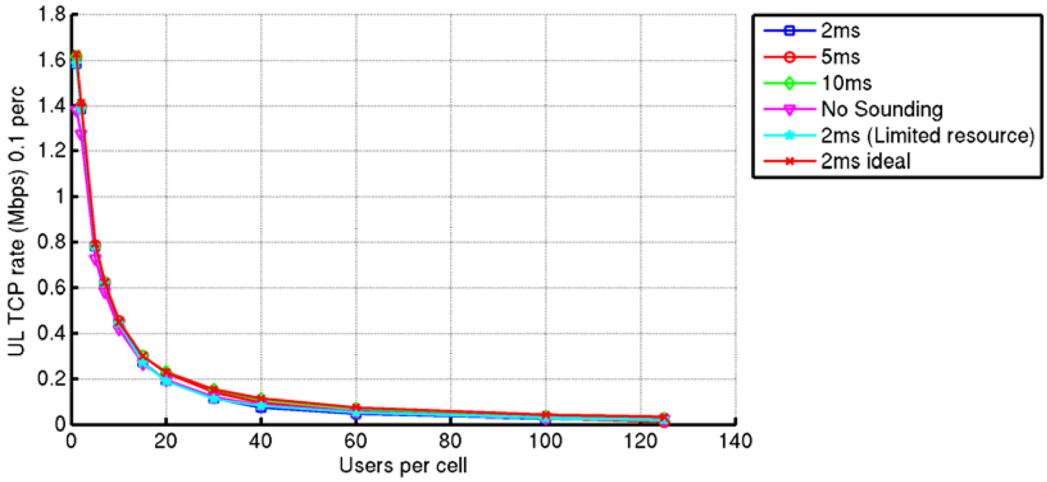

Figure 29: Average throughput for the bottom 10 percentile users

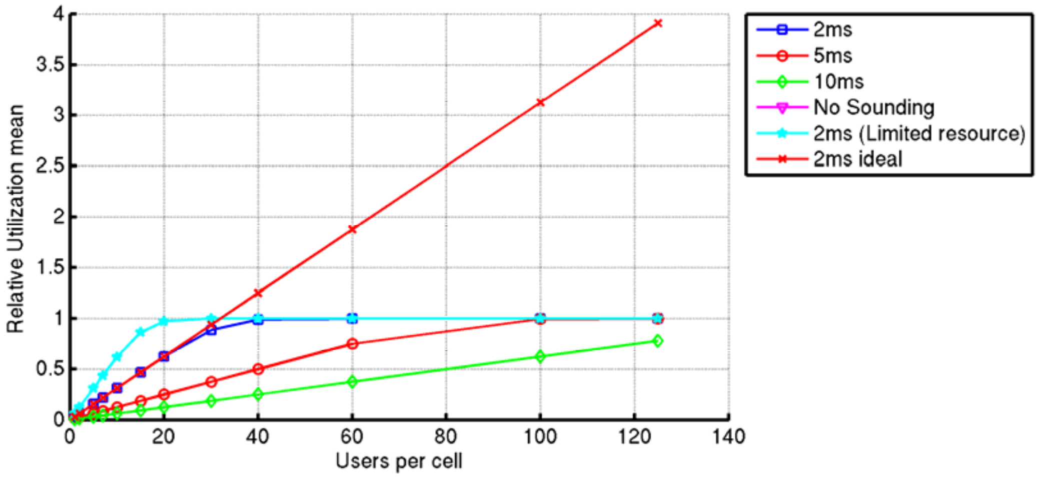

Figure 30: Relative utilization of sounding resources

Comparing Figure 27 and Figure 28 one can conclude that the cell throughput quickly deteriorates when increasing the number of users when using sounding with higher periodicities. This is due to the fact that a higher sounding periodicity also leads to a more limited sounding resource as fewer time offsets are available to divide users on.

Figure 30 shows the relative utilization of the sounding resources, no additional resources are given to new users when this relative utilization reaches 1, except for the ideal sounding case where unlimited resources are available.

One can also see that the worst throughput is reached when the number of users is the same as the total number of available sounding resources. The slight increase in cell throughput, that is achieved when increasing the number of users further, can be attributed to these additional users not being given a sounding resource. The quality of the sounding estimates with full resource utilization is in other words so low that users that to do not get to transmit sounding will achieve higher throughputs when only relying on channel estimates gained from the demodulation reference symbols (DMRS).

It should be noted that these simulations were run to stress the maximum capacity of the sounding resource. Even 20 uploading users per cell is more than the LTE system is optimally designed for.

7.2 UE movement speed

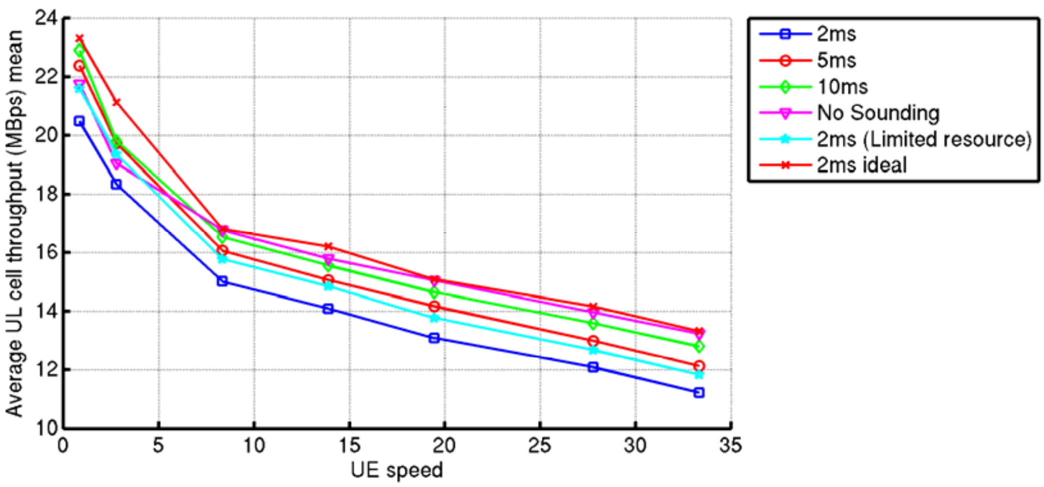

Figure 31: Uplink cell throughput with varying movement speeds

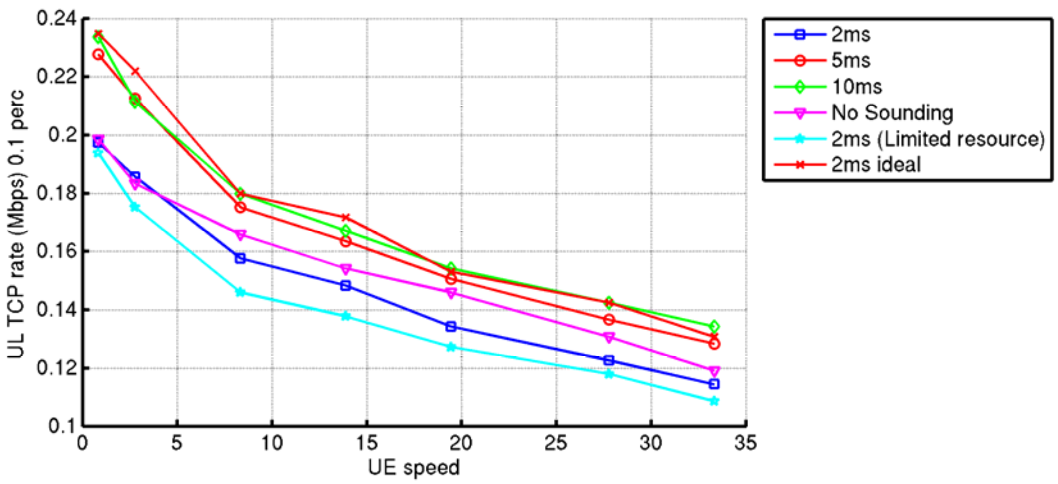

Figure 32: Uplink throughput for the lowest 10 percentile of users.

The simulations run to evaluate the impact of different sounding configurations for UEs with different movement speeds show that choosing a higher periodicity does not noticeably help a fast moving user as the estimate grows obsolete even when using the highest possible periodicity.

8 Discussion

The amount of detail in the system simulator for which the error model is created is much more complex than most other similar simulators. This leads to that there is an apparent lack of previous art when it comes to the modeling of the sounding estimation error. Typically the error is studied in a signal filtered with the channel estimate rather than in the channel estimate itself. This is not the case here since the channel estimate is not used for filtering but for other applications, such as frequency selective scheduling. It is therefore not completely obvious that the estimation error, even when correctly estimated in size, should be applied as added noise to the channel estimate.

Modeling the estimation error as additive noise has some obvious advantages in terms of low complexity when applying the error to the channel. This is because it is easily scaled when having calculated an MSE or a standard deviation. While additive noise does prove sufficient to give a quantitative model of the error one drawback is that some deterministic behavior of the actual estimation error may not be recreated properly. This might not be considered a major flaw on it's own but might encourage faulty algorithms used to better exploit the channel estimates from sounding. An example of this could be to apply some low pass time filtering of the channel estimates when all UEs have low movement speed. This would decrease the impact of the applied error, as it has zero mean, but in a fashion that would probably not be reproducible in a real life implementation. A more deterministic approach may also address many of these issues.

One thing not taken into account is the fact that the actual parameter values may actually vary over time. This is due to the fact that the time dispersion of the channel may also vary over time. One example of this is the factor that describes the error due to truncating the impulse response (α in the equation in chapter 5.5). It is a constant and thus implies that the truncated part of the impulse response always has the same relative size. For the current error model implementation it is therefore assumed that these parameters are properly described by a time average. Model testing implies that this assumption is correct but it is notable that erroneous error estimates of shorter sequences may stem from this phenomenon.

9 Conclusions

The proposed method seems able to sufficiently describe and recreate most aspects of the estimation error.

The modeled estimation error may further be used to give a maximum acceptable size of the sounding estimation error and in turn give an upper bound to which sounding resource utilization should be allowed. This is a powerful tool for dimensioning of the sounding resource for network capacity and different traffic and radio scenarios.

The main limiting factor for the sounding resource seems to be the interference stemming from sounding users in adjacent cells. The interference from same cell users is also not negligible but the code multiplexing serves to suppress this interference a lot. Using intelligent power control algorithms may greatly reduce the impact of the interference from neighboring cells by for instance decreasing the transmission power of cell edge users. This will of course lead to degradation in the channel estimate of the user in question, but this may be outweighed by the user not interfering with the sounding transmissions of other users.

Simulation results clearly show that the sounding resource is a very limited one. In order to fully take advantage of the possible gains from the sounding channel estimates, a more intelligent sounding resource handler is needed. If sounding resources could be given only to the users who may benefit the most from getting high quality channel estimates, the interference between sounding users would decrease a lot since fewer users are transmitting sounding within the same subframe. This would of course, in turn, also lead to higher uplink throughput.

One interesting conclusion that may be drawn is that with high sounding resource allocation the resulting channel estimates may be of such bad quality that they actually decrease the throughput, compared to using no sounding, far beyond the 7% that the sounding resource costs to have.

10 Items for further study

10.1 Modeling of time dependency

In a scenario where the quality of the estimate is mainly affected by interference from other users and the channel conditions change slowly, it is safe to assume that there exists a time correlation in the estimation error. This is due to the fact that the sources of the errors are the same. As the current model implementation adds the estimation error as white noise, this time dependency is not taken into account.

The modeled estimation error might thus further be improved by adding the errors due to interference not as white noise but as an AR-process.

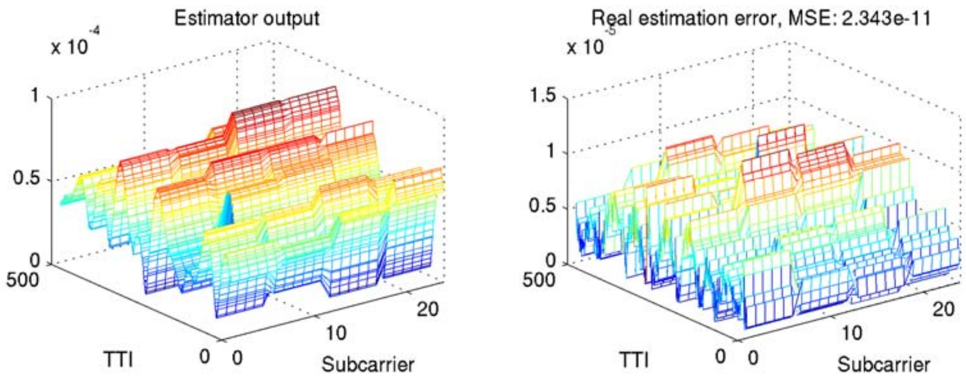

Figure 33: Estimator output and actual error

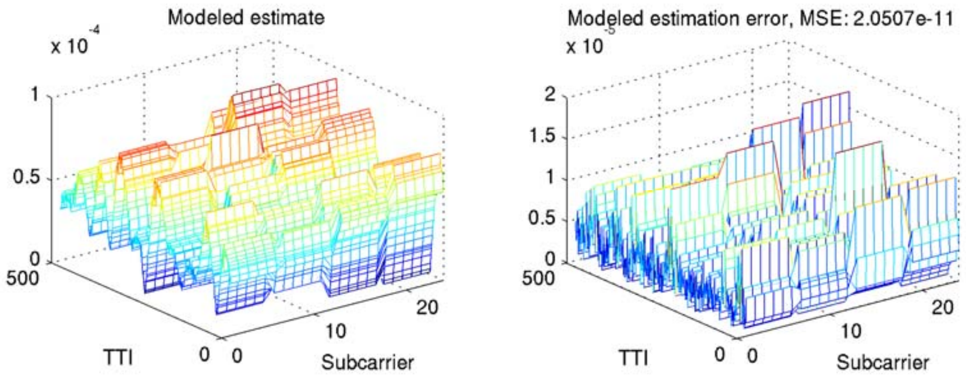

Figure 34: Modeled channel estimate and error with no filtering

Additional experimenting using a 4:th order auto-regressive process (Described in Equation 32) to model the noise caused by interference while still adding the effects of background noise as a white Gaussian process show that it is clearly possible to get the modeled estimation error to behave more like the actual estimation error while maintaining more or less the same MSE. The AR-parameters were trained in a scenario with very strong received signal power from the sounding UEs so that the background noise will have as small an effect as possible on the parameter values.

$$x(n) = ax(n-1) + bx(n-2) + cx(n-3) + dx(n-4) + w(n)$$

Equation 32

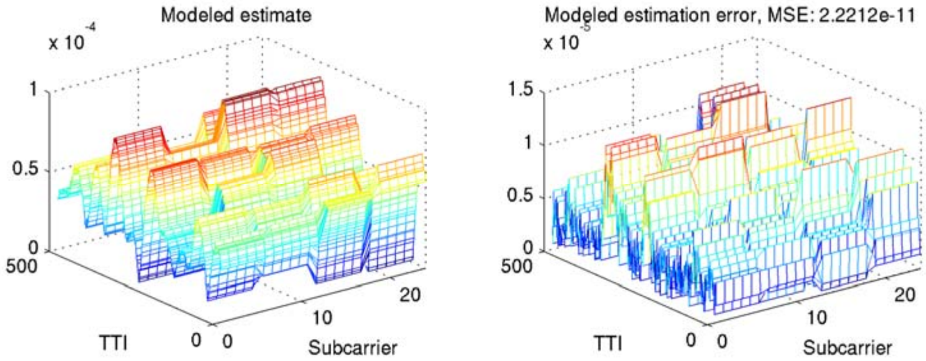

Figure 35: Modeled channel estimate and error with AR-filtering applied

This AR-processing does, however, have a few drawbacks. Not only do separate parameters need to be trained for different combinations of sounding periodicities and UE movement speeds but upon implementing the error model in a simulator environment, every instance needs to be aware of a lot more about the current set up. This may not be desirable from a programming point of view. Although it may be possible to set the AR-parameters as a function of movement speed and sounding periodicity, the complexity of the model is still increased a lot. Some problems with discontinuity may also arise when additional users enter the cell.

Also noteworthy is that since the AR-processing includes infinite impulse response filtering, the error model may give too large variations in the estimated MSE when running shorter simulations. This due to the fact that the tail of the impulse response of the filter is very long and single outliers in the driving random process may thus have a disproportionately large impact on the statistics when simulation times are short. This does, however, not mean that applying additional filtering to the should be completely excluded as a means of further mimicking the behavior of the estimation error.

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