System Level Behavioral Modeling of IEEE 802.11a(OFDM) Standard using Matlab & Simulink

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Abstract - **The aim of the project is to model and simulate the IEEE802.11a standard on system level. The project is designed to be completed in three stages and each stage deals with in more detail than the previous stage. These stages are also called as behavioral modeling because they deals with the modeling of our system in the Simulink and obtain the simulation results. The results clearly show that we achieved fairly high data rates.**

*Index Terms* - **OFDM, QAM, BER, ISI, SNR, LS, FFT, IFFT, STS, MUX, NCBPS, AWGN**
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I. INTRODUCTION

This is a wireless communication system based on IEEE802.11a standard. This system consists of transmitter, receiver and a channel between them. On the transmitter side source coding is performed to compress and channel coding is used for forward error correction [1]. The signal is then modulated to transmit the digital data over an analog channel. On the receiver side the signal is demodulated after considering the synchronization issues. Channel decoding and source decoding is then performed to get the required signal.

II. DESCRIPTION OF THE SYSTEM

WLAN system based on IEEE802.11a standard. OFDM with 64 QAM is used to modulate the signal. Transmission is based on standard protocol over a wireless channel between the transmitter and receiver. For simplicity reasons a wire can be used as channel instead of wireless transmission as the main purpose of system is to design and test the functionality of transmitter and receiver [9]. On the receiving end the received signal is then demodulated to get the originally transmitted signal. Figure below describe the transmitter and receiver of IEEE802.11a.

Figure 1 OFDM Transmitter & Receiver

Transmitter

Receiver
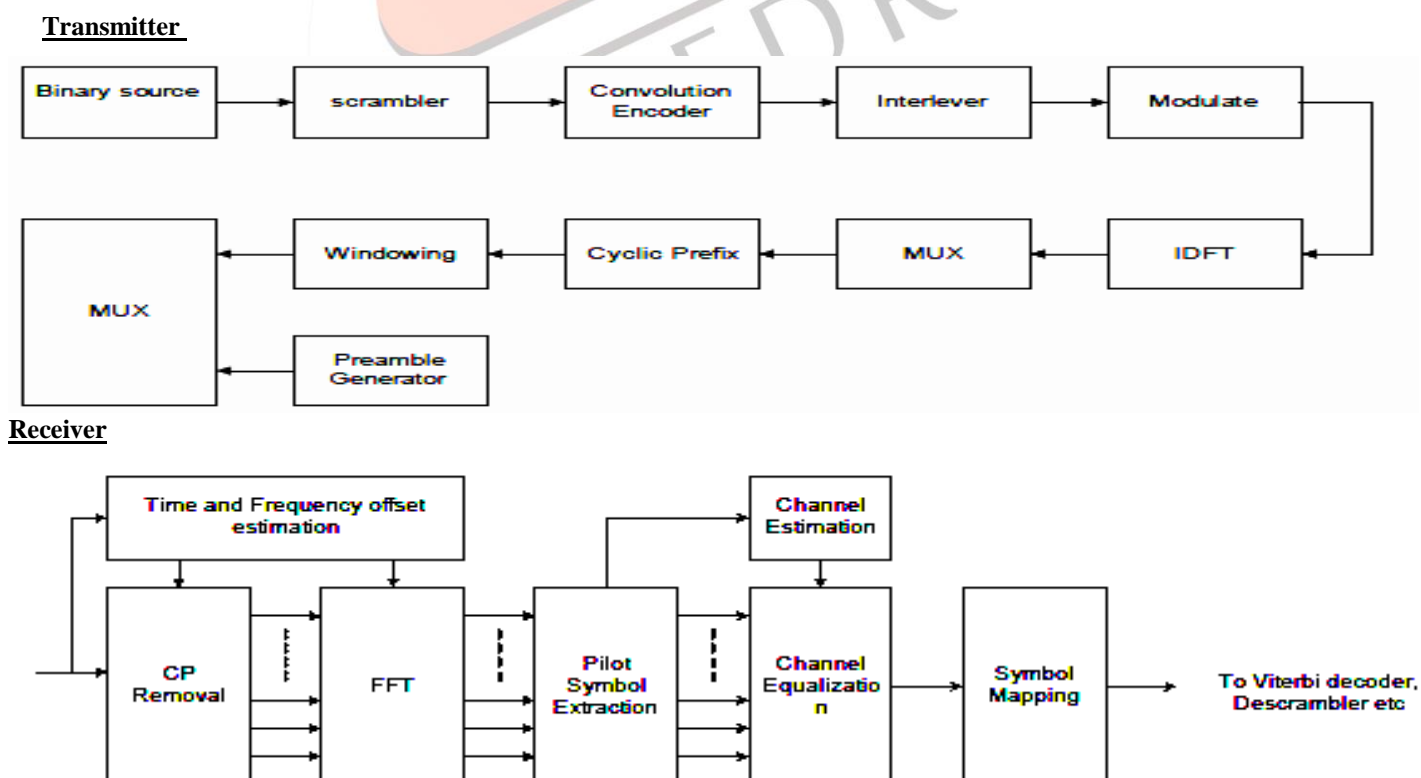
III. MODEL DESCRIPTION

The Variable Data Source is used for generation of 216 samples per OFDM Symbol. To make the transmitted signals less vulnerable to the effects of noise and fading, a channel encoder is used and we have used Convolution encoding in this project.

Convolution Encoder and Viterbi Decoder

Convolution encoding is kind of FEC (Forward Error Correction). The aim of coding is to make sure that in channels with AWGN we can use the best of channel. In this process some extra information is added to the original data. We also used poly2trellis function to create a trellis using the constraint length, code generator and feedback connection. Generally by entering m bit input to the encoder we will have n symbols at the output of the encoder, the proportion of the m/n is called the code rate [2]. There could be different code rates, the coding rate of ¾ is used in this project. To meet the requirement in the standard which is ¾ we have to produce 6 outputs and blind two of them which is called puncturing code. The task for the decoder is to estimate the transmitted code word from a received vector. This will be done using the Viterbi algorithm that has the advantage of having fixed decoding time and it is easier to implement it in hardware design [10]. The Viterbi algorithm calculates the most likely bit pattern by comparing the received vector to all possible patterns. The algorithm reduces the complexity of the calculations by continuously eliminating patterns that aren't optimal.

Interleaver and De-Interleaver

We have used the existing Simulink blocks for interleaving and de-interleaving. The interleaver matrix has 160 rows and NCBPS/160 columns. The interleaver gets NCBPS bits at a time. In our case NCBPS = 2880. So it means the columns=18.

OFDM Symbol Assembly

Coding rate of ¾ is used in convolution encoding that results in 2880 interleaved bits. 64 QAM modulator maps the data into constellations and after buffering we get ten symbols each containing the 48 symbols. The I and Q parts each of data sub carriers are fed into OFDM symbol assembler which inserts pilot, DC and guard carriers to make total of 53 carriers for OFDM realization. An OFDM symbol is made up of three types of basic sub-carriers; 48 Data subcarriers, 4 Pilot subcarriers and a DC components The OFDM assembler is consisting of Pilot Insertion block [3]. These pilot tones are inserted in the original data at locations (6, 19,31, 44),and a zero DC component is inserted at 25th location. To make the Frame length to 64 samples per frame we append zeros in the end of original data. This is done by Pad block placed after OFDM assembler. Internally we have used two major blocks called as Select data block and Assemble subcarriers blocks. The select data block will give the input frames given by QAM to Assemble block in a sequence of (1, 5; 6,18; 19,24; 25,30; 31,44; 45,48).In between these samples the pilot tones and the zero DC components are inserted. We split data into some frames, frame is divided according to the following Matlab matrix format:

{1:5, 6:18, 19:24, 25:30, 31:43, 44:48}. We have some pilot symbols that can be placed between these five length symbol frames. These pilots help the frames to be transmitted to the receiver. OFDM frame is mapped using 64 QAM [4]. After zero padding, IFFT is performed and addition of cyclic prefix, signal is transmitted over channel.

Cyclic Prefix

If two time domain signals are circularly convolved, it should be equivalent to the multiplication of the two DFTs of the time domain signals in the frequency domain. Circular convolution can be achieved in our system using linear convolution by just adding cyclic prefixes in our linear convolution. By using cyclic prefix we fake the circular convolution by just using linear convolution. Cyclic prefix prevent ISI. We appended last 16 samples to the beginning as cyclic prefix [5]. On the receiver side Removal of Cyclic Prefix is performed. Then FFT is taken and Zeros and DC components are removed. Signal is then demodulated and de-interleaved. Error rate is calculated after Viterbi decoding to measure the performance of system. Packet of 960 bits is formed with 2 two long training symbols and 10 frames where each frame consists of 48 data symbols.

STS Insertion and Noise Addition

After the OFDM multiplexer when we get the packet of 960 samples, we added the ten short training sequences (STS) each of 16 bits. Extra noise is added to the data so that at receiver side the STS and noise can be differentiated. We used multipath Rician Fading channel along with AWGN channel to carry out simulations.

Delay and Correlate Algorithm

We use signal auto-correlation, and taking advantage of the repetition, and correlating the received sequence samples with a delayed copy of the sequence, with the delay being equivalent to the length of one symbol. A moving average of this correlation can be taken over a range of one symbol   [11].

$$R(d) = \left| \sum_{m=0}^{L-1} \left(r^*_{d+m} r_{d+m+L} \right) \right|$$

The R(d) represents the value of the dth incoming sample, R(d)* represents the conjugate of R(d), and for STS, L = 16 samples as we cannot detect the packet only with the R(d) alone due to variance of power of incoming signal. We used an improved method in which auto-correlation is normalized by a moving sum of the received power given by following equation:

$$M(d) = \frac{|R(d)|^2}{(P(d))^2}$$

The obtained value is compared with a threshold which is chosen to minimize the false positive detection. If M(d) is greater than threshold, the packet is detected. The input to packet detector is output of sampled channel. The output of packet detection circuit

is control signal indicating the detection of packet. The output is positive (non-zero) if packet detection signal remains non zero for certain number in past clock periods.

Channel Analysis

In a multi-path environment, a transmitted symbol takes different times to reach the receiver by passing through different routes. It introduces time dispersion which makes duration of received symbol wider, so currently received symbol overlaps to the previous received symbols and results in inter-symbol interference (ISI) [6]. Given below is a simulation result using Matlab BER tool for different channels using 64QAM. Channel Equalization plays very important part at the receiver during reception of a signal coming from a multi-path environment/channel. At the output of FFT, the sample at each sub-carrier is multiplied by the coefficient of the corresponding channel equalizer. We have taken a picture of transmitted vs received data constellation after passing through channel equalizer.

FIGURE 2 BER TOOL RESULTS FOR DIFFERENT CHANNELS
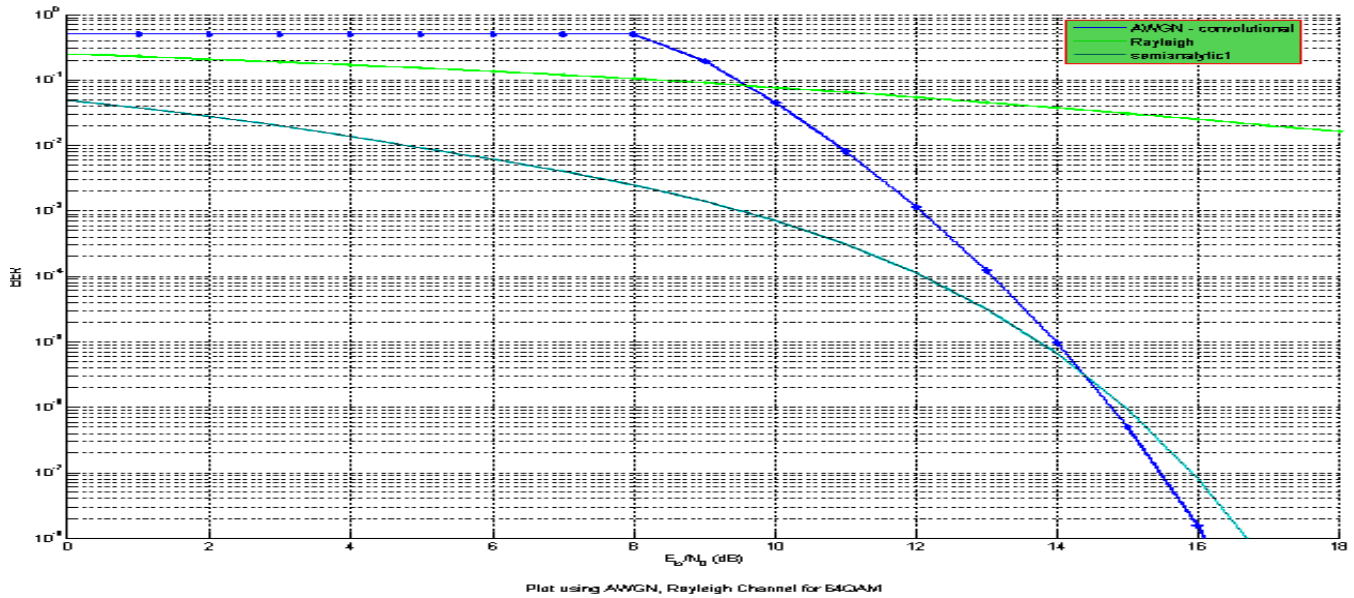
Equalizer Block

Based on training sequence of transceiver, we can get the channel estimation result to achieve frequency domain equalization. Getting rid of the DC component from training sequence firstly, than we utilizes Least Square Estimation. Set the original training sequence b,the received training sequence a, and c=a/b, so the LS estimation result is: H=conjugate(c)/square(abs(c)) [7].

Simulation Results of Behavioral Level Model

The implemented model was tested to verify the functionality. To test the system under real world conditions, the delay, AWGN channel and multipath Rician fading channel were inserted between the transmitter and receiver module.

The Multipath Rician channel introduces the phase distortion. The constellation on right side of figure below is the one which is transmitted and the received data after passing through channel equalizer is shown in the constellation at left hand side in figure below:

FIGURE 3 : TRANSMITTED VS RECEIVED CONSTELLATIONS (SIMULATION RESULTS)
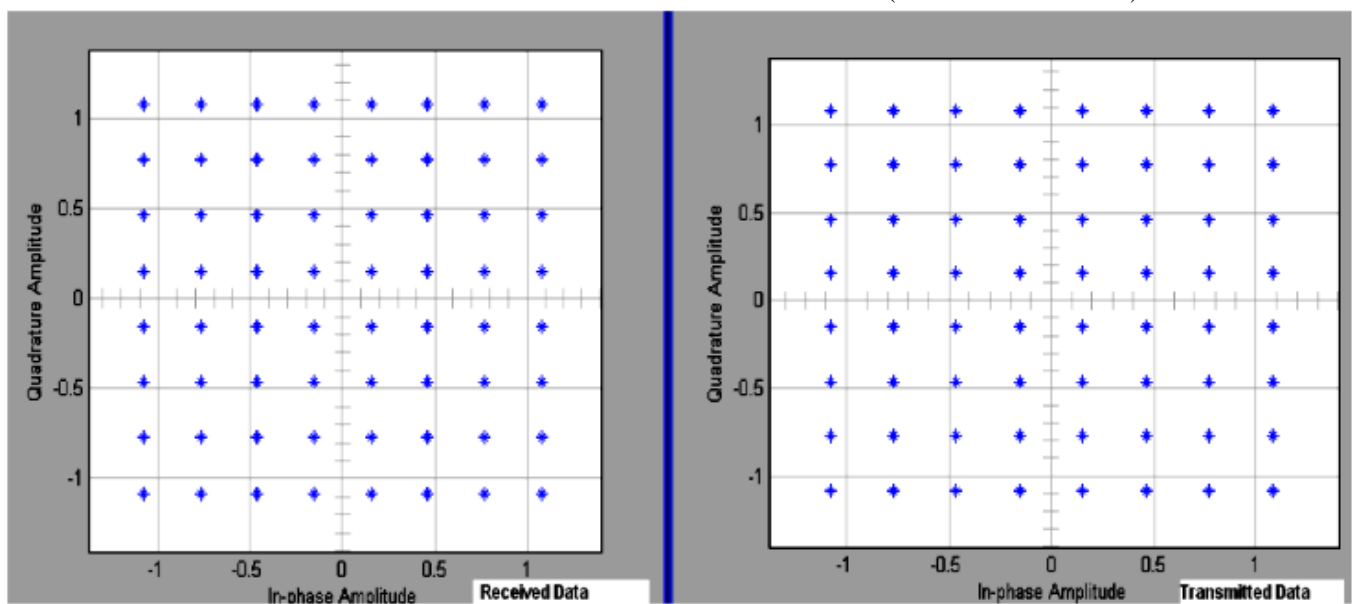
In the figure below graph 1 shows the the power of correlation $(R(d))2$ whereas the power of original signal $(P(d))2$. is shown in graph 2. If the ratio of both is greater than threshold then the packet is detected. This is used to minimize the false detection in case if correlating power is lower and original signal power is high, as packet should be detected only when ratio of $(R(d))2$ and $(P(d))2$ is resulting in the value above the threshold. The result (ratio) is plotted in graph 3. The graph 3 is shown as graph four after smoothing is performed. Last graph shows the detection of packet. In the circuit for delay and correlate algorithm, we have made a comparison between the output of correlated power and signal power that if both have value above than analytical threshold at the same clock cycle then the packet is considered to be detected. For this a logic AND gate is used in the circuit to get a high signal when both have value above the threshold [8].

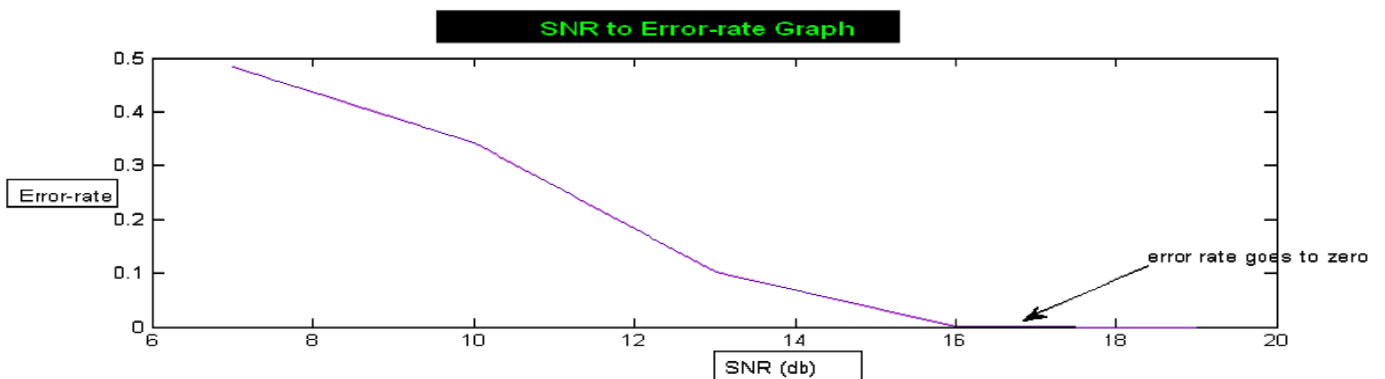FIGURE 4 : SIMULINK PACKET DETECTION SIMULATION RESULTS
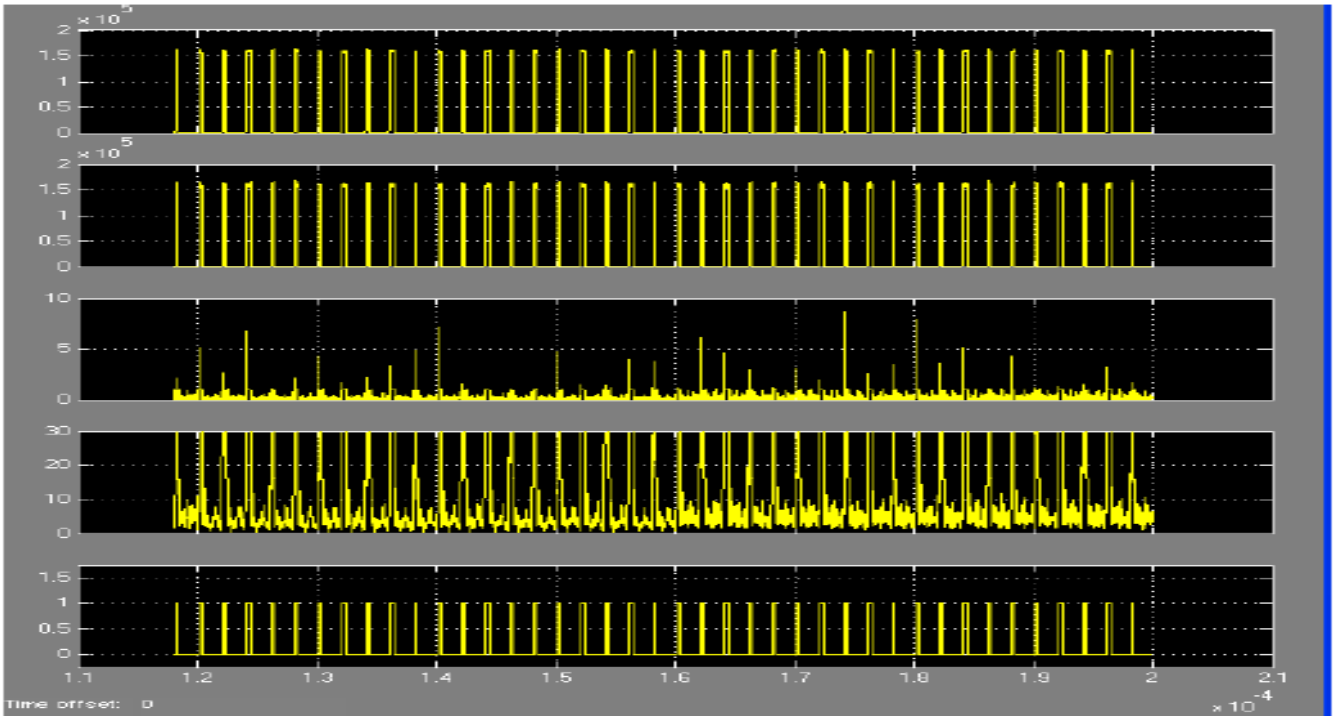
Error rate vs SNR

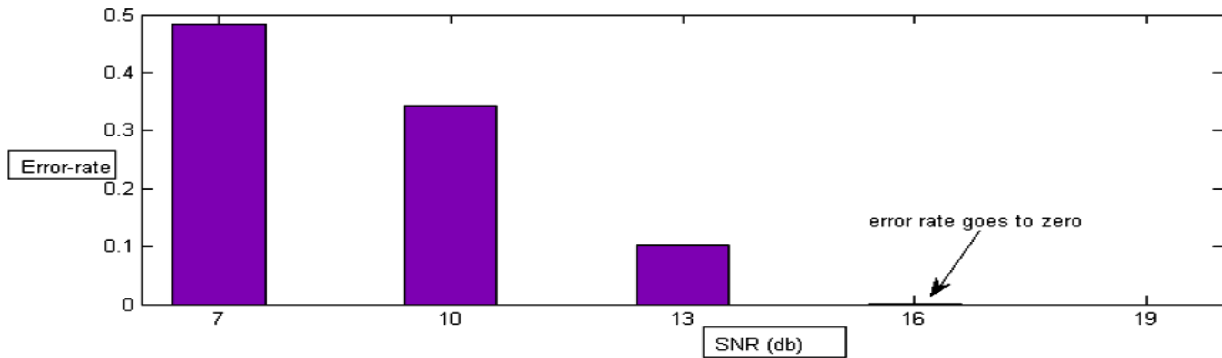Below is the table of error estimation for different values of SNR

TABLE 1: SNR Vs BER SIMULATION RESULTS

SNR	BER
7	0.4838
10	0.3435
13	0.1
16	0
19	0.0000

GRAPH 1: SNR Vs ERROR RATE SIMULATION RESULTS

GRAPH 2: SNR VS ERROR RATE BAR GRAPH SIMULATION RESULTS

IV. CONCLUSION

The above graphs and tables obtained from the simulation results clearly show that as the signal to noise ratio (SNR) increases which implies that when the signal power is increased, the error rate decreases, and it goes to zero at an SNR value of 16 db and onwards. We have also observed that at an SNR value of 7db the error rate goes to around 0.4838.

In this paper we described our implementation of IEEE 802.11a standard and simulated it on Matlab & Simulink using a step by step approach and achieved such high data rates. Simulink was used as the modeling tool. 64 QAM was used as the modulation technique. Our results are more than satisfactory and in future we plan to implement it on FPGA so that we can have a physical realization of this system.

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