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**On the importance of realistic traffic models
for wireless network evaluations**

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On the importance of realistic traffic models for wireless network evaluations

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Abstract—Little is known about the characteristics of mobile Internet data traffic generated by modern smartphones and laptops. A detailed understanding of these characteristics however is crucial for the design of low-overhead protocols, the development of efficient PHY layer techniques and to actually improve user experience instead of gross transmission rates. With this paper, we want to emphasize the importance of realistic traffic modeling for wireless network evaluations. We present a sample evaluation of the application layer performance of Coordinated Multipoint transmission schemes. We use traffic models with different level of detail and show which effects can be captured by which models. Finally, we want to encourage researchers to contribute and to expand our knowledge about mobile Internet data traffic, with the ultimate goal to design better networks for the future.

I. INTRODUCTION

Data traffic volume in cellular networks is growing at high speeds. A this year's traffic forecast by Cisco Systems predicts that worldwide mobile data traffic will more than double every year through 2014 [1]. According to this forecast, the traffic volume from 2009 to 2014 will increase by a factor of 39. The largest part of this increase is due to the more widespread usage of laptops, netbooks and other portable computing devices. The share of smartphones however is increasing. In current forecasts, mobile video is expected to be make up for the largest part of future data traffic.

Surprisingly, little is really known about the usage behavior and the traffic characteristics of nomadic and mobile Internet data services. If we knew how the profile of highly mobile users differ from stationary or nomadic users, we could use this information in the design of resource allocation algorithms in our base stations. If we knew about the distribution size of objects transferred in uplink and downlink direction, we could minimize protocol overhead and adapt holding times for bearers to save battery on mobile devices. A profound understanding of traffic characteristics is also crucial for the design and evaluation of new PHY and MAC layer techniques of future wireless networks. It is of little help if we are capable of achieving high peak gross data rates at the PHY layer, as long as applications cannot really benefit from it. The consequences of a lack of understanding of traffic characteristics could be observed a couple of years ago with the introduction of GPRS, where TCP flows transmitted over GPRS networks experienced large transmission delays and lead to bad link utilization [2].

In this paper, we want to emphasize the importance of

realistic traffic modeling for wireless network evaluations. As an example, we present an evaluation of the application layer performance of a generic Coordinated Multipoint (CoMP) transmission scheme. We employ an abstract model of a CoMP transmission scheme, which is described in the following section II. Section III presents our analysis of the interactions between traffic characteristics and the CoMP algorithm for traffic models with different level of detail. We present results on throughput and delay as seen by the application, which reveals that traffic characteristics have significant impact on coordination gains. In section IV, we review related work in characterizing mobile Internet data traffic and sum up the most important points in our conclusion.

II. GENERIC COMP MODEL

Coordinated multipoint (CoMP) transmission schemes are a promising technique to increase the spectral efficiency of cellular networks. Following the 3GPP terminology in the development of the LTE-Advanced specifications [3], we distinguish between the CoMP schemes Joint Transmission, Joint Detection, Coordinated Beamforming and Coordinated Scheduling. The common aspect of all these schemes is to improve the signal-to-interference ratio at the user terminals by coordinated resource allocation and scheduling among neighboring base stations. This improvement is achieved either by choosing orthogonal resources for interfering transmissions in the time, frequency or space domain, or by simultaneous transmission of the same data from several base stations. An overview and brief description of the various CoMP schemes is provided in [4]. In the following, we focus on Coordinated Scheduling and Coordinated Beamforming (CS/CB) in the downlink of an LTE-like system. We further assume all eNodeBs to cooperate using the X2 interface and do not consider deployments with remote radio heads.

The setup of a CS/CB transmission involving multiple base stations generally includes the following steps:

- 1) In addition to CQI and PMI measurements for its serving cell, the user terminal is requested to perform and report additional measurements to identify interferers.
- 2) Based on this information, the base station decides which neighboring cells are part of the CoMP set and sends signaling messages over the X2 interface to start a negotiation on resource blocks and PMIs to be used for this transmission. The coordination algorithms

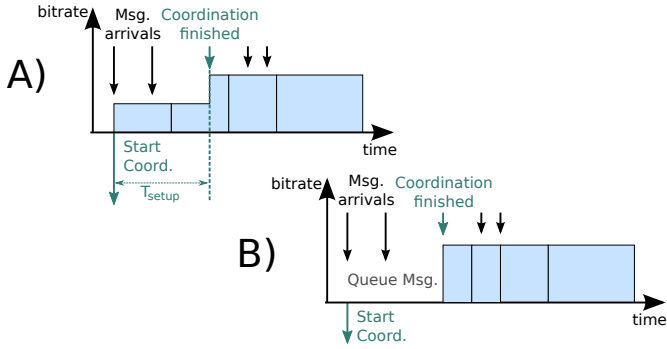


Fig. 1: Variants on the initiation of coordinated transmission

usually require one or more round-trip-times over the X2 interface to complete.

- 3) After this preparation, the base station starts transmitting data to the user terminal on the coordinated resources.

To abstract from a particular coordination algorithm, we define the term *coordination setup time* T_{setup} as the sum of steps 1 and 2. We measure the coordination setup delay starting from the time the user data arrives at the base station. Our setup delay T_{setup} thus also includes the time required for scheduling and is slightly larger than just the signaling message exchange over the X2 interface.

In our analysis, we take the point of view of one particular base station which initiates or stops coordination processes with its neighbors. To gain a general insight in possible interactions between CoMP algorithms and traffic characteristics, we do not explicitly model channel measurements nor signaling message exchanges because they are algorithm-specific and differ from one CoMP algorithm to another. We also do not account for the fact that neighboring base stations might also initiate coordinated transmissions.

A. Variants and Parameters of Coordinated Transmissions

We consider two basic alternatives about when to start a coordinated transmission. For *variant A* in Fig. 1, as soon as a message is received in the downlink, the message is being transmitted on uncoordinated resources at a certain constant rate R_U . At the same time, the base station initiates the coordination process with its neighbor base stations. After time T_{setup} , the coordination process is finished and the base stations have agreed on a set of resources, e.g. resource blocks and corresponding precoding vectors. In our model, the remaining data of the user queue is then transmitted at a higher rate $R_C = g \cdot R_U$ with $g > 1$.

For *variant B*, an incoming message also initiates the coordination process, but the data is buffered until the coordination process is complete. The message is thus sent completely at the higher rate R_C , but suffers from a small additional delay.

During user data transmission, the coordination process among the neighboring base stations has to be continuously ongoing to account for user mobility and varying radio channel quality. Once the user queue runs empty, all user data has been transmitted and coordination can be stopped. To relax

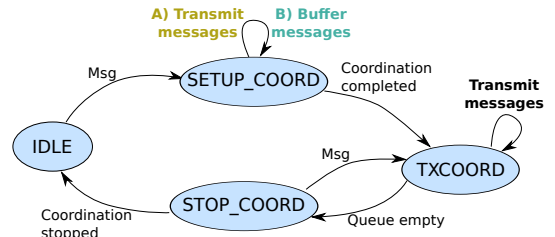


Fig. 2: Coordination model as finite state machine

the constraints on the time/frequency resources imposed by coordination, again signaling messages have to be exchanged with neighbor base stations.

B. Single User Model

To keep our evaluation model simple, we only consider a single user in the downlink of an LTE-like system. A user is served with constant (but different) rates while he is transmitting data on coordinated or uncoordinated resources. We thus abstract from any details of the radio channel. This model is adequate, as we are interested in average values over rather long time intervals. Evaluations with web traffic have to be conducted over multiple hours, due to the heavy-tailed distribution of HTTP object sizes. A per TTI model including fast fading, inter-cell interference and other effects of the radio channel would neither be feasible nor is it necessary here.

Figure 2 depicts the single-user coordination model described in the previous paragraphs as a state machine with four states. The only difference between variants A and B is in the handling of messages in the *SETUP_COORD* state, i.e. when we wait for the coordination process among the base stations to complete. The actual state machine is slightly more complicated due to the handling of coordination intervals, start and stop durations, but its implementation is straightforward.

III. ANALYSIS OF THE INTERACTION BETWEEN TRAFFIC CHARACTERISTICS AND CoMP ALGORITHMS

Data traffic in packet-switched networks, in particular the traffic generated by applications in the Internet, exhibits a very bursty behavior. A data transmission usually consists of several sequences of packets of different size with periods of inactivity in between. The length of these sequences, packet sizes and inactivity periods vary depending on the application and the protocols being used. Traffic models usually model only some aspects of the real world traffic characteristic. In the following, we briefly review a number of well-known models with different level of detail and apply them to our CoMP model.

A. Full buffer traffic model

CoMP schemes have been analyzed by a number of researchers. Large theoretical gains have been reported and its potential has also been confirmed by field trials. For details, consider [5] and the references therein. While early studies were conducted in an ideal setup with perfect channel state

information and zero delay for the signaling between base stations, more recent studies show that CoMP gains are sensitive to imperfect channel state information [6], [7]. However, in most of these studies, a full buffer traffic model was assumed, meaning that users always have data to transmit. This is a perfectly valid assumption to determine the theoretically achievable gains of the various CoMP schemes, but will lead to overly optimistic results if Internet data traffic is considered. Internet data traffic consists of burst arrivals of packets and idle times between these bursts. It thus constitutes a time-varying process whose characteristics have to be taken into account to determine the benefits of CoMP from a user perspective.

For the abstract CoMP considered here, the shortcomings of the full-buffer assumptions are obvious: if the buffer of our user never runs empty, the overhead from coordination setup and teardown will disappear and we only measure the higher coordinated data rate R_C . Besides throughput, the one-way-delay of a packet sent from the base station to the user is a very important characteristic to assess the responsiveness of an application and hence the quality of a service as experienced by the user. The full buffer model however falls short when we are interested in delay metrics, given that it only allows measuring transmission time and propagation delay but not queuing delay.

B. On/off traffic model

The next step towards more realistic traffic models is to use a so-called *on/off* traffic model. In this model, traffic is modelled as a sequence of on and off phases, which is represented by a state machine with two states. In the *on* state, there is data to send. After a certain time or after having sent a certain amount of data, the generator goes into the *off* state and stops sending data. The *on/off* model is an easily applicable model to mimic the bursty behavior of Internet data traffic.

The main parameters of this model are the activity factor, which is the ratio of *on* to *off* phases, and the average length of the *on* phase. By adjusting the activity factor and/or the length of the *on* phase, different load levels can be configured. However, there are still many degrees of freedom in the parameterization of this model, as e.g. the choice of the distribution of the length of *on* and *off* phases, which depend on the used protocols, the considered application and on the user behavior. There is no agreed reference parameterization, which makes results of different studies using this model hard to compare. It is therefore recommended to use similar, but application-specific models whose parameterization is well-described in literature.

C. Behavioral traffic models

A well-known behavioral model of web traffic is the model described in [8]. The model is similar to the *on/off* model described before and was used for evaluations in the context of LTE and is similar to the model presented by [9]. Web traffic here is regarded as a sequence of thinking times, transmission periods and parsing times. A web site consists of a main object and a number of embedded objects, which are

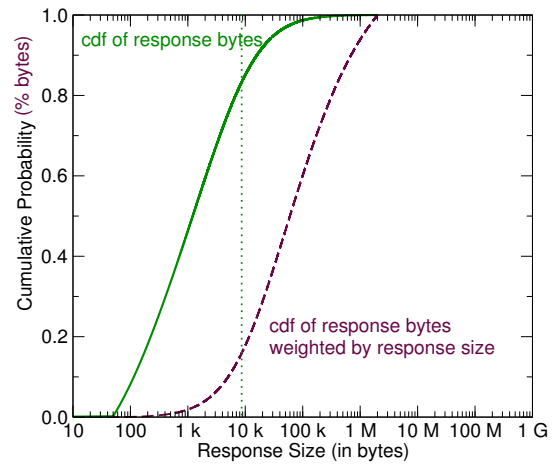


Fig. 3: Response object size in web traffic model

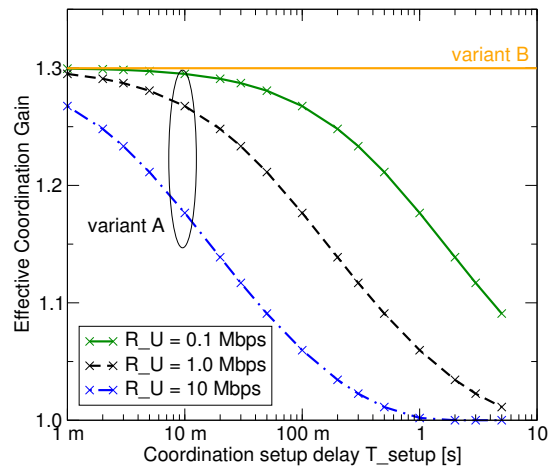


Fig. 4: Effective coordination gains as a function of T_{setup}

downloaded sequentially with parsing times in between. Main object and embedded object sizes follow a truncated lognormal distribution with different parameters, whereas the number of embedded objects follows a truncated heavy-tailed Pareto distribution. The parameters have been empirically derived from web traffic measurements in the 1990s. The model does not support HTTP pipelining nor keepalives.

Fig. 3 shows the cumulative distribution function of the object sizes of this model. The solid line gives the CDF of the response object size, which has a mean value of about 10 KBytes. The dashed line gives the response object size, weighted by its contribution to the overall traffic volume for a synthetic trace of 96 GBytes length. As it can be seen, over 80% of the objects are smaller than average, but account for only 10% of the traffic volume. This has quite a large impact on the observable gains of our generic CoMP transmission scheme. In our CoMP variant A, due to the small object sizes, it happens quite often that the user data buffer has run empty before a coordinated transmission could have been set up. This results into a lower effective CoMP gain, given that only a fraction of the overall traffic volume is being sent at the higher

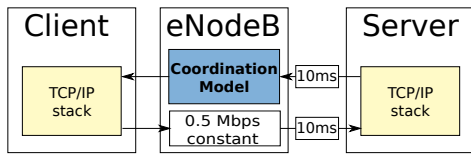


Fig. 5: Illustration of simulation setup and parameters

rate.

Figure 4 depicts this decay of the effective coordination gains as a function of T_{setup} , if an uncoordinated transmission rate R_U of 1 Mbps and a maximum gain of $g = 1.3$ is assumed¹. This simple analysis reveals that for high transmission rates and coordination setup times of several 10 ms, only a fraction of the higher throughput from coordinated transmission can actually be utilized in the system if variant A is used. For variant B, where data transmission does not start before the coordinated transmission with neighbor BS has been set up, the gains in throughput are obviously not affected by the object size distribution or the coordination delay. Variant B effectively trades higher throughput against additional queuing delay. The additional queuing delay in this model is the time required to set up a coordinated transmission T_{setup} .

D. Transport protocol effects

The traffic model presented in the previous section specifies the object sizes (web pages, images etc.) at the application layer. To model the packet arrival process at the base station, effects of the transport protocols and the underlying link layer technology have to be taken into account. The main effect of the link layer technology is the definition of minimum and maximum packet sizes. In the Internet, the packet size distribution has found to be bi-modal, with frequent packet sizes being around 40 Bytes and 1500 Bytes [10].

The predominant transport protocol in the Internet is TCP. Many different versions of TCP exist, which differ in aspects such as RTT estimation, ACK behavior and congestion control. After connection setup, TCP gradually increases the data volume which is sent out before the sender needs to wait for an acknowledgment. With every received acknowledgment this window size is increased, which results in an exponential increase of the data rate. This is known as the TCP slow start algorithm. As a consequence of TCP slow start, it takes a couple of round-trip-times to fully utilize the available transmission resources. Furthermore, during slow start, data transmission is not continuous, but is a sequence of IP packet bursts with idle times in between. Both effects might lead to interactions with CoMP algorithms. For more details on TCP, it is referred to literature, e.g. [11].

In order to get a deeper insight in the delay characteristics of the CoMP variants A and B, we use a simulation setup based on our institutes's simulation library IKR Simlib and

¹Given that CoMP gains in literature vary largely depending on which scenario is assumed, we arbitrarily chose $g = 1.3$ to illustrate our results. For other values of g , the absolute numbers differ but the principal behavior remains the same.

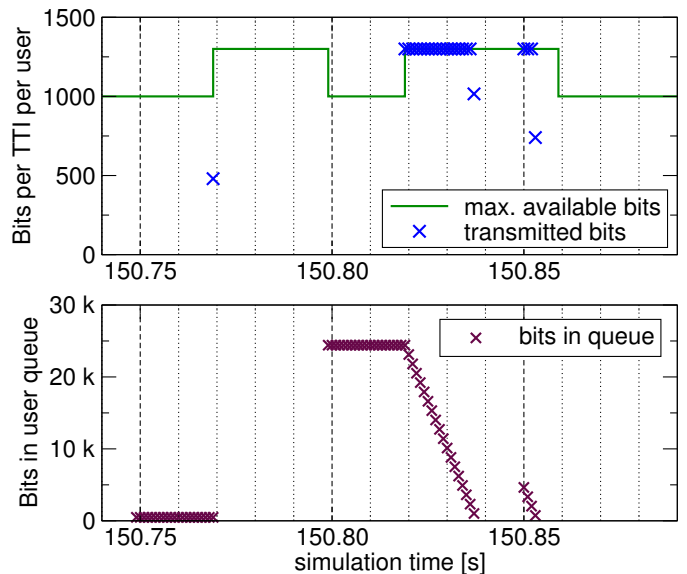


Fig. 6: Example of a transmission of a web page in variant B

the *Network Simulation Cradle (NSC)* [12]. With the NSC, we run the TCP/IP stack of the Linux kernel 2.6.26 in our simulation to account for the influence of TCP. The stack is configured to use TCP CUBIC with default parameters and runs on a 32 bit systems. Our traffic generator is implemented as a client process that requests web pages from a server process according to the web traffic model described in the previous section. The client is configured to open a new TCP connection for every new web page to sequentially download main and embedded objects. Our simulation setup is illustrated in Fig. 5.

Figure 6 depicts an example of the transmission of a single web page for variant B for $T_{\text{setup}} = 20$ ms. The lower graph shows the number of bits in the user buffer over the simulation time. The upper graph gives the number of bits that could be sent to the user in a single TTI (solid line) and the number of bits that have actually been transmitted (X symbols). Around $t = 150.75$, a SYN + ACK packet for TCP connection setup has arrived at the eNodeB. The packet is buffered for $T_{\text{setup}} = 20$ ms, before it is transmitted at rate R_C . Around $t = 150.80$, the HTTP response arrives at the eNodeB. Again, the data is buffered before transmission is started.

It can be seen that the additional delay introduced by CoMP variant B is substantially larger than just the coordination setup time $T_{\text{setup}} = 20$ ms. Due to the nature of the window-based transport protocol TCP and the structure of the web page downloads. A web page download consists of multiple request-response actions. Depending on the gaps in between and depending on the chosen coordination interval length T_{int} and coordination stop time T_{stop} (here $T_{\text{setup}} = T_{\text{stop}}$, a coordinated transmission might have already been stopped before the download of the web page is complete. The downloads thus suffer from multiple coordination setup delays, which increase the RTT seen by TCP. This in-turn leads to a slower increase

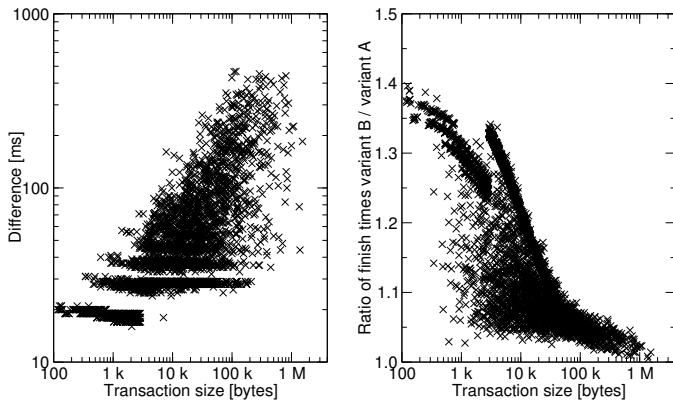


Fig. 7: Transaction finish times of variants A and B

of the congestion window size as compared to variant A and further slows the downloads down.

Figure 7 depicts the additional delay in terms of the *web page download time*. We define the download time as the time required to completely download a web page, which is the duration from the web page request of the client until the delivery of the last embedded object of the corresponding web page, measured at the client side. Figure 7 shows the difference and the ratio between the transaction finish times for variant B and for variant A. The ratio is largest for small transaction sizes and gets smaller as transaction size increases. For modern highly interactive web applications (e.g. web mailer, web office applications, gaming etc.), the additional delay for small messages might result in a less responsive behavior of the application and thus in a reduced quality of service. For large transaction sizes, the ratio is smaller but the absolute difference is in the order of several hundred milliseconds, which might well be noted by a user. A possible way to improve the transmission delay of variant B is to avoid buffering of small TCP packets, such as ACKs and SYN packets.

IV. RELATED WORK

Besides behavioral traffic models, there exists a class of models which try to generate characteristic packet patterns of aggregated traffic of a large number of users. These models fall short when it comes to modeling the service-specific behavior of single user sessions or small traffic aggregates, because of the largely distinct behavior of different application layer protocols, usage patterns and the inability to reflect the adaptive behavior of transport protocols [13]. These models are thus usually not applicable to wireless network evaluations, given that we are very often interested in the traffic from or to only a few user terminals.

In the class of behavioral traffic models, especially for web traffic, several alternative models exist. They range from simple request-response models with Pareto or Weibull distributed web page sizes [9], [14] to more complex models including inline objects, such as the SURGE model [15] or [16], which is a hierarchical model trying to mimic the correlations at web page and transport connection level. The downside of very

detailed source-level models is their complexity in terms of analytical tractability and processing time, which requires to trade accuracy against simulation time.

Besides interactive web traffic, conversational services such as circuit-switched voice telephony, Voice-over-IP (VoIP) or Internet telephony are most relevant to wireless networks. While human speech can be seen as a sequence of talk-spurt/silence periods that can be modeled as two-state Markov chains [17], codecs such as G.711 and G729.1 produce streams of fixed size packets, whereas G.723.1 and GSM AMR codecs detect silence periods and suppress packet generation or even insert additional silence insertion descriptor frames [18], [19]. The popular Internet telephony application Skype uses the iSAC code and adapts its encoding to the current network condition [20]. Further application classes include FTP and E-Mail, online gaming, video streaming and video conferencing, for which traffic models and reference parameter sets for studies in wireless networks are described in [8], [21].

However, using many detailed service-specific models is a large effort, how to choose the mix of the different traffic types is another unknown parameter and it is not clear whether model parameters extracted from measurements several years ago are still accurate in face of new mobile Internet applications. Another approach consists in using publicly available trace data of recent Internet traffic measurements. Just replaying packet traces however is not sufficient. Because of the adaptive behavior of applications, protocols and schedulers, a specific sequence of packets is only valid for the specific system state in which the trace was captured.

The transport protocol research community has addressed this problem by transforming the packet level traces into a network-independent description of data traffic [22]. The trace data provided by the University of North Carolina (UNC) [23] consist of the TCP/IP headers of the traffic from and to the UNC campus were captured over a period of several hours in 2008. The traces were processed to a sequence of request-response patterns that are typical for web usage and other applications, following the methodology described in [24]. The trace data is suited for the so-called *a-b-t* model, which is a non-parametric traffic model for request-response types of applications [25]²: A request-response communication in this model is described as a sequence of *connection vectors*, which are triplets of request size *a*, response size *b* and idle time *t*. A typical web session can thus be described as a sequence of connection vectors. An implementation of this traffic model is described in [26].

The advantage of the *a-b-t* model is, that it can be easily generated from packet-level traces, which can be collected in a live network in a non-invasive way. Subject to the availability of up-to-date trace data, the model provides a nice way to accurately model packet and object size distributions, inter-arrival times, traffic volume and application behavior. However, up to now, no recent trace data of the mobile Internet usage in

²The *a-b-t* model is also applicable to other communication patterns than request-response sequences, but this is not considered here.

wireless networks is available. Mobile network operators are very reluctant to collect these traces. The authors of [27] have recently published a promising approach, where they install special apps on the smartphones of test users and collect the traffic characteristics directly on the device. By doing this, the authors are able not only to collect packet level traces, but also to identify the application and to collect information about the context in which the application was executed. The study still suffers from a small data base, but nevertheless already provided a number of valuable insights regarding the protocol overhead in wireless networks. Hopefully, in one or two years, the data base is large enough to allow for more accurate models of Internet data traffic in wireless networks that will help our community to design more efficient networks.

V. CONCLUSION

Our contribution in this paper is two-fold: First, we presented an analysis of the interactions between data traffic characteristics and Coordinated Multipoint transmission algorithms. We revealed that traffic characteristics, if not taken into account in the design of CoMP algorithms, can lead to largely reduced coordination gains or additional transmission delays with a possibly negative impact on user experience.

Second, we made a case for the development and publication of up-to-date traffic models for wireless network evaluations. As of today, little is known about the characteristics of mobile Internet data traffic. This prevents us from optimizing our protocols and algorithms for common usage scenarios, which results in a poor overall efficiency of our networks. Being oblivious to the traffic characteristics of mobile Internet applications might undermine the large effort spent on the PHY layer to achieve higher data rates, if in the end the applications are not able to benefit from it. In this paper, we provided a brief review of available traffic models and made suggestions about how and which models to use in wireless network evaluations. We encourage other researchers to contribute and collect measurement data on the usage of mobile Internet data service. With this data, our community will be able to design much more efficient networks which can provide a better user experience than today.

REFERENCES

- [1] Cisco Systems, "Cisco visual networking index: Global mobile data traffic forecast update, 2009-2014," White Paper, Cisco Systems, <http://www.cisco.com>, February 2010.
- [2] R. Chakravorty, S. Katti, I. Pratt, and J. Crowcroft, "Using tcp flow-aggregation to enhance data experience of cellular wireless users," *Selected Areas in Communications, IEEE Journal on*, vol. 23, no. 6, 2005.
- [3] 3GPP TR 36.814, "Further advancements for E-UTRA physical layer aspects," 3GPP Technical Report, Release 9, Mar. 2010.
- [4] M. Sawahashi, Y. Kishiyama, A. Morimoto, D. Nishikawa, and M. Tanno, "Coordinated multipoint transmission/reception techniques for lte-advanced," *Wireless Communications, IEEE*, vol. 17, no. 3, 2010.
- [5] V. Jungnickel, L. Thiele, T. Wirth, T. Haustein, S. Schiffermuller, A. Forck, S. Wahls, S. Jaeckel, S. Schubert, H. Gabler, C. Juchems, F. Luhn, R. Zavrtak, H. Droste, G. Kadel, W. Kreher, J. Mueller, W. Stoermer, and G. Wannemacher, "Coordinated multipoint trials in the downlink," in *IEEE GLOBECOM Workshops*, 2009.
- [6] P. Marsch and G. Fettweis, "On downlink network mimo under a constrained backhaul and imperfect channel knowledge," in *IEEE GLOBECOM 2009*, 2009.
- [7] R. Ballentin and M. Doll, "Downlink coordinated scheduling - simulation results," ITG 5.2.4 Workshop on IMT-Advanced, available at <http://www.ikr.uni-stuttgart.de/Content/itg/fg524/Meetings/2010-07-08-Heidelberg/index.html>, July 2010.
- [8] NGMN Alliance, "Radio access performance evaluation methodology," available online at <http://www.ngmn.org/>, June 2007.
- [9] H.-K. Choi and J. O. Limb, "A behavioral model of web traffic," in *Proceedings of the Seventh Annual International Conference on Network Protocols*, ser. ICNP '99. Washington, DC, USA: IEEE Computer Society, 1999, pp. 327-.
- [10] R. Pries, F. Wamser, D. Staehle, K. Heck, and P. Tran-Gia, "Traffic measurement and analysis of a broadband wireless internet access," in *IEEE 69th VTC Spring*, 2009.
- [11] A. Afanasyev, N. Tilley, P. Reiher, and L. Kleinrock, "Host-to-host congestion control for tcp," *IEEE Communications Surveys & Tutorials*, vol. 12, no. 3, 2010.
- [12] S. Jansen and A. McGregor, "Simulation with real world network stacks," in *Simulation Conference, 2005 Proceedings of the Winter*, 2005.
- [13] N. Vicari, "Modeling of internet traffic: Internet access influence, user interference, and TCP behavior," Ph.D. dissertation, University of Würzburg, Apr. 2003.
- [14] B. Mah, "An empirical model of http network traffic," in *IEEE INFOCOM*, vol. 2, 1997.
- [15] P. Barford and M. Crovella, "An architecture for a www workload generator," World Wide Web Consortium Workshop on Workload Characterization, Oct. 1997.
- [16] E. Casilari, A. Reyes-Lecuona, F. Gonzalez, A. Diaz-Estrella, and F. Sandoval, "Characterisation of web traffic," in *Global Telecommunications Conference, 2001. GLOBECOM '01. IEEE*, vol. 3, 2001, pp. 1862-1866.
- [17] P. Brady, "A statistical analysis of on-off patterns in 16 conversations," *Bell Systems Technical Journal*, vol. 48, no. 9, 1969.
- [18] A. Estepa, R. Estepa, and J. Vozmediano, "A new approach for voip traffic characterization," *Communications Letters, IEEE*, vol. 8, no. 10, pp. 644-646, 2004.
- [19] M. Menth, A. Binzenhöfer, and S. Mühlecke, "A note on source models for speech traffic," *SIGMETRICS Perform. Eval. Rev.*, vol. 35, no. 3, pp. 22-24, 2007.
- [20] T. Hoffeld and A. Binzenhöfer, "Analysis of skype voip traffic in umts: End-to-end qos and que measurements," *Computer Networks*, vol. 52, no. 3, pp. 650-666, Feb. 2008.
- [21] WiMAX Forum, "WiMAX system evaluation methodology," WiMAX Forum, Version 2.1, Jul. 2008.
- [22] L. Andrew, C. Marcondes, S. Floyd, L. Dunn, R. Guillier, W. Gang, L. Eggert, S. Ha, and R. I., "Towards a common tcp evaluation suite," in *PFLDNet*, 2008.
- [23] California Institute of Technology, Networking Lab, "Traffic traces for tcp evaluation," available online at <http://wil.cs.caltech.edu/suite/TrafficTraces.php>, 2008.
- [24] F. D. Smith, F. H. Campos, K. Jeffay, and D. Ott, "What tcp/ip protocol headers can tell us about the web," in *Proceedings of the 2001 ACM SIGMETRICS conference*, USA, 2001.
- [25] F. Hernandez-Campos, "Generation and validation of empirically-derived tcp application workloads," Ph.D. dissertation, University of North Carolina at Chapel Hill, Departement of Computer Science, 2006.
- [26] M. C. Weigle, P. Adurthi, F. Hernandez-Campos, K. Jeffay, and F. D. Smith, "Tmix: A tool for generating realistic tcp application workloads in ns-2," *ACM Sigcomm Computer Communication Review*, vol. 36, no. 3, July 2006.
- [27] H. Falaki, D. Lymberopoulos, R. Mahajan, S. Kandula, and D. Estrin, "A first look at traffic on smartphones," in *Internet Measurements Conference (IMC)*, Melbourne, Australia, November 2010.