

Real-time Network Simulation of 3GPP Long Term Evolution

I. Viering¹, C. Buchner¹, E. Seidel¹, A. Klein²

¹Nomor Research, ²Nokia Siemens Networks

{viering, buchner, seidel}@nomor.de, axel.klein@nsn.com

Abstract—In 2004 the 3GPP has started to standardize the new system “Long Term Evolution”. The key advantages are higher data rates, smaller latency, simpler network architecture and higher spectral efficiency.

The worldwide first live demonstration of LTE using our real-time simulation platform was given in September 2006. From that time the network simulator has been continuously enhanced with new functionality also considering latest progress in 3GPP standardization. The simulator shows the behavior of a future LTE network for various scenarios, under different load conditions. Real-time IP traffic is passed through the simulator demonstrating the user perception of future applications with different QoS configurations (e.g. video streaming and gaming). In particular, the aspect of radio resource management including scheduling functionality is highlighted and comprehensively illustrated via an elaborate graphical user interface.

I. INTRODUCTION

3GPP Long Term Evolution (LTE) [1] is the consequent evolution of the UMTS/HSPA [2] family towards 4G systems. Commercial rollout of this OFDMA based technology is expected in 2010. Being a purely packet switched system with only shared data channels, the network architecture will be much simpler compared with the current UMTS/HSPA. Flexible spectrum (1.25MHz – 20MHz bandwidth), higher data rates (peak rates up to 100Mbps) and smaller end-to-end latencies (down to 20ms) will be the corner stones.

The downlink is based on OFDMA. Consecutive subcarriers are grouped to resource blocks (RBs), which are individually assigned to the users based on both QoS requirements (from higher layers) and frequency selective channel conditions (from the physical layer).

In a purely OFDMA based uplink, the peak-to-average power ratio would make a reuse of the existing HSPA sites impossible, unless more expensive power amplifiers are employed in the UEs. Therefore 3GPP has decided to go for single carrier FDMA (SC-FDMA). This can be described as a derivative of OFDMA with the constraint that all subcarriers assigned to a user have to be adjacent. In this case, a linear predistortion can be applied to keep the PAPR small for the power amplifiers. Similar to the downlink, the subcarriers are grouped into RBs which are individually assigned to the users.

Hence, we are using frequency domain scheduling in both uplink and downlink. Note that although being also OFDMA

based, the resource assignment in WiMAX is not frequency selective which for sure is simpler, but also less efficient. The time granularity on which the resource allocation is re-decided, is 1ms. This period is called transmission time interval (TTI) in LTE.

This opens a new dimension to the MAC scheduler, compared with HSPA or WiMAX. The new degree of freedom has to be exploited to increase spectral efficiency whilst guaranteeing QoS for very individual users. Note that the scheduler is the only entity responsible for QoS, since only shared channels administrated by the MAC are available. Concluding, the MAC scheduler will be the key to a properly working system (satisfied users) with high spectral efficiency (satisfied mobile network operators).

II. DEMONSTRATOR

Since the first demonstration in September 2006 [3] our demonstrator is pioneering this emerging LTE technology.

A. Network simulator

In principle, it is a system level simulation modelling a cellular LTE network with many cells and many concurrent users in every cell. Due to the obvious computational complexity we do not have an OFDM-symbol-wise implementation of the physical layer (PHY), but the PHY models used are still rather exact. Basically those models calculate frequency-selective and time varying SINR's for every TTI of 1ms and every RB of 180kHz. The calculation considers frequency-selective small scale (“fast”) fading, and the prevailing (and also fluctuating) interference situation.

B. Applications

The purpose of every new communication system is not the technology itself, but to provide new applications to the end user. Therefore, we do not focus on investigating abstract statistics of throughput or delay distributions, we want to really experience the user perception which can be expected from the emerging technology. We want to see how real applications behave when being transmitted via the new systems. We are able to show different kinds of IP based applications via the LTE network simulation such as video streaming, gaming, web browsing, FTP, VoIP, etc.

Note that due to the shared channel nature, there will be a lot of interaction between the application and the MAC scheduler. In other words, the conventional way of generating error patterns offline to be used in the application testing

afterwards is not going to lead to meaningful insights in LTE. This adds a number of further requirements to the network simulation.

The most important one is the need for real-time capability. Applications obviously can only run properly if the connection is real-time. Hence, computational complexity is the most critical limitation, and reasonable trade-offs between accuracy and simplicity become the most important know-how.

Whereas typical network simulations model either uplink or downlink, we have to do both at the same time (i.e. bi-directional). This increases the computational problem; however, there are end-to-end effects which may remain hidden if one direction is replaced by an ideal link (e.g. end-to-end delays).

Since we want to transmit real IP packets, we cannot neglect any component of the protocol stack. We have to consider every segmentation, concatenation, retransmission, etc. on the long way from IP packets to physical transport blocks.

C. Visualization

We have already mentioned that the resource allocation is re-decided every 1ms. This short time scale has two different reasons, efficiency and latency. On one hand, we all know that the mobile channel suffers from fast variations which go down to the range of milliseconds (small scale fading). In the past we have learned that we can turn this suffering into a gain which is called multi-user diversity. This is only possible if we can track the channel by processing UE measurements sufficiently fast.

On the other hand, we have to quickly react on changing traffic conditions. We must be able to immediately allocate resources if requested. In addition, it is important to have very fast retransmissions of erroneous packets. In order to get the required small latencies (which is of significant importance for many applications), we have to observe those targets.

For ultimate comprehension of those interrelations it is inevitable to have instantaneous output graphs which need to operate on a 1ms time scale.

In addition, it would be extremely helpful for the examination of the numerous mechanisms, if parameters could be changed during run-time to see the immediate reaction of the system performance on the modification.

Figure 1 illustrates the graphical user interface of our demonstrator. In the lower left corner we can find a radio cell served by one base station. Many users are moving within this cell. There are also many surrounding cells producing interference. Every user is associated with a color which occurs in the output graphs on the right hand side.

In this example, we have shown only downlink plots. The lowest plot is the 2-dimensional SINR of 3 representative users. The brighter the color the better the channel decision. We can observe the aforementioned granularity of 1ms x 180kHz. We can see which user gets which resource in which TTI. The user throughputs are depicted above. The strategy is a proportional fair like, and thus the throughputs depend on the position of the users. At the top we see the cell throughput which is the

sum over all user throughputs.

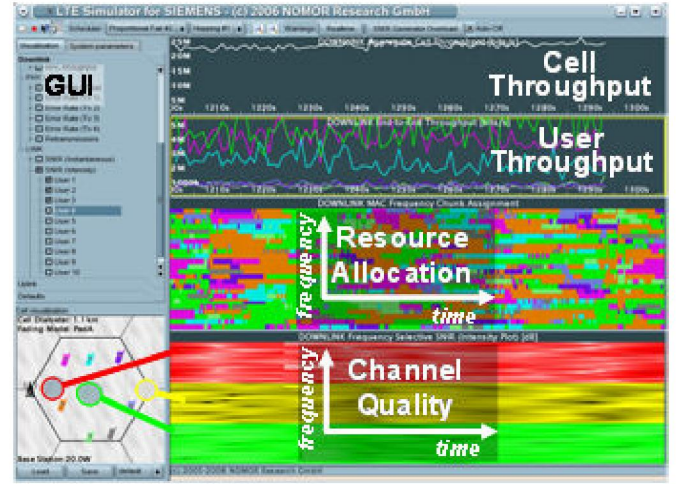


Figure 1 Graphical User Interface

On the left hand side we see a selection of the available options. We can add and remove individual plot, and we can change parameters for both downlink and uplink. In particular, we can choose between several scheduler strategies.

D. Summary: Cross-layer design

The combination of the aforementioned components is obviously a highly effective way to study and understand complicated (and sometimes hidden) cross-layer effects. As a consequence, such a demonstrator is a powerful assistant for a proper system design across all layers.

III. HARDWARE SETUP

Figure 2 shows the hardware setup. In principle we are running applications such as gaming or streaming between one or several clients and a server. The goal is to demonstrate how those applications behave in an LTE network under realistic traffic conditions, i.e. with many other users with different QoS requirements, and surrounding cells producing interference.

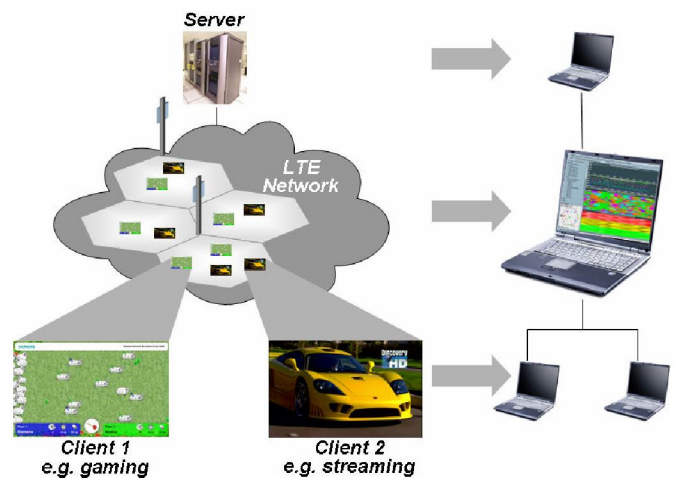


Figure 2 Hardware Setup

In our setup, we replace the loaded LTE network by the described network simulator which runs on a powerful laptop. Note that the simulator models the radio conditions not only for the “live” clients, but it also simulates many concurrent users of different QoS by synthetic traffic models, as well as surrounding cells by accurate interference models.

The application server is another laptop connected to the simulator via ethernet. The clients are also laptops and connected via ethernet hub to the simulator as well. Typically we are using 2 clients such that two users can play a game against each other.

IV. SHOWCASE

We have prepared two special showcases to illustrate the key advantages of LTE, a video streaming scenario to show the high throughput potential, and a gaming scenario to show the small latency capability of LTE. Furthermore, we can show numerous LTE specific effects upon request of the visitors.

A. Video Streaming

Both clients downstream an HD video which is H.264 encoded with variable bit rate and an average of approximately 4Mbps.

Within their cell, there are many other concurrent users, which we have configured as best effort (BE) users, i.e. they are using FTP or web browsing. The scheduling strategy is close to proportional fair, guaranteeing the required QoS to the video users through additional priority handling. The cell throughput is in the range of 20Mbps.

First of all the video users are close to the BS. In this case, their resource consumption is moderate. If we move a user to the cell edge, its resource consumption increases significantly, it has to get most of the available resources. The throughputs of the BE users degrade significantly, as well as the cell throughput. However both videos are still running stable.

If we remove the QoS awareness of the scheduler, or if we switch to a less intelligent scheduler such as Round Robin or max throughput, we observe clear codec artifacts, or the videos stop (in particular the cell edge user). The same happens if we deactivate other mechanisms such as MIMO/Rx diversity. If we go back to the original setting, the videos recover, and we can observe peak data rates of ~30-40Mbps for a short moment, since the buffers have to be cleared.

B. Gaming

The most challenging application for latency is gaming. However, please note that the majority of IP based applications is very sensitive to the latency. Although one can argue that a human being cannot distinguish between 20ms and 50ms delay, the applications amplify those delays to a perceivable level, e.g. due to TCP or due to multiple involvement such as in MS Outlook.

Two users can play a simple reaction game against each other. Although configuring a loaded up- and downlink, the latency introduced by the network is not perceivable. The gaming application measures roundtrip (end-to-end) latencies in the range of 30ms. A ping measurement in the background

shows even below 20ms.

Note that this comprises a number of mechanisms. Since there is only a shared channel, the terminal first has to request uplink bandwidth once the player has done a “click”. Only if it is granted radio resources by the BS, it is able to transmit the necessary gaming information. In case of erroneous transmission, retransmissions are necessary. Once the information has reached the server, similar procedures occur on the downlink.

C. Other Showcases

In addition to the described scenarios we can show a number of detailed effects depending on the interest of the visitors, e.g.:

- We have already a clear view on the radio resource management of the SC-FDMA uplink. We have implemented a consistent solution.
- We have also implemented interference mitigation schemes (for cell edge performance) for both uplink and downlink.
- We will also be able to show the impact of MIMO.

V. CONCLUSION

We are able to demonstrate the key advantages in a cellular LTE network under realistic traffic load already by now. Basically, we have a real-time system level simulation modeling many cells with many users. We can run real IP based applications over the simulator which provides the LTE “look and feel” years before its rollout. In addition, the demonstrator has an extensive graphical user interface, which allows a “hot” access (i.e. during runtime) to important system parameters, and which visualizes the complex procedures on a 1ms time scale. This combination enables comprehensive cross-layer studies of entire system design. Two showcases are prepared to illustrate the major advantages of LTE, video streaming for throughput and gaming for latency. Detailed LTE mechanisms can also be demonstrated for the more interested audience.

VI. REFERENCES

- [1] 3GPP 36.300, “E-UTRA and E-UTRAN: Overall description; Stage 2”
- [2] H. Holma, A. Toskala, “HSDPA/HSUPA for UMTS”, Wiley, 2006.
- [3] http://www.siemens.com/index.jsp?sdc_p=cfi1408787lmo1408787ps5u z1&sdc_bcpa th=1026937.s_5%2C%3A1408787.s_5%2C&sdc_sid=26504752647& (Press release)