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Analysis and Modeling of Very Large Network Topologies

(Invited Paper)

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***Abstract*—We consider the issue of modeling huge, random network topologies that are too large to capture in full details. Such enormous, hard-to-describe network topologies are becoming ubiquitous in numerous settings. The Internet and its logical overlay networks, such as the World Wide Web, as well as online social networks, are well known examples. At the same time, extensive and rapidly growing wireless ad hoc and sensor networks also lead to hard topology modeling questions. In the current paper we primarily focus on large, random wireless networks. We provide a common generalization of various models that covers a number of known models as special cases. We also demonstrate that such a higher level abstraction, despite its very general nature, can still be meaningfully analyzed, and offers quite useful and unique help in solving certain hard networking problems.**

I. INTRODUCTION

Many of the communication networks that we use today, or expect to use in the future, have enormous size. This applies not only to the physical networks, including the Internet as well as emerging ubiquitous wireless networks and large scale sensor networks, but also, or even more, to logical overlay networks, such as the World Wide Web. For example, the number of web pages, according already to a 2006 article [31], was as high as 53.7 billion, already at the time of writing that study. Out of the 53.7 billion, 34.7 billion web pages were indexed by Google. Since then, these numbers grew even further. Beyond the sheer size, the usage of these networks is also expected to be extremely heterogeneous, encompassing a huge number of different applications, traffic patterns, diverse requirements and areas, including business, science, learning, entertainment, social networking and a great many more. At the same time, their physical basis is also heterogeneous, including wired, wireless, optical subnetworks. All this is expected to eventually merge into a ubiquitous, global socio-technical infrastructure.

To understand and reason about huge socio-technical networks, including methods for designing/optimizing them, the traditional network analysis and modeling approaches are generally insufficient, due to their *limited scalability.* Simulation is usually feasible only up to a rather limited network size. Conventional analysis methods, such as teletraffic theory, queuing network modeling etc., also face an uphill battle, quickly losing ground in huge networks. At the same time, modeling and analysis is still indispensable, since one may not be able to experiment with the different variants of a new solution via large scale practical deployment, as it can have a prohibitive cost.

This situation, in which one deals with networks of practically infinite size, has naturally led to the emergence of novel analysis and modeling approaches. They can generally be characterized by having a more abstract, "bird's eye" view of the network and often relying on *asymptotic analysis* on the mathematical side. The special advantage of the asymptotic analysis is that it converts the growing size from a foe to a friend: the larger, the better, from the asymptotic point of view. While it is clear that such methods cannot help much in *local* technical tasks, such as configuring a specific router, they have their important place in the higher layers of the network modeling hierarchy. In the next section we briefly survey how this approach emerged.

II. HISTORY

The first major wave of work in the considered direction was the **experimental statistical analysis** of the Web graph, in which the nodes are web pages and the edges are the hyperlinks. Several research groups in the late 90s independently observed that the node degrees in this graph are distributed according to a *power law* (Kumar et al. [36], Barabási and Albert [5], [6], Broder et al.[12]). Similar phenomena were observed by Faloutsos et al. [19] in the physical Internet topology. All this was in sharp contrast with traditional random graph models that have independent edges, and exhibit (asymptotically) Poisson node degree distributions. The latter models are known as Erdős-Rényi random graphs.

To describe the observed network structure, Barabási and Albert [5] coined the term *"scale-free network"*, based on the observation that in a power law distribution the rescaling of the considered quantity preserves the same power law, changing it only with a constant factor. This quickly became very popular, and triggered the statistical analysis of "scale-freeness" of network topologies not only in (physical or logical) communication networks, but also in networks that arise in biology, genetics, epidemiology, linguistics, electric power distribution, social sciences and in many other areas; see, e.g., the books [10], [11], [13], [14], [42], and hundreds of further references therein.

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In retrospect, one may say that "scale-free networks" generated somewhat more hype than substance. It was rightfully pointed out, e.g., by Li et al. [40] and Alderson et al. [4] that the power law degree distribution alone can easily fall short from adequately modeling the Internet topology, if no other domain specific knowledge is applied.

A parallel major wave of research was to create **generative models**. In contrast to experimental statistical analysis, generative models aim at explaining the observed network structures, and provide algorithmic approaches to generate them, also offering the opportunity for in-depth mathematical analysis. The first such model that became well known was the *Preferential Attachment* model of Barabási and Albert [5]. This model generates a graph such that new nodes are more likely to get connected to those that already have a higher degree. Although the authors did not provide a rigorous analysis, only an approximate reasoning, based on the mean-field approach of physics, the model certainly had intuitive appeal ("the rich get richer" principle). This model had an explanatory power and generated scale-free graphs, so it triggered many follow-up investigations. The first truly rigorous formulation and deep analysis of a preferential attachment model, called Linearized Chord Diagram (LCD) model, was provided by Bollobás at al. [9].

Since then, numerous static and evolving graph models of networks have been proposed and analyzed, both experimentally and with serious mathematical rigor, primarily focusing on asymptotic properties. A few examples are: the ACL model of Aiello, Chung and Lu [2]; the copying model of Kleinberg et al. [33]; the growth-deletion model of Chung and Lu [15]; the self-similar Kronecker-graph model of Leskovec et al.[37]; the compressible Web model of Chierichetti et al. [16]; the forest fire model of Leskovec, Kleinberg and Faloutsos [38]; the geometric preferential attachment model of Flaxman, Frieze and Vera [26]; the spatial preferential attachment model of Aiello et al. [1]; the random perturbation model of Flaxman [25]; as well as a large number of other models and variants, with a lot of intellectual power in their analysis.

About the same time when the above outlined investigations began, another independent wave of asymptotic network modeling was initiated by Gupta and Kumar [29]. This direction focused on analyzing the **scalability of large wireless networks**, primarily ad hoc and sensor networks, from the viewpoint of fundamental limits for transport capacity and related properties. This line of research also attracted much attention. Interestingly, and unfortunately, most of the results are negative. Specifically, they show under various conditions, that the achievable per node throughput tends to 0 with growing network size. Even maintaining global network connectivity is impossible under rather general conditions, if we want to apply nodes with finite processing capacity, see Faragó [21]. Nevertheless, there are also notable exceptions, utilizing various effects, such as mobility (Grossglauser and Tse [28]), restricted traffic pattern (Li et al. [39]), using infrastructure (Liu, Liu and Towsley [41]), or relaxing the condition of full connectivity (Dousse, Franceschetti and Thiran [18], Faragó [24]), to mention only a few examples. Therefore, the issue of wireless network scalability is still under further research.

The graph models that are used in the wireless network investigations are very different from the Internet and Web models. The random graphs in wireless network analysis are based on geometric considerations, and termed **geometric random graphs**. They also have a rich set of analytical results, see, e.g., the books of Franceschetti and Meester [27], and Penrose [44]. In a sense, geometric random graphs are between the classical Erdős-Rényi model and those graph models that are used to describe the Internet and Web topologies. Specifically, geometric random graphs have (asymptotically) Poisson node degree distributions, just like the Erdős-Rényi random graphs. That is, geometric random graphs (modeling wireless network topologies) do *not* exhibit scale-free behavior. On the other hand, their edges are not independent, just like in the Internet/Web models, so they have many properties that are distinctively different from the Erdős-Rényi random graphs.

The current situation. As briefly reviewed above, there exists a vast and rather diverse body of various graph based network models that are mostly analyzed from the viewpoint of asymptotic properties. Note that beyond the theoretical advances they also have emerging practical applications, such as Internet topology generators, search engine optimization, protocol design and optimization in wireless networks etc.

The current situation on the model development and analysis side (which is our primary interest) can be characterized with the following:

- The diversity of models also led to the diversity of analysis methods. With minimal exaggeration one can say that a new analysis method has to be invented almost each time when a new model is proposed. There is a sense of missing unification and a lack of general methods that apply to large families of different models.
- The analysis is often very hard and typically cannot rely on the well developed methods of classical random graph theory, as pointed out by leading experts in the theory of random graphs (Bollobás and Riordan [8]).
- Despite the existence of emerging applications, there is still a large gap between analysis results of descriptive nature and methodology/algorithms that provide meaningful help in network design problems.
- Validation of models is a problem. As pointed out by A.D. Flaxman [25]: "Unfortunately, it is much easier to propose a generative model than to refute one."

III. The Case of Large, Multi-hop Wireless Networks

Wireless networks of large size, random topology and no supporting infrastructure, such as ad hoc and sensor networks, are expected to play an important role in the future. The random network topology of these systems is frequently described by various random graph models, most often by some variant of geometric random graphs. First we review some of the typical classes of graphs that are used in this context.

A. Some Frequently Used Graph Classes for Wireless Network Topology Modeling

An important class is the *Unit Disk Graph (UDG)* [17] model of the network topology. A UDG is a graph that is defined by the (planar) geometry of node positions. It is assumed that each node has the same transmission radius r, and two nodes are connected by a link if and only if they are within distance r (which is often normalized to $r = 1$, hence the name). In other words, the radio range of each node is just a circular disk. As a critical difference from the physical model, in a UDG it does not matter where the rest of the nodes are located and how much interference they generate.

A clear advantage of UDGs is that a number of important algorithmic problems that are NP-complete for general graphs become solvable in polynomial time for this special class [45], thus allowing much more efficient protocols.

Unfortunately, the UDG model is quite simplistic, it is rather far from accurately reflecting the actual radio network topology. A refinement is the *Quasi-Unit Disk Graph (Q-UDG)* model [34], in which a *shrink factor* ρ is added, with $0 < \rho < 1$, for describing the radio range of a node by two concentric circular disks, the outer one with radius r, and the inner one shrunk by the factor ρ, yielding radius ρr. If two nodes are at most ρr distance apart, then they are always connected by a link. If they are more than r apart, then they are never connected. Finally, if the distance is between ρr and r, then the link may or may not exist. Geometrically this means that the radio range of a node can have arbitrary shape, but moderated by the requirement that it should be between a circumscribed circle of radius r and an inscribed circle of radius ρr.

A nice feature of Q-UDGs is that, while providing a more general network topology model, they still preserve the algorithmic advantages of UDGs, at the price of an additional $1/\rho^2$ factor in complexity [34]. Thus, if the shrink factor ρ is a not too small constant, then most of the UDG advantages carry over, with only a constant factor penalty in complexity.

Another natural issue is that different nodes may transmit with different power, or have different spectrum-dependent attenuation of the transmission signal [3]. This leads to the concept of *Disk Graph (DG)*, which differs from the UDG in that each node i has its own, possibly different, transmission radius r_i, and two nodes are connected by an undirected link if they are mutually in each other's range. DGs are somewhat less friendly from the algorithmic point of view than UDGs and Q-UDGs, but still better than general graphs and still allow efficient solutions or approximations for a number of algorithmic problems, as we investigated in [46].

Similarly to the generalization that leads to the Q-UDG concept, one can also introduce *Quasi-Disk Graphs (Q-DG)*, by adding a shrink factor ρ that allows to refine the radio range description as for Q-UDG.

All the above graph models can naturally be extended to higher dimensions, replacing the disks by balls in the appropriate space.

A common nontrivial generalization of all these graphs, the *Bounded Independence Graph (BIG)* model is also worth mentioning [45]. (It is also referred to as *Bounded Growth Graph* [35].) This class is defined by the requirement that the maximum number of independent nodes[1] within the k-hop neighborhood $\mathcal{N}_k(v)$ of any node v is bounded by a polynomial of k. Although this definition is based purely on the graph structure and does not have a direct geometric meaning, it can still be related to geometry through the concept of *doubling metric spaces* [45]. These are metric spaces in which any ball of radius r can be covered by a finite number of balls of radius $r/2$. This property does not hold for all metric spaces, although it holds for Euclidean spaces of any finite dimension[2]. It can be shown that if a geometric graph is defined in a doubling metric space, in analogy with UDG or DG, then it is always a Bounded Independence Graph [45]. A nice feature of this class is that a number of hard algorithmic problems become efficiently solvable in it [35].

So far we have described these classes deterministically, ignoring randomness. Of course, from each graph class one can generate random members, according to various probability distributions. These are usually defined indirectly, through some generating mechanism. For example, if we pick the node positions uniformly at random in a planar domain, e.g., a square, and then construct a UDG over these nodes, then we get a Random Unit Disk Graph.

All these graph classes are related to some kind of geometric insight. It is not surprising, since geometry and distance play a key role in forming the radio network topology. On the other hand, radio propagation (with possible obstacles and other irregularities) can induce much more complicated distances that may not satisfy the mathematical distance axioms, primarily the triangle inequality. Nevertheless, even in this more complicated situation, it is still possible to meaningfully analyze geometric-like graphs and prove nontrivial results about important properties, such as connectivity, as we are going to see in connection with our *Abstract Geometric Random Graphs.*

B. The Issue of Connectivity

Because of the random network topology, it is not at all guaranteed that any two nodes can send messages to each other, as the random graph that represents the network topology may not be connected. To guarantee that all nodes can reach each other, a minimum requirement is that the network topology (which is usually represented by an undirected graph) is *connected.* Since connectivity is a particularly important property, we select it as the focus of our discussion.

Unfortunately, the connectivity requirement is not as innocent as it may look, due to random node positions and limited wireless transmission ranges. It turns out (see, e.g., Gupta and Kumar [29], [30]) that in typical cases, such as placing

[1] A set of nodes in a graph is called independent if there is no edge between any two of them.

[2] Radio propagation properties may lead to a "radio-distance" that is quite different from Euclidean.

the nodes in a planar disk independently and uniformly at random, the price of connectivity is very high: the transmission range needs to be set such that it asymptotically results in an infinitely growing number of neighbors.

This phenomenon is a serious *threat to scalability* in these networks. After all, one cannot expect that a small wireless node with limited power and modest capabilities can serve an unbounded number of neighbors.

One might hope at this point that for different modeling assumptions the situation may perhaps improve. For example, one may try different deployment domains, different probability distributions, different distance metrics, etc. Unfortunately, however, it has been proven in a very general model that none of these can relieve the scalability bottleneck, see Faragó [23]. It appears that unbounded node degrees are unavoidable whenever full connectivity is required in the limit in a random, geometrically induced topology. This is, of course, bad news for hoping a scalable implementation.

It is therefore of keen importance whether better scalability can be achieved if we are willing to give up full connectivity and substitute it with the milder requirement of *partial connectivity*. This means, as a price for keeping the node degrees bounded, we accept that only most, but not all, nodes are in a connected component. The motivation is that in many potential applications, such as a network of randomly placed sensors, it is acceptable to have only a majority (say, 99%) of nodes in a connected component and the rest are possibly disconnected.

We review some results on the fundamental limits related to such partial connectivity, under the most general modeling assumptions we can set up. Specifically, based on our earlier work [22], we exhibit the asymptotically optimal trade-off between the fraction of nodes that can be kept in a connected component as a function of the bound on the expected node degrees.

IV. A MOTIVATING EXAMPLE

Let us consider a large sensor network. Due to the limited processing capabilities of the small sensor nodes, each one is capable of maintaining connections only to at most three other nodes in our example. The existence of wireless links depends on distance, but the actual form of the dependence is unknown. Moreover, random obstacles to radio waves are also present, and two nodes can only communicate if no such obstacle separates them.

The sensor nodes are distributed in space independently, according to a common, but unknown probability distribution. The locations of the random obstacles are also independent of each other and of the node locations, but otherwise the position, size and shape of each obstacle can have an arbitrary probability distribution, which is again unknown. We only assume that the events that links are blocked by an obstacle can be considered independent.

Without further information about this sensor network, is it possible to provide a nontrivial lower bound on the number of sensors that will be necessarily pushed to the “periphery”? By periphery we mean those nodes that are not part of the largest connected component of the network topology.

The traditional approach to answer this question would be to specify the probability distributions and other parts of the model (such as how link existence depends on distance, etc.), and then do (tedious) calculations under the specific conditions. If, however, anything changes in the conditions, the results may not carry over. Our general approach will make it possible to avoid this, and provide a nontrivial bound that is valid for all practically relevant cases.

V. RANDOM GRAPH MODELS IN THE MOST GENERAL SETTING

In order to build up our modeling approach, let us first explain what we mean by random graphs and a random graph model in the possibly most general sense.

In full generality, by a *random graph* on a fixed number of nodes (n) we mean a random variable that takes its values in the set of all undirected graphs on n nodes. We are going to denote by G_n a random graph on n nodes. At this point, it is still completely general, possibly generated by any mechanism, with arbitrary dependencies among its parts, it is just *any* graph-valued random variable, taking its values among undirected graphs on n nodes.

Having introduced general random graphs, a *random graph model* is given by a sequence of graph valued random variables, one for each possible value of n:

$$\mathcal{M} = (G_n;\ n \in \mathbf{N}).$$

Next we introduce some general features that apply to any random graph model.

A. Degrees and Connectivity

Let G_n be any random graph on n nodes and let us denote by $e(G_n)$ the number of edges in the graph. We characterize the degrees of G_n by the expected degree of a randomly chosen vertex, which we call he *expected average degree* of G_n. It is denoted by $\overline{d}(n)$ and defined by

$$\overline{d}(n) = \frac{2\mathrm{E}(e(G_n))}{n}.$$

It is based on the fact that the actual average degree in any graph G on n nodes is $2e(G)/n$. Often the expected degree of each individual node is also equal to $\overline{d}(n)$, but in a general model it may not hold. (Note that even if the expected degree of each node is equal to the expected average degree, it does not mean that the *actual* random degrees are also equal.)

Ideally, we would like a random graph model in which $\overline{d}(n)$ remains bounded by a constant and the model is *asymptotically almost surely (a.a.s.)* connected, meaning

$$\lim_{n\to\infty} \Pr(G_n \text{ is connected}) = 1.$$

Note: Whenever we write down a limit, such as the one above, we also assume that the limit exists.

Since, as mentioned in Section III-B, asymptotic connectivity is not possible in most models without unbounded degrees, therefore, one may hope that if less than full connectivity is required, then there is a better chance to keep the node degrees bounded. To this end, let us define a weaker version of connectivity.

Definition 1: **(β-connectivity)** *For a real number $0 \leq \beta \leq 1$, a graph G on n nodes is called β-connected if G contains a connected component on at least βn nodes.*

When we consider a sequence of graphs with different values of n, then the parameter β may depend on n. When this is the case, we write β_n-connectivity. Note that even if $\beta_n \to 1$, this is still weaker than full connectivity in the limit. For example, if $\beta_n = 1 - 1/\sqrt{n}$, then we have $\beta_n \to 1$, but each graph on n nodes can still have $n - \beta_n n = \sqrt{n}$ nodes that are not part of the largest connected component.

VI. ABSTRACT GEOMETRIC RANDOM GRAPH MODELS

Let us now introduce a model class that reflects a typical feature of geometric random graph models. This feature is that in geometric random graphs the primary random choice is picking random nodes from some domain and then the edges are already determined by some geometric property (typically some kind of distance) of the random nodes. We elevate this approach to an abstract level that includes many special cases of interest. The most general version of our abstract geometric model is built using the components detailed below.

A. Representing the Nodes: Node Variables

The nodes are represented by an infinite sequence $X_1, X_2, \ldots$ of random variables, called *node variables*. They take their values in an arbitrary (nonempty) set S, which is called the *domain* of the model. In most practical cases the domain is a simple subset of the Euclidean plane or of the 3-dimensional space. In general, however, S can be any abstract set from which we can choose random elements[3]. When we want to generate a random graph on n nodes, then we use the first n entries of the sequence, that is, $X_1, \ldots, X_n$ represent the nodes in G_n. It is important to note that we do not require the node variables to be independent.

B. Representing the Links: Edge Functions

We denote by $Y_{ij}^{(n)} \in \{0,1\}$ the indicator of the edge between nodes X_i, X_j in the random graph G_n. Since loops are not allowed, we always assume $i \neq j$, without repeating this condition each time. The (abstract) geometric nature of the model is expressed by the requirement that the random variables $Y_{ij}^{(n)}$ are determined by the nodes $X_1, \ldots, X_n$, possibly with additional independent randomization. Specifically, we assume that there exist functions $f_{ij}^{(n)}$, such that

$$Y_{ij}^{(n)} = f_{ij}^{(n)}(X_1, \ldots, X_n, \xi_{ij})$$

[3]To avoid mathematical complications that would only obscure the main message, we assume that all considered sets, functions etc. are measurable with respect to the used probability measures and all considered expected values exist. This is satisfied in in every practically relevant model.

where ξ_{ij} is a random variable that is uniformly distributed on $[0,1]$ and is independent of all the other defining random variables of the model (i.e., the node variables and all the other ξ_{kl} variables). Henceforth the role of ξ_{ij} is referred to as *independent randomization*[4]. The undirected nature of the graph is expressed by the requirement $Y_{ij}^{(n)} = Y_{ji}^{(n)}$, which can simply be enforced by computing all values for $i < j$ only and defining the $i > j$ case by exchanging i and j.

C. Restrictions

Regarding the abstract geometric random graph model in the presented very general form, it is clear that allowing *totally arbitrary* node variables and edge functions offers little hope for meaningful analysis. Therefore, next we introduce some restricting conditions. Later we are going to see that one has to make only surprisingly mild restrictions to meaningfully analyze the trade-off between node degrees and β-connectivity.

1) Locality: Up to now we allowed that an edge in G_n can depend on all the nodes, and the dependence expressed by the $f_{ij}^{(n)}$ functions can be arbitrary and different for each edge. To get a little closer to the usual geometric random graph model, let us introduce the following property, called *locality.* Informally, it restricts the dependence of an edge to its endpoints, in a homogeneous way, but still via an *arbitrary* function.

Definition 2: **(Locality)** *An abstract geometric random graph model is called* local*, if for every n and $i, j \leq n$ the existence of an edge between X_i, X_j depends only on these nodes. Moreover, the dependence is the same for every i, j, possibly with independent randomization. That is, there are functions $f^{(n)}$ such that the edge indicators are expressible as*

$$Y_{ij}^{(n)} = f^{(n)}(X_i, X_j, \xi_{ij})$$

where ξ_{ij} represents the independent randomization.

2) Name Invariance: Our second condition called *name invariance* refers to the joint distribution of nodes. If we allow totally arbitrary joint distribution, then it offers little chance for meaningful analysis. On the other hand, restricting ourselves only to independent, identically distributed (i.i.d.) node variables would exclude important cases, such as clustering. Therefore, we introduce a condition that allows more general than i.i.d. node variables, but still makes meaningful analysis possible. To introduce it, let us first recall a useful concept from probability theory, called exchangeability.

Definition 3: **(Exchangeable random variables)** *A sequence of random variables is called* exchangeable *if for any $k \geq 1$, it holds that if we select any k of the random variables, the joint distribution of the selected random variables depends only on k, but is independent of which particular k variables are selected, and in which order.*

Note that i.i.d. random variables are always exchangeable, but the converse generally does not hold, so exchangeable random variables form a larger family.

[4]Note that the specified distribution of ξ_{ij} does not impose a restriction, since the functions $f_{ij}^{(n)}$ are arbitrary.

Now let us introduce the condition that we use to restrict the arbitrary dependence of node variables.

Definition 4: **(Name invariance)** *An abstract geometric random graph model is called* name invariant*, if its node variables are exchangeable.*

We call it the *name invariance* of the model because it means the names (the indices) of the nodes are irrelevant in the sense that the joint probabilistic behavior of any fixed number of nodes is invariant to renaming (reindexing) the nodes. In particular, it also implies that the individual node variables are identically distributed, but they do not have to be independent.

Name invariance is naturally satisfied with the most frequently used random node choices, such as uniform independent random points in a planar domain, or a Poisson point process in the plane, or in higher dimension. We allow, however, much more complex node generation (over an arbitrary set!) since dependencies are not excluded by name invariance.

A simple example for a dependent, yet still name invariant, node generation process is a "clustered uniform" node generation. As an example, let S be a sphere in 3-dimensional space, i.e., the surface of a 3-dimensional ball. Let R be the radius of the ball. Let us first generate a pivot point Y uniformly at random from S. Then generate the nodes $X_1, X_2, \ldots$ uniformly at random and independently of each other from the neighborhood of radius $r \ll R$ of the random pivot point Y (on the sphere). It is directly implied by the construction that exchangeability holds. Moreover, any particular X_i will be uniformly distributed over the *entire* sphere, since Y is uniform over the sphere. On the other hand, the X_i are far from independent of each other, since they cluster around Y, forcing any two of them to be within distance $2r$. The setting can be generalized to applying several pivot points and non-uniform distributions, creating a more sophisticated clustering.

VII. SPECIFIC CLASSES WITHIN ABSTRACT GEOMETRIC RANDOM GRAPHS

Before turning to results, let us present some examples to show the usefulness and comprehensiveness of the generalization provided by our abstract geometric random graphs. These examples illustrate that most practically relevant models for wireless network topologies fit in the common generalization that we provided by introducing abstract geometric random graphs.

A. Geometric Random Graphs

All the usual geometric random graph models fit naturally in our general framework. For example, the base set S can be chosen as a unit disk or square in the plane or a unit ball or cube (or any other domain) in higher dimension. Let us choose i.i.d. points $X_1, X_2, \ldots$ from S, according to some probability distribution. Let $\rho(x, y)$ denote the distance of the points $x, y \in S$, it can be any distance function. Finally, let $r > 0$ be a radius (possibly depending on n). Then the edge function

$$f^{(n)}(X_i, X_j, \xi_{ij}) = \begin{cases} 1 & \text{if } \rho(X_i, X_j) \le r \\ 0 & \text{if } \rho(X_i, X_j) > r \end{cases} \tag{1}$$

defines a geometric random graph in the usual sense. (The independent randomization is not used here, so the edge function does not depend on ξ_{ij}.) It is clear that this includes all the usual geometric random graph models, allowing any metric space as the basis. Moreover, we can also use non-independent points, such as the "clustered uniform" example in the previous section, as long as the distribution is exchangeable.

B. Erdős-Rényi Random Graphs

The by now classical random graph model of Erdős and Rényi (see, e.g., [7], [32]), where each possible edge is included independently with some probability p is also included as a direct special case. We can set $S = \{1, \ldots, n\}$ and for $X_i, X_j \in S$

$$f^{(n)}(X_i, X_j, \xi_{ij}) = \begin{cases} 1 & \text{if } \xi_{ij} \le p \\ 0 & \text{if } \xi_{ij} > p \end{cases}$$

Note that now the edge function depends only on the independent randomization, so indeed each edge is included independently with probability p.

C. Geometric But Non-Metric Example: Battery Levels

In the geometric random graph models ρ satisfies the triangle inequality. This, however, cannot capture all situations that occur in ad hoc or sensor networks. As an example, assume the nodes are located in the plane. Let x_i, y_i be the coordinates of the i^{th} node. Furthermore, we also characterize a node with its battery level $E_i > 0$. E_i represents the remaining energy, assuming the node is not fully out of energy. Thus, a node is represented by a triple $X_i = (x_i, y_i, E_i)$. Let $d(E_i)$ be the distance over which a node can communicate, given its energy level E_i. (The function $d(E_i)$ can be derived from the physical characteristics of the node and from radio propagation conditions.) Now, a possible example of such a "distance" function is

$$\rho_1(X_i, X_j) = \frac{\sqrt{(x_i - x_j)^2 + (y_i - y_j)^2}}{\min\{d(E_i), d(E_j)\}}$$

If we take $r = 1$ and use the above ρ_1 function in (1), then it expresses the condition that a link exists if and only if its end nodes are at most at a distance that can be bridged by the energy levels of both nodes. Note that the above function ρ does not satisfy the triangle inequality, so it does not lead to a geometric random graph model in the usual sense. On the other hand, it still fits in our framework, as in (1) we did not require the triangle inequality to hold for ρ.

D. Another Non-Metric Example: Link Blocking

We can capture some features of traffic dependent network characteristics, as well. Let each node i be characterized by a triple $X_i = (x_i, y_i, \lambda_i)$, where x_i, y_i are planar coordinates and λ_i is the traffic demand of the node. Let B_{ij} be the blocking probability of the link (i, j), given that the link exists. We may compute B_{ij} as a function of λ_i, λ_j from some traffic model. For example, if we use Erlang's well known formula, assuming a capacity of C units on the link and its load is taken as the sum of its end nodes' traffic load $\lambda_i + \lambda_j$, then we obtain

$$B_{ij} = \frac{(\lambda_i + \lambda_j)^C / C!}{\sum_{i=0}^{C} (\lambda_i + \lambda_j)^i / i!}.$$

(Of course, we may use other traffic models, as well, this is just an example.) Now we can take the "distance" function

$$\rho_2(X_i, X_j) = \frac{1}{1 - B_{ij}} \sqrt{(x_i - x_j)^2 + (y_i - y_j)^2}$$

and use it in (1) with some radius r. We can observe that for small blocking probability ($B_{ij} \ll 1$) $\rho_2(X_i, X_j)$ will be approximately the same as the Euclidean distance. On the other hand, as B_{ij} approaches 1, the factor $\frac{1}{1-B_{ij}}$ tends to infinity and, therefore, high blocking probability makes the existence of the link in the model less likely, even if the physical distance is small. This example also violates the triangle inequality, so it is not a geometric random graph.

E. Log-Normal Shadowing

A typical phenomenon in the radio environment is *fading*. An example of fading is a relatively slow random fluctuation in the signal strength, which occurs even if the locations are fixed. Measurements show that this random variation can be accurately modeled by a log-normal distribution (see, e.g., [43]). Hence the name *log-normal shadowing*, which is widely used for this phenomenon. A way to capture it in our model is this. Let us characterize a node i by a triple $X_i = (x_i, y_i, \eta_i)$, where x_i, y_i represent a random position in the plane and each η_i is an infinite sequence of independent, log-normally distributed random variables:

$$\eta_i = (\eta_j^{(i)};\ j = i, i+1, i+2, \ldots).$$

The "distance" is defined as

$$\rho_3(X_i, X_j) = \eta_b^{(a)} \sqrt{(x_i - x_j)^2 + (y_i - y_j)^2}$$

where $a = \min\{i, j\}$ and $b = \max\{i, j\}$. (The reason for we need an infinite sequence of log-normal random variables is that this way we can have independent log-normal shadowing for every link.) This distance can express the fact that from the radio communication point of view we really perceive an "effective distance", which is a log-normally modulated random variant of the physical distance. Using this ρ_3 in (1) leads again to a random graph that is not geometric, as ρ does not satisfy the distance axioms.

F. Directional Antennas

We can also represent directional antennas in the model. As a simple example, let Y_i be the position of a node in the Euclidean plane, α_i be the angle (with respect to some fixed coordinate axis) at which its antenna is directed, and δ_i be the angular width of the beam (assuming an idealized directional antenna). Let us represent the node by the variable $X_i = (Y_i, \alpha_i, \delta_i)$. Let $S(X, \alpha, \delta)$ denote the planar angular sector pointed at X, with its axis of symmetry directed at α and of angular width δ. Further, let $||.||$ denote the Euclidean norm. Then we can introduce the following "distance":

$$\rho_4(X_i, X_j) = \begin{cases} ||Y_i - Y_j|| & \text{if } X_i \in S(X_j, \alpha_j, \delta_j) \text{ and} \\ & \quad X_j \in S(X_i, \alpha_i, \delta_i) \\ \infty & \text{otherwise} \end{cases}$$

If we use this function $\rho_4(X_i, X_j)$ in (1), then we get a model of a random ad hoc network topology with directional antennas.

G. Terrain Variations, Obstacles

Another example is to take into account uneven radio propagation characteristics due to terrain variations or propagation obstacles. For example, let us assume that the nodes operate in a frequency band in which only line of sight communication is possible (such as infrared). Then two nodes can only communicate if there is no obstacle covering them from each other. This feature can also be built into the model. Let X_i be the plane position of a node. Assume there exists a set $\mathcal{R} = \{R_1, R_2, \ldots\}$ of random obstacles in the area. Let $s(X_i, X_j)$ be the line segment connecting the points X_i, X_j, and let $L(X_i, X_j, \mathcal{R})$ be the "line of sight" function:

$$L(X_i, X_j, \mathcal{R}) = \begin{cases} 1 & \text{if } s(X_i, X_j) \cap R_k = \emptyset \ \ (\forall k) \\ \infty & \text{otherwise} \end{cases}$$

To express that only those nodes can communicate that are in line of sight of each other, let us introduce the "distance"

$$\rho_5(X_i, X_j) = ||X_i - X_j|| L(X_i, X_j, \mathcal{R}).$$

If this is used in (1), then we get a network topology model that can deal with radio propagation obstacles.

H. Combinations

The various conditions in the preceding examples can be combined into more complex models. For example, if we want that all the conditions expressed by the $\rho_1, \ldots, \rho_5$ functions are satisfied, then we can use

$$\rho(X_i, X_j) = \max\{\rho_1(X_i, X_j), \ldots, \rho_5(X_i, X_j)\}$$

in (1).

VIII. THRESHOLD FUNCTION FOR PARTIAL CONNECTIVITY

We define a concept that will characterize the trade-off between node degrees and the type of partial connectivity that we introduced as β-connectivity in Definition 1. For notational convenience, the set of nonnegative real numbers, extended with ∞, will be denoted by $\mathbf{R}_0^\infty$. Real functions are also extended to ∞ by $f(\infty) = \lim_{x\to\infty} f(x)$, whenever the limit exists (it will always exist in our cases). The value of β is always assumed to be in $[0, 1]$.

Let us first explain the threshold function concept informally. We define a threshold for β-connectivity, such that whenever β is above the threshold, then it is impossible to achieve a.a.s. β-connectivity for any model in the considered family of random graph models. On the other hand, if β is below the threshold, then this is not the case anymore, that is, there is at least one model in the family that achieves a.a.s β-connectivity with this β. Thus, the threshold separates the cases when a.a.s. β-connectivity is impossible, from the cases when it is possible. Since the threshold will depend on the expected average degree, we call it threshold function.

Now let us present the formal definition. Recall that the expected average degree in a random graph G_n is defined as $\overline{d}(n) = 2\mathrm{E}(e(G_n))/n$.

Definition 5: **(Threshold for β-connectivity)** *Let $\mathcal{F}$ be a family of random graph models. For any model $\mathcal{M} \in \mathcal{F}$ let G_n denote the random graph on n nodes generated by $\mathcal{M}$ and let*

$$D_{\mathcal{M}} = \limsup_{n\to\infty} \overline{d}(n)$$

be the limiting expected average degree. A function $f : \mathbf{R}_0^\infty \mapsto [0,1]$ is called a β-connectivity threshold function for $\mathcal{F}$ if the following two conditions are satisfied:

(i) *For any model $\mathcal{M} \in \mathcal{F}$ and for every $\beta > f(D_{\mathcal{M}})$*

$$\lim_{n\to\infty} \Pr(G_n \textit{is } \beta\textit{-connected}) < 1$$

holds, where G_n is generated by $\mathcal{M}$.

(ii) *If β is below the threshold, then (i) does not hold anymore, in the following sense. For every $\epsilon > 0$ there exists a model $\mathcal{M}_0 \in \mathcal{F}$ and a*

$$\beta \le f(D_{\mathcal{M}_0}) - \epsilon$$

such that

$$\lim_{n\to\infty} \Pr(G_n \textit{is } \beta\textit{-connected}) = 1$$

where G_n is generated from $\mathcal{M}_0$.

The importance of this concept is the following. If for a considered class $\mathcal{F}$ of random graph models we can find out what the corresponding β-connectivity threshold function is, then we can tell precisely what range of expected average degrees allow a.a.s. β-connectivity for a given β. Or, conversely, if we know the (asymptotic) expected average degree for a particular model $\mathcal{M}$ in the considered class, then we can decide what level of connectivity can be asymptotically achieved for this model.

IX. Computing the Threshold

Now we state the theorem that conveys the surprising message that for the very general class of abstract geometric random graph models we can still find the *precise* β-connectivity threshold function, if we assume that the models satisfy the conditions of locality and name invariance. The previously presented examples all satisfy these conditions, so they show that even with these restrictions we can still include many complex and practically important models. For the proof of the theorem, see [22].

Theorem 1: **(Threshold function for local and name invariant abstract geometric graphs)** *Let $\mathcal{F}$ be the family of local and name invariant abstract geometric random graph models. For any model $\mathcal{M} \in \mathcal{F}$ set*

$$D_{\mathcal{M}} = \limsup_{n\to\infty} \overline{d}(n).$$

Then the β-connectivity threshold function for $\mathcal{F}$ is

$$f(D_{\mathcal{M}}) = 1 - \mathrm{e}^{-D_{\mathcal{M}}}.$$

X. Consequences for Full Connectivity

It is worth mentioning that the definition of the threshold function and Theorem 1 directly imply that bounded expected average degrees in $\mathcal{F}$ exclude a.a.s. β_n-connectivity when $\beta_n \to 1$. As a result, a.a.s. full connectivity, which corresponds to $\beta = 1$, is also excluded. These claims are formally stated below, the proof is a direct application of Theorem 1.

Theorem 2: Let $\beta_n \to 1$ be an arbitrary sequence in $[0,1]$. Then for any local and name invariant abstract geometric random graph model $\mathcal{M}$ it holds that if $D_{\mathcal{M}} < \infty$, then the random graphs generated by $\mathcal{M}$ cannot be a.a.s. β_n-connected.

The interpretation of this result is that (asymptotically) the requirements of full connectivity and bounded degrees are incompatible, in the broad class of models we have considered.

At this point one may wonder whether there is *any* meaningful random graph model at all, in which a.a.s. full connectivity is possible, yet the node degrees remain bounded. Note that our results do not exclude this, since they only apply to local and name invariant abstract geometric random graph models. Although this class is quite comprehensive, it does not contain *all* meaningful models.

A nontrivial example worth mentioning here is the (random) Euclidean minimum spanning tree (MST). Let us choose n i.i.d. random points in the d-dimensional unit cube and view them as vertices of a complete graph, where each edge is assigned a weight that is equal to the (random) distance of its endpoints. Let T_n be the MST of this graph. Note that T_n is unique with probability 1. It is clear that T_n is connected, as, by definition, it is a spanning tree. Moreover, the following nontrivial fact is known: for every fixed dimension the maximum degree of the Euclidean MST is bounded by a constant, depending only on the dimension, but not on n (see, e.g., [47]). Thus, the model $\mathcal{M} = (T_n;\ n \in \mathbf{N})$ has the property that it is fully connected, yet its node degrees remain bounded.

It is clear that the Euclidean MST model is name invariant, since nothing depends on how the nodes are indexed. Does it then contradict to our results? No, because it does not satisfy the requirement of locality. Of course, the usual definition of the MST is indeed not local. But now our results imply that the non-locality is *unavoidable* in this case, as long as we want to preserve name invariance. In other words, it is impossible to define the Euclidean MST in a local way, such that, at the same time, the model is also name invariant.

Note that the fact that the Euclidean MST cannot be defined locally, with name invariance, is nontrivial. For example, one might try to define new node variables that contain enough information to decide for any pair whether an MST edge connects them, without looking at other nodes. A possibility is to introduce new node variables $Y_i = (X_i, \ldots, X_n, X_1, \ldots, X_{i-1})$, with edges that connect two such new nodes if their first components are connected by an MST edge, among the original X_i variables. In this way we can create a locally defined MST model, since one can decide from Y_i, Y_j alone, whether X_i, X_j are connected by an MST edge, as the information about *all* the original nodes are available in each of the new node variables. Thus, in the transformed domain we have a *local* model. The MST over the Y_i variables will be isomorphic (with probability 1) to the MST over the X_i node variables, so the new model is equivalent to the original, yet it is local. Then, however, the name invariance would be destroyed. Even though each Y_i individually has the same distribution (since it does not matter in what order the X_i are listed), but the joint distribution of Y_1 and Y_2 will not be the same as the joint distribution of Y_1 and Y_3. The reason is that the first coordinate of $Y_1 = (X_1, \ldots, X_n)$ is the same as the last coordinate of $Y_2 = (X_2, \ldots, X_n, X_1)$, but such a relationship does not hold between Y_1 and $Y_3 = (X_3, \ldots, X_n, X_1, X_2)$.

Generally, it follows from our results and from the aforementioned properties of the Euclidean MST that no matter how tricky local definition we invent for this random graph model, it cannot preserve name invariance. The fact that name invariance *excludes* the possibility of a local Euclidean MST definition appears to be hard to prove without our results.

XI. Solving the Motivating Example

In the motivating sensor network example, we observe that the model is described by a local and name invariant abstract geometric graph model, no matter what the unknown probability distributions are. The reason for locality is that once we choose the positions of two sensors, the probability that a link exists between them does not depend on the locations of other sensors. Although it does depend on the obstacles, but they are independent of the sensor positions, and block the links independently. Name invariance also holds in this example, as the sensor positions are i.i.d., which is a special case of an exchangeable system of random variables. The node degree bound of 3 yields $D_{\mathcal{M}} \leq 3$. By Theorem 1, the threshold function for β-connectivity in our case satisfies

$$f(D_{\mathcal{M}}) = 1 - \mathrm{e}^{-D_{\mathcal{M}}} \leq 1 - \mathrm{e}^{-3} \approx 0.95.$$

Thus, we can conclude that despite the very vague information about the system, we are still able to calculate that at least about 5% of the nodes cannot belong to the largest connected component.

Thus, our general result was able to easily come to a conclusion that would otherwise be rather hard to obtain without having further information.

XII. Conclusion

After briefly reviewing a number of models that are used to capture large network topologies, we focused on analyzing a notorious obstacle to wireless network scalability. This obstacle is the phenomenon that in geometrically generated random network topologies the requirement of asymptotic full connectivity results in infinitely growing expected node degrees. To address the issue, we have set up a general modeling framework, the abstract geometric random graph model. This contains many different possible random graph models as special cases. In this framework we can quantify the precise trade-off between the expected node degrees and the fraction of nodes that can belong to the largest connected component.

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Discrete Stochastic Optimization Based Parameter Estimation for Modeling Partially Observed WLAN Spectrum Activity

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***Abstract*—Modeling and parameter estimation of spectrum usage in the ISM band would allow the competing networking technologies to adjust their medium access control accordingly, leading to the more efficient use of the shared spectrum. In this paper we address the problem of WLAN spectrum activity model parameter estimation. We propose a solution based on discrete stochastic optimization, that allows accurate spectrum activity modeling and can be implemented even in wireless sensor nodes with limited computational and energy resources.**

***Index Terms*—cognitive networks, WLAN spectrum activity, discrete stochastic optimization**

I. Introduction

EMERGING wireless technologies for local and personal area communication all use the open Industrial, Scientific and Medical (ISM) band. While the variety of introduced solutions increases, the protocol stacks are usually optimized for a given application area, and at the same time assume the exclusive use of the spectrum space. However, most of the time the different technologies coexist, and communication efficiency and performance guarantees can only be achieved, if the networks have cognitive capabilities [1], that is, they are aware of each other and optimize their transmission parameters and communication protocols accordingly.

Key technologies operating in the ISM band are the IEEE 802.11 wireless local area networks (WLANs). As WLAN carrier sensing is designed to detect WLAN signals, it is *blind* towards the low power, narrow band WSN transmissions. Consequently, if the WSN does not adjust itself to the WLAN operation, it will experience harmful interference from the WLAN, while the WLAN itself is not affected significantly by the narrow band low power WSN interferers.

Previous work in the area of cognitive WSNs includes proposals for novel carrier sensing and medium access control, and the characterization of the channel usage in WLAN cells. In [2] the interfering technology is identified based on spectral signature. In the case of WLAN interferers, the sensors force the WLAN to back off by sending short, high power jamming signals. The POMDP framework [3] introduces the concept of partial channel knowledge and proposes optimal sensing and channel access strategies considering a Markovian channel occupancy model. A Markovian model, however, may lead to suboptimal WSN operation, and therefore several works deal with a more accurate channel characterization, considering sub-geometric [4], hyper-exponential [5] and Pareto [6] idle time distributions.

In [7][8] it is recognized, that the characterization of the idle time can lead to more efficient cognitive access control, if it captures the two basic sources of WLAN inactivity, the short, almost uniformly distributed contention windows and the long, heavy-tailed white space periods, when the WLAN users are inactive. We follow this approach in our previous work, where we propose cognitive medium access control and next hop selection for the WSN [9], given the known WLAN channel idle time distribution. In [10] we define the *Local View* model of WLAN channel activity that extends the solution of [7] and takes into account the limited detection range of the WSN nodes, and propose computationally efficient ways to estimate the model parameters based on time limited continuous sensing at the sensors.

In this paper we provide a deep analysis of the Local View parameter estimation based on discrete stochastic optimization. We follow the approach presented in [11], show that the algorithm converges almost surely to the optimal parameter set, and evaluate how the size of the state space, the size of the sample set and the number of iterations affect the estimation accuracy.

The rest of the paper is organized as follows. Section II defines the considered networking scenario along with the WLAN channel activity models and formulates the parameter estimation as an optimization problem. In Section III we give an overview of the discrete stochastic optimization algorithm proposed in [11]. In Section IV we show that the algorithm converges in the case of the considered parameter estimation problem and in V we evaluate the performance of the algorithm under practical constraints. We conclude the paper in Section VI.

II. WLAN Idle Time Modeling

We consider an IEEE 802.15.4 compliant WSN operating in the transmission area of an IEEE 802.11 WLAN. The transmission power of the WLAN terminals is orders of magnitude higher than that of the coexisting WSN, and the WLAN terminals are *blind* towards the WSN transmissions. The protocol stack of the energy constrained WSN is enhanced by cognitive functionality to optimize the WSN operation. To perform cognitive control, the WSN needs to know the WLAN channel occupancy distribution. For this, the sensors

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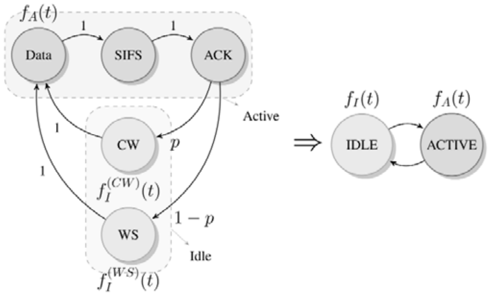

Fig. 1. The Global View model with all channel states and the reduced two-state semi-Markovian model.

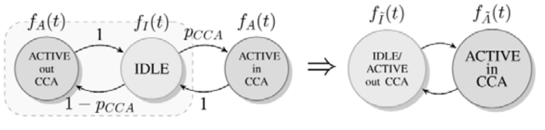

Fig. 2. The 3-state semi-Markovian chain and its 2-state equivalent for the Local View channel activity modeling.

perform continuous sensing and collect samples of busy and idle WLAN period lengths. The sensing is based on the usual Clear Channel Assessment (CCA) process with energy detection, resulting in a limited sensing range.

According to [7][8], the *Global View* of WLAN channel occupancy can be modeled by a semi-Markovian system of Active and Idle periods [12]. Figure 1 depicts all the states of the WLAN channel and their merging into a two-state semi-Markovian chain. The states of Data, SIFS and ACK transmission are grouped together into a single *Active* state, while the states that represent the WLAN Contention Window period (CW) and the WLAN White Space (WS) due to user inactivity are merged into a single *Idle* state. The sojourn times in the Active state can be modeled by the *uniform* distribution $f_A(t)$ within $[\alpha_{\text{ON}}, \beta_{\text{ON}}]$, which denote the minimum and maximum frame-in-the air duration, respectively. The idle period distribution, $f_I(t)$, is a *mixture* distribution with a weight p, that is $f_I(t) \triangleq p \cdot f_I^{CW}(t) + (1-p) \cdot f_I^{WS}(t)$. $f_I^{CW}(t)$ is the distribution of the CW periods, and can be modeled with a *uniform* distribution within $[0, \alpha_{\text{BK}}]$, where α_{BK} denotes the maximum WLAN back-off duration, given by the WLAN specification. The WS periods, however, exhibit a heavy-tailed behavior, and their distribution $f_I^{WS}(t)$ is well approximated by a zero-location *generalized Pareto* distribution with parameters (ξ, σ).

Thus, the distribution of the sojourn time in the Idle state, $f_I(t)$, is given as:

$$f_I(t) \triangleq \begin{cases} p \cdot \dfrac{1}{\alpha_{\text{BK}}} + (1-p) \cdot \dfrac{1}{\sigma}\left(1 + \xi \dfrac{t}{\sigma}\right)^{\left(-\frac{1}{\xi}-1\right)} & t \le \alpha_{\text{BK}} \\ (1-p)\dfrac{1}{\sigma}\left(1+\xi\dfrac{t}{\sigma}\right)^{\left(-\frac{1}{\xi}-1\right)} & t > \alpha_{\text{BK}} \end{cases}$$

This Global View, however, is not fully observable at the individual sensors, that can detect WLAN transmissions only within a given detection range. Therefore, in [10] we define the *Local View*, that describes the WLAN channel occupancy as seen by an individual sensor. Assuming that consecutive WLAN transmissions are not correlated, we introduce a 3-state semi-Markovian system (Figure 2), distinguishing between detected, and un-detected WLAN activity, that occurs with probabilities p_{CCA} and $(1 - p_{\text{CCA}})$, respectively. To model the *observable* sojourn time distributions $f_{\tilde{A}}(t)$ and $f_{\tilde{I}}(t)$ we define the 2-state *Local View* model by merging the states at which the sensor detects an idle channel. It holds that $f_{\tilde{A}}(t) = f_A(t)$, but $f_{\tilde{I}}(t) \neq f_I(t), \quad \forall p_{\text{CCA}} < 1$.

Our objective is to estimate the parameters of $f_A(t)$ and $f_I(t)$ and the *observable load*, p_{CCA}, from a set of samples of $f_{\tilde{A}}(t)$ and $f_{\tilde{I}}(t)$ obtained through channel sensing.

As the active period distribution, $f_{\tilde{A}}(t)$, is uniform, its parameters α_{ON} and β_{ON} are estimated by the lowest and the largest measured active period according to Maximum Likelihood Estimation (MLE). The estimation of the rest of the parameters is more difficult. An idle channel period observed by an arbitrary sensor consists of a random number of WLAN "cycles", that is, consecutive idle and un-detected active periods, followed by an additional idle period. The locally observable idle period distribution, $f_{\tilde{I}}(t)$, is, therefore, a function of the idle and active time distributions $f_I(t)$ and $f_A(t)$, and of the observable load, p_{CCA}, and can not be expressed in a closed form, even if $f_A(t)$ and $f_I^{CW}(t)$ are known.

As we show in [10], closed form expression exists in the Laplace domain and, therefore, we propose to estimate the parameters of $f_{\tilde{I}}(t)$ in the Laplace domain. Since according to the semi-Markovian Local View model the number of consecutive WLAN cycles is geometrically distributed, the Laplace Transform (LT) of $f_{\tilde{I}}(t)$ obtains the following form:

$$f_{\tilde{I}}^*(s) = f_I^*(s)\frac{p_{\text{CCA}}}{1-(1-p_{\text{CCA}})f_I^*(s)f_A^*(s)}, \tag{1}$$

where $f_I^*(s), f_A^*(s)$ denote the LT of $f_I(t), f_A(t)$, respectively.

III. An Algorithm for Discrete Stochastic Optimization for Parameter Estimation

In this Section we review the algorithm for stochastic optimization introduced in [11], that we use to estimate the parameters of the Local View model. First we define the necessary notation and then we give the stochastic optimization algorithm, along with the constraint on convergence.

Let us define by $\boldsymbol{\mathcal{K}} \triangleq \{\mathcal{K}_1, \mathcal{K}_2, ..., \mathcal{K}_K\}$ the discrete space of the different alternatives. The number of discrete states, $K = |\boldsymbol{\mathcal{K}}|$, is finite. The optimization problem we aim at solving is of the following form:

$$\mathcal{K}^* = \arg\min_{\mathcal{K}_n \in \boldsymbol{\mathcal{K}}} \{c(n) = E[X_{\mathcal{K}_n}]\}. \tag{2}$$

That is, the function $c(n)$ can not be evaluated analytically and needs to be estimated through a sequence of random samples $\{X_{\mathcal{K}_n}\}$. We denote by:

$$\boldsymbol{\mathcal{L}} = \{\mathcal{L}_1, ..., \mathcal{L}_L\} \subset \boldsymbol{\mathcal{K}} \tag{3}$$

the set of *global* minimizers of the function c, that is:

$$\begin{aligned} &\forall \mathcal{L}_i \in \boldsymbol{\mathcal{L}}, \mathcal{K}_n \in \boldsymbol{\mathcal{K}} \setminus \boldsymbol{\mathcal{L}},\ c(\mathcal{L}_i) < c(\mathcal{K}_n) \text{ and} \\ &\forall i, j = 1, 2, ..., L,\ c(\mathcal{L}_i) = c(\mathcal{L}_j). \end{aligned} \tag{4}$$

In the following we give the original stochastic optimization algorithm as it is proposed in [11] (Algorithm 1). The search process starts from an arbitrary state, $\mathcal{K}_i$. In each iteration step, m, it selects a new state $\mathcal{K}_j$ uniformly at random and obtains the observation of a random variable $Z_{l_m}^{\mathcal{K}_i \to \mathcal{K}_j}$ to compare the two states. $Z_{l_m}^{\mathcal{K}_i \to \mathcal{K}_j}$ is a function of the random variables $\{X_{\mathcal{K}_i}\}_{l_m}, \{X_{\mathcal{K}_j}\}_{l_m}$. Thus, its value can depend on the two states, $\mathcal{K}_i, \mathcal{K}_j$, and on l_m, which is a function of the iteration step m. The algorithm moves to the new state if $Z_{l_m}^{\mathcal{K}_i \to \mathcal{K}_j} > 0$.

Let denote $\mathcal{K}_m$ the state after iteration m and $Q_m(\mathcal{K}_n)$ the "popularity" of state $\mathcal{K}_n \in \boldsymbol{\mathcal{K}}$, i.e. the number of times the algorithm has visited state $\mathcal{K}_n$ until iteration m. The output of the algorithm, $\mathcal{K}^*$, is chosen as the most visited state.

Algorithm 1 A global search for discrete stochastic optimization [11].

Step 0:

Select a starting point $\mathcal{K}_0 \in \boldsymbol{\mathcal{K}}$.
$Q_0(\mathcal{K}_0) \leftarrow 1$ and $Q_0(\mathcal{K}_n) \leftarrow 0, \forall \mathcal{K}_n \in \boldsymbol{\mathcal{K}}, \mathcal{K}_n \neq \mathcal{K}_0$.
$m \leftarrow 0$ and $\mathcal{K}_m^* \leftarrow \mathcal{K}_0$. Go to Step 1.

Step 1:

Generate a uniform random variable $\mathcal{J}_m$ such that for all $\mathcal{K}_n \in \boldsymbol{\mathcal{K}}, \mathcal{K}_n \neq \mathcal{K}_m$, $\mathcal{J}_m \leftarrow \mathcal{K}_n$ with probability $\frac{1}{K-1}$. Go to Step 2.

Step 2:

Generate an observation R_m of $Z_{l_m}^{\mathcal{K}_m \to \mathcal{J}_m}$.
if $R_m > 0$ **then**
  $\mathcal{K}_{m+1} \leftarrow \mathcal{J}_m$.
else
  $\mathcal{K}_{m+1} \leftarrow \mathcal{K}_m$.
end if Go to Step 3.

Step 3:

$m \leftarrow m+1$, $Q_m(\mathcal{K}_m) \leftarrow Q_{m-1}(\mathcal{K}_m)+1$ and $Q_m(\mathcal{K}_n) \leftarrow Q_{m-1}(\mathcal{K}_n)$ for all $\mathcal{K}_n \neq \mathcal{K}_m$.
if $Q_m(\mathcal{K}_m) > Q_m(\mathcal{K}_{m-1}^*)$ **then**
  $\mathcal{K}_m^* \leftarrow \mathcal{K}_m$.
else
  $\mathcal{K}_m^* \leftarrow \mathcal{K}_{m-1}^*$
end if Go to Step 1.

It is shown in [11] that the algorithm converges almost surely to a minimizer, i.e. a member of $\mathcal{L}$, after sufficiently large number of iterations, if the following conditions hold:

Condition 1. *For each $\mathcal{K}_i, \mathcal{K}_j \in \boldsymbol{\mathcal{K}}$ and $l \in \mathbb{N}$, there exists a random variable $Z_l^{(\mathcal{K}_i \to \mathcal{K}_j)}$ such that the limit $\lim_{l\to\infty} P\{Z_l^{(\mathcal{K}_i \to \mathcal{K}_j)} > 0\}$ exists for all $\mathcal{K}_i, \mathcal{K}_j \in \boldsymbol{\mathcal{K}}$ and for all $\mathcal{K}_i \in \mathcal{L}, \mathcal{K}_j \notin \mathcal{L}, \mathcal{K}_n \neq \mathcal{K}_i, \mathcal{K}_j$, and $l \in \mathbb{N}$,*

$$\lim_{l\to\infty} P\{Z_l^{(\mathcal{K}_j \to \mathcal{K}_i)} > 0\} > \lim_{l\to\infty} P\{Z_l^{(\mathcal{K}_i \to \mathcal{K}_j)} > 0\}, \quad (5)$$

$$\lim_{l\to\infty} P\{Z_l^{(\mathcal{K}_n \to \mathcal{K}_i)} > 0\} \geq \lim_{l\to\infty} P\{Z_l^{(\mathcal{K}_n \to \mathcal{K}_j)} > 0\}, \quad (6)$$

$$\lim_{l\to\infty} P\{Z_l^{(\mathcal{K}_i \to \mathcal{K}_n)} \leq 0\} \geq \lim_{l\to\infty} P\{Z_l^{(\mathcal{K}_j \to \mathcal{K}_n)} \leq 0\}. \quad (7)$$

Condition 2. *$\{l_m\}$ is a sequence of positive integers such that $l_m \to \infty$ as $m \to \infty$.*

Condition 3. *The Markov matrix $\mathcal{P}$ defined in the following equations is irreducible.*

$$\mathcal{P}(\mathcal{K}_i, \mathcal{K}_j) = \tfrac{1}{K-1} \lim_{l\to\infty} P\{Z_l^{(\mathcal{K}_i \to \mathcal{K}_j)} > 0\}$$
$$\forall \mathcal{K}_i, \mathcal{K}_j \in \boldsymbol{\mathcal{K}}, \mathcal{K}_i \neq \mathcal{K}_j,$$
$$\mathcal{P}(\mathcal{K}_i, \mathcal{K}_i) = \tfrac{1}{K-1} \sum_{\mathcal{K}_j \in \boldsymbol{\mathcal{K}} \setminus \{\mathcal{K}_i\}} \lim_{l\to\infty} P\{Z_l^{(\mathcal{K}_i \to \mathcal{K}_j)} \leq 0\}$$
$$\forall \mathcal{K}_i \in \boldsymbol{\mathcal{K}}.$$

IV. Local View Parameter Estimation

A. The Estimation Process

We apply Algorithm 1 to estimate the parameters of $f_{\bar{I}}(t)$. We discretize the model parameters ξ, σ and p within the reasonable intervals, and define the state $\mathcal{K}_n$ as the set of model parameters:

$$\mathcal{K}_n \triangleq (\xi_n, \sigma_n, p_n).$$

Since the value of these model parameters give, together with p_{CCA}, the estimated average observable idle period length, we do not include p_{CCA} directly in the algorithm state space, but determine it through Moment Evaluation (ME), considering the sample mean of the measured observable idle period lengths, μ, and the rest of the parameters, i.e.

$$p_{\mathrm{CCA}} = \frac{\frac{p\alpha_{\mathrm{BK}}}{2} + \frac{(1-p)\sigma}{1-\xi} + \frac{\alpha_{\mathrm{ON}}+\beta_{\mathrm{ON}}}{2}}{\bar{\mu} + \frac{\alpha_{\mathrm{ON}}+\beta_{\mathrm{ON}}}{2}}.$$

We would like to determine the optimal state, $\mathcal{K}^* \in \boldsymbol{\mathcal{K}}$, that is the optimal model parameter set $\mathcal{K}^* \triangleq (\xi^*, \sigma^*, p^*)$, that minimizes the Mean Square Error (MSE) between the Laplace transform of the idle distribution, $f_{\bar{I}}^*(s)$, and the LT given by the system state, $f_{\bar{I}}^*(s; \mathcal{K}_n)$, over $\mathcal{S} = \{s_1, s_2, \ldots s_S\}$, the finite discrete subset of the s-domain, that is,

$$(\xi^*, \sigma^*, p^*) = \arg\min_{\mathcal{K}_n \in \boldsymbol{\mathcal{K}}} \frac{1}{S} \sum_{k=0}^{S} (f_{\bar{I}}^*(s_k) - f_{\bar{I}}^*(s_k; \mathcal{K}_n))^2. \quad (8)$$

As $f_{\bar{I}}^*(s)$ is not known, it needs to be evaluated through the idle period samples obtained by channel sensing.

To ensure fast parameter estimation, we propose to run the estimation process, that is, Algorithm 1 parallel to the channel sensing. That is, in each iteration step, m, n_m new idle period samples are integrated in the empirical LT. The total number of samples integrated up to iteration m is $N_m = \sum_{k=0}^{m} n_m$. We define the *empirical* LT, $f_{\bar{I}e}^*(s; N_m)$, of the observable idle time distribution directly from a set of N_m measured idle period samples, $(t_1, ..., t_{N_m})$ as:

$$f_{\bar{I}e}^*(s; N_m) = \frac{1}{N_m} \sum_{i=1}^{N_m} e^{-st_i}. \quad (9)$$

Comparing the general expression in (2) and the Mean Square Error minimization problem in (8) we have:

$$X_{\mathcal{K}_n} = \mathrm{MSE}_n^{(N)} = \frac{1}{S} \sum_{k=0}^{S} \left[f_{\bar{I}e}^*(s_k; N) - f_{\bar{I}}^*(s_k; \mathcal{K}_n)\right]^2, \forall \mathcal{K}_n, \quad (10)$$

where $\text{MSE}_n^{(N)}$ denotes the MSE calculated with the N samples, and, consequently,

$$c(n) = E\left[\frac{1}{S}\sum_{k=0}^{S}[(f_{Ie}^*(s;N) - f_I^*(s;\mathcal{K}_n))^2]\right] = \\ = \frac{1}{S}\sum_{k=0}^{S} E[(f_{Ie}^*(s;N) - f_I^*(s;\mathcal{K}_n))^2]. \tag{11}$$

Accordingly, for Algorithm 1 we select $l_m = N_m$ and define:

$$Z_{l_m}^{(\mathcal{K}_i\to\mathcal{K}_j)} \triangleq X_{\mathcal{K}_i} - X_{\mathcal{K}_j} = \text{MSE}_{\mathcal{K}_i}^{(l_m)} - \text{MSE}_{\mathcal{K}_j}^{(l_m)},\ \forall \mathcal{K}_i, \mathcal{K}_j \in \boldsymbol{\mathcal{K}}. \tag{12}$$

That is, the observation of variable $Z_{l_m}^{\mathcal{K}_i\to\mathcal{K}_j}$, generated at step m in our algorithm, is the difference between the mean square errors evaluated at states $\mathcal{K}_i, \mathcal{K}_j$ and over N_m total idle period samples. The process moves to the new state $\mathcal{K}_j$ if the MSE is decreased this way.

B. On the Convergence of the Estimation Process

To prove that the proposed parameter estimation algorithm solves the optimization problem in (8), we proceed as follows. The proof that Condition 2 holds is trivial; since $\{l_m\}$ defines the number of samples that are integrated in the empirical LT calculation until step m, it is a sequence of integers that tends to ∞ as $m \to \infty$. With Lemma 1 we prove that $f_{\tilde{I}e}^*(s;N)$ is an unbiased estimator of $f_{\tilde{I}}^*(s)$, and converges to $f_{\tilde{I}}^*(s)$ as $N \to \infty$. Based on Lemma 1, we prove with Corollary 1 that the minimization of $c(n)$ solves the original problem in (8). Lemma 2 proves that the particular selection of the random variable $Z_{l_m}^{(\mathcal{K}_i\to\mathcal{K}_j)}, \forall \mathcal{K}_i, \mathcal{K}_j \in \boldsymbol{\mathcal{K}}$ satisfies Condition 1. Lemma 3 shows that in our problem Algorithm 1 converges to the optimal state, bypassing the requirement for Condition 3 to hold.

Lemma 1. *The empirical Laplace Transform as a function of N i.i.d. samples $\{t_1, ..., t_N\}$ can be approximated as*

$$f_{Ie}^*(s;N) = \frac{1}{N}\sum_{l=1}^{N} e^{-st_l}.$$

and is an unbiased estimator of $f^*(s)$, converging to $f^*(s)$ as $N \to \infty$.

Proof: We, first, generate the empirical distribution function, $F_e(t;N)$ based on the N collected time period samples, $T = \{t_1, ..., t_N\}$,

$$F_e(t;N) = \frac{\#\text{samples in } T \le t}{N}.$$

It is known that $F_e(t;N)$ converges almost surely to the actual CDF, $F(t)$, as $N \to \infty$, based on the strong law of large numbers. In addition, $F_e(t;N)$ is an unbiased estimator of $F(t)$, i.e. $E[F_e(t;N)] = F(t)$. $F_e(t_l) - F_e(t_{l-1})$ is, then, an unbiased estimator for $P\{t \in (t_{l-1}, t_l)\} = F(t_l) - F(t_{l-1})$, $l = 1, 2, ..., N$. We define the empirical density function, $f_e(t;N)$ being non-zero only on the set T, as follows:

$$f_e(t;N) = \sum_{l=1}^{N}[F_e(t_l;N) - F_e(t_{l-1};N)]\,\delta(t - t_l),$$

where $\delta(t)$ is the Dirac function and by convention $t_0 = 0, F_e(t_0) = 0$. Clearly,

$$\lim_{N\to\infty, t_l - t_{l-1}\to dt_l} F_e(t_l;N) - F_e(t_{l-1};N) = f(t_l)dt_l$$

and so

$$\lim_{N\to\infty}\sum_{l=1}^{N}[F_e(t_l;N) - F_e(t_{l-1};N)]\,\delta(t - t_l) = \\ = \int_{t_l} f(t_l)\delta(t - t_l)dt_l = f(t),$$

consequently $f_e(t;N)$ converges to the actual pdf. The empirical Laplace Transform is defined as

$$f_e^*(s;N) \triangleq \int_0^\infty f_e(t;N)e^{-st}dt.$$

Since $F_e(t_l) - F_e(t_{l-1}) = 1/N$, the above becomes

$$f_e^*(s;N) = \int_0^\infty \sum_{l=1}^{N}\frac{1}{N}\delta(t - t_l)e^{-st}dt = \\ = \sum_{l=1}^{N}\frac{1}{N}\int_0^\infty \delta(t - t_l)e^{-st}dt = \frac{1}{N}\sum_{l=1}^{N} e^{-st_l}.$$

The convergence of $f^*(s;N)$ is ensured due to the convergence of $f_e(t;N)$. Finally,

$$E[f_e^*(s;N)] = \sum_{l=1}^{N}\frac{1}{N}E[e^{-st_l}] = E[e^{-st}] = f^*(s), \tag{13}$$

so $f^*(s;N)$ is an unbiased estimator of the LT transform. ■

Corollary 1. *The minimization of $c(n)$ in* (11) *solves the original problem in* (8).

Proof: We have:

$$c(n) = \frac{1}{S}\sum_{k=0}^{S} E\left[\left(f_{\tilde{I}e}^*(s_k;N) - f_{\tilde{I}}^*(s_k;\mathcal{K}_n)\right)^2\right] =$$

$$= \frac{1}{S}\sum_{k=0}^{S} E\left[\left(f_{\tilde{I}e}^*(s_k;N)\right)^2 - 2f_{\tilde{I}e}^*(s_k;N)f_{\tilde{I}}^*(s_k;\mathcal{K}_n)\right] +$$

$$+ \left(f_{\tilde{I}}^*(s_k;\mathcal{K}_n)\right)^2 =$$

$$= \frac{1}{S}\sum_{k=0}^{S} E\left[\left(f_{\tilde{I}e}^*(s_k;N)\right)^2\right] - 2f_{\tilde{I}}^*(s_k)f_{\tilde{I}}^*(s_k;\mathcal{K}_n) +$$

$$+ \left(f_{\tilde{I}}^*(s_k;\mathcal{K}_n)\right)^2.$$

For $N \to \infty$ it holds from Lemma 1 that $\text{Var}\left[f_{\tilde{I}e}^*(s_k;N)\right] = 0$, and consequently, $E\left[\left(f_{\tilde{I}e}^*(s_k;N)\right)^2\right] = E\left[f_{\tilde{I}e}^*(s_k;N)\right]^2$, so $c(n)$ converges to $\frac{1}{S}\sum_{k=0}^{S}(f_{\tilde{I}}^*(s_k) - f_{\tilde{I}}^*(s_k;\mathcal{K}_n))^2$. ■

Let us now prove, that Condition 1 is satisfied.

Lemma 2. *Let us select a random variable $Z_{l_m}^{(\mathcal{K}_i\to\mathcal{K}_j)}$ as follows:*

$$Z_{l_m}^{(\mathcal{K}_i\to\mathcal{K}_j)} = MSE_{\mathcal{K}_i}^{(l_m)} - MSE_{\mathcal{K}_j}^{(l_m)}.$$

The variable $Z_{l_m}^{(\mathcal{K}_i\to\mathcal{K}_j)}$ satisfies Condition 1.

Proof: We start with showing that (5) is satisfied. Let $\mathcal{K}_i \in \mathcal{L}$, $\mathcal{K}_j \notin \mathcal{L}$, so that

$$\frac{1}{S}\sum_{k=0}^{S}(f_I^*(s_k) - f_I^*(s_k;\mathcal{K}_i))^2 < \frac{1}{S}\sum_{k=0}^{S}(f_I^*(s_k) - f_I^*(s_k;\mathcal{K}_j))^2 \tag{14}$$

We show, first, by direct computation that the mean of $Z_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)}$, defined in (12) is negative.

$$\begin{aligned}
&E[Z_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)}] = \\
&= E[\mathrm{MSE}_{\mathcal{K}_i}^{(l_m)} - \mathrm{MSE}_{\mathcal{K}_j}^{(l_m)}] = \\
&= E[\tfrac{1}{S}\textstyle\sum_{k=0}^{S}(f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_i))^2 - \\
&\quad -\tfrac{1}{S}\textstyle\sum_{k=0}^{S}(f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_j))^2] \\
&= \tfrac{1}{S}E[\textstyle\sum_{k=0}^{S}(f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_i))^2 - \\
&\quad -(f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_j))^2] \\
&= \tfrac{1}{S}E[\textstyle\sum_{k=0}^{S}(2f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_i) - f_I^*(s_k;\mathcal{K}_j)) \cdot \\
&\quad \cdot (f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i))] \\
&= \tfrac{1}{S}\textstyle\sum_{k=0}^{S}(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i)) \cdot \\
&\quad \cdot E[2f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_i) - f_I^*(s_k;\mathcal{K}_j)] \\
&\overset{(13)}{=} \tfrac{1}{S}\textstyle\sum_{k=0}^{S}(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i)) \cdot \\
&\quad \cdot (2f_I^*(s_k) - f_I^*(s_k;\mathcal{K}_i) - f_I^*(s_k;\mathcal{K}_j)) \\
&= \tfrac{1}{S}\textstyle\sum_{k=0}^{S}(f_I^*(s_k) - f_I^*(s_k;\mathcal{K}_i))^2 - (f_I^*(s_k) - f_I^*(s_k;\mathcal{K}_j))^2 \\
&\overset{(14)}{<} 0.
\end{aligned}$$

We now show that $\lim_{l_m \to \infty} Z_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)}$ is symmetric around its mean. We write:

$$\begin{aligned}
&Z_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)} = \\
&= \tfrac{1}{S}\textstyle\sum_{k=0}^{S}(f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_i))^2 - \\
&\quad -\tfrac{1}{S}\textstyle\sum_{k=0}^{S}(f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_j))^2 \\
&= \tfrac{1}{S}\textstyle\sum_{k=0}^{S}(2f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_i) - f_I^*(s_k;\mathcal{K}_j)) \cdot \\
&\quad \cdot (f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i)) \\
&= \tfrac{1}{S}\textstyle\sum_{k=0}^{S}2f_{Ie}^*(s_k;l_m)(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i)) - \\
&\quad -\tfrac{1}{S}\textstyle\sum_{k=0}^{S}(f_I^*(s_k;\mathcal{K}_j) + f_I^*(s_k;\mathcal{K}_i)) \cdot \\
&\quad \cdot (f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i)).
\end{aligned}$$

The second term is deterministic and, thus, excluded from the calculations. We concentrate on the first term.

$$W_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)} = \frac{1}{S}\sum_{k=0}^{S} 2f_{Ie}^*(s_k;l_m)(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i)) \tag{15}$$

As shown above the empirical LT is generated as $f_{Ie}^*(s_k;l_m) = \frac{\sum_{l=1}^{l_m} e^{-s_k t_l}}{l_m}$. We rewrite (15) as

$$\begin{aligned}
&W_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)} = \\
&= \tfrac{1}{S}\textstyle\sum_{k=0}^{S}2f_{Ie}^*(s_k;l_m)(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i)) \\
&= \tfrac{1}{S}\tfrac{1}{l_m}\textstyle\sum_{k=0}^{S}\sum_{l=1}^{l_m}2e^{-s_k t_l}(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i)) \\
&= \tfrac{1}{S}\textstyle\sum_{l=1}^{l_m}\left[\tfrac{1}{l_m}\sum_{k=0}^{S}2e^{-s_k t_l}(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i))\right].
\end{aligned}$$

Since the variables t_l are i.i.d. so are the variables

$$g_l^{ji} = \sum_{k=0}^{S}\frac{2}{S}e^{-s_k t_l}(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_i)). \tag{16}$$

As a result, the variable $\frac{1}{l_m}\sum_{l=1}^{l_m} g_l^{ji}$ approaches a Gaussian distribution, due to the Central Limit Theorem, and becomes, thus, symmetric around its mean value. Consequently, $\lim_{l_m \to \infty} W_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)}$ is symmetric around its mean, and so does $\lim_{l_m \to \infty} Z_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)}$. Since, additionally, $E\left[Z_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)}\right] < 0$, it follows that

$$P\{Z_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)} < 0\} > P\{Z_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)} > 0\}. \tag{17}$$

From the last equation, along with $P\{Z_{l_m}^{(\mathcal{K}_i \to \mathcal{K}_j)} < 0\} = P\{Z_{l_m}^{(\mathcal{K}_j \to \mathcal{K}_i)} > 0\}$, follows Eq. (5).

We proceed with showing that (6) is satisfied. For that, we need to show, first that the variance of $Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_j)}$ is finite. By direct computation we obtain:

$$\begin{aligned}
&\mathrm{Var}\left[Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_j)}\right] \\
&= \mathrm{Var}\left[\mathrm{MSE}_{\mathcal{K}_n}^{(l_m)} - \mathrm{MSE}_{\mathcal{K}_j}^{(l_m)}\right] \\
&= \mathrm{Var}[\tfrac{1}{S}\textstyle\sum_{k=0}^{S}(f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_n))^2 \\
&\quad -(f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_j))^2] \\
&= \mathrm{Var}[\tfrac{1}{S}\textstyle\sum_{k=0}^{S}(2f_{Ie}^*(s_k;l_m) - f_I^*(s_k;\mathcal{K}_n) - f_I^*(s_k;\mathcal{K}_j)) \cdot \\
&\quad \cdot (f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_n))]
\end{aligned}$$

We neglect the deterministic term in $Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_j)}$, resulting in the expression

$$\begin{aligned}
&\mathrm{Var}\left[\tfrac{1}{S}\textstyle\sum_{k=0}^{S}2f_{Ie}^*(s_k;l_m)(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_n))\right] \\
&= \mathrm{Var}\left[\tfrac{2}{l_m S}\textstyle\sum_{k=0}^{S}\sum_{l=1}^{l_m}e^{-s_k t_l}(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_n))\right] \\
&= \left(\tfrac{1}{l_m}\right)^2 \mathrm{Var}\left[\textstyle\sum_{l=1}^{l_m} g_l^{nj}\right],
\end{aligned}$$

where g_l^{nj} is given as in (16). For the time distribution that we consider in [10], $\mathrm{Var}[t_l]$ is finite, as a result $\mathrm{Var}[e^{-s_k t_l}]$ is finite. Consequently, the variance of g_l^{nj} is finite as as summation over non-independent variables indexed by s_k:

$$g_l^{nj}(s_k) = \frac{2}{S}e^{-s_k t_l}(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_n))$$

where

$$\begin{aligned}
&\mathrm{Var}\left[g_l^{nj}(s_k)\right] = \\
&\quad = \tfrac{4}{S^2}(f_I^*(s_k;\mathcal{K}_j) - f_I^*(s_k;\mathcal{K}_n))^2\,\mathrm{Var}[e^{-s_k t_l}] < \infty,
\end{aligned}$$

$$\mathrm{Var}\left[g_l^{nj}\right] = \sum_{s_k=0}^{S}\sum_{s_o=0}^{S}\mathrm{Cov}\left[g_l^{nj}(s_k), g_l^{nj}(s_o)\right] < \infty.$$

Since $\mathrm{Var}[g_l^{nj}] < \infty$, and the g_l^{nj} are i.i.d., it is clear that

$$\lim_{l_m \to \infty}\frac{1}{l_m^2}\mathrm{Var}\left[\sum_{l=1}^{l_m} g_l^{nj}\right] = 0.$$

Consequently,

$$\lim_{l_m \to \infty}\mathrm{Var}\left[Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_j)}\right] = 0. \tag{18}$$

For $i, n \in \boldsymbol{\mathcal{L}}$ it, then, holds

$$\lim_{l_m \to \infty} P\{Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_i)} > 0\} \geq \lim_{l_m \to \infty} P\{Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_j)} > 0\} = 0,$$

since

$$E[Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_j)}] < 0.$$

For $n \notin \boldsymbol{\mathcal{L}}$

$$\lim_{l_m \to \infty} P\{Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_i)} > 0\} = 1 \geq \lim_{l_m \to \infty} P\{Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_j)} > 0\}, \tag{19}$$

since

$$E[Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_i)}] > 0.$$

This proves statement (6). Finally, the proof of (7) follows from the proof of (6) and from

$$\begin{aligned} &P\{Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_j)} \leq 0\} = \\ &= P\{\mathrm{MSE}_{\mathcal{K}_n}^{l_m} - \mathrm{MSE}_{\mathcal{K}_j}^{l_m}\} P\{Z_{l_m}^{(\mathcal{K}_j \to \mathcal{K}_n)} > 0\}, \\ &\forall j, n \in \boldsymbol{\mathcal{K}}. \end{aligned} \tag{20}$$

■

Lemma 3. *Algorithm 1 converges almost surely to a minimizer state.*

Proof: Consider, first, the case when Condition 3 holds. Since Conditions 1,2 hold as well, the requirements for convergence, according to Theorem 3.1 in [11] are satisfied and Algorithm 1 leads to mean square error minimization. Consider, now, the case when Condition 3 does not hold. Assume $\mathcal{K}_n \notin \boldsymbol{\mathcal{L}}$. If $\mathcal{K}_n$ is transient, then with probability one the sequence $\{\mathcal{K}_m\}$ of visited states will not converge to $\mathcal{K}_n$ as $m \to \infty$. Assume now that $\mathcal{K}_n$ is positive recurrent. By (19) in Lemma 2 we have that $\lim_{l_m \to \infty} P\{Z_{l_m}^{(\mathcal{K}_n \to \mathcal{K}_i)} > 0\} > 0, \forall \mathcal{K}_i \in \boldsymbol{\mathcal{L}}$. Consequently, $\{\mathcal{K}_m\}$ and all states of $\boldsymbol{\mathcal{L}}$ belong to the same communicating class, denoted by $\boldsymbol{\mathcal{K}^I}$, as well as all the other positive recurrent $\mathcal{K}_i \notin \boldsymbol{\mathcal{L}}$ states. The system is thus reduced to a set of states $\boldsymbol{\mathcal{K}^I}$. Clearly, Condition 1 holds for all states in $\boldsymbol{\mathcal{K}^I}$, and a result, the requirements for Theorem 3.1 in [11] are fulfilled. ■

V. PERFORMANCE EVALUATION

The performance of the discrete stochastic optimization based parameter estimation depends on the granularity of the discretization for each dimension of the state space, $\boldsymbol{\mathcal{K}}$, and on the number an the location of the s-domain points, on which the empirical and the analytic LTs are compared, for the MSE calculation. These parameters affect the accuracy of the parameter estimation, even if the optimal parameter vector is determined by exhaustive search.

In addition, we consider a limited idle period sample size, and terminate the algorithm when all idle period samples are integrated. This on one hand minimizes the time spent for parameter estimation, but on the other hand, does not ensure that the algorithm finds the optimal parameter vector. To evaluate the achievable estimation performance, we perform parameter estimation with exhaustive search and with early termination, considering a large set of model compliant traffic traces. We select 10^4 $(\xi, \sigma, p, p_{\mathrm{CCA}})$ parameter vectors, generate a sequence of idle and active periods for each vector, and run the estimation algorithms. The parameters are randomized according to Table I, to cover a wide range of traffic patterns. For the evaluations presented here we fix $S = 10^3, s_k \in (10^0, 10^5), 1 \leq k \leq S$, and integrate one new idle period sample in each iteration step.

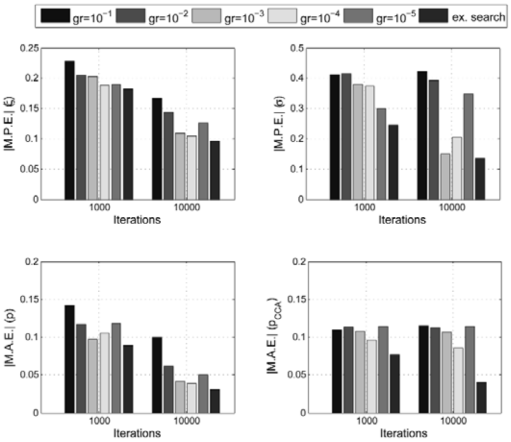

Fig. 3. The accuracy of the LT-based estimation with respect to the number of iterations, and the granularity of the state space. Exhaustive search results are shown for comparison.

TABLE I
MODEL PARAMETERS

Parameter	Distribution	Min	Max	Mean	StdDev
ξ	Truncated Gaussian	0.1	0.4	0.3095	0.1
σ	Truncated Gaussian	1e-4	0.1	0.02	0.2
p	Uniform	0.1	1.0		
p_{CCA}	Uniform	0.1	1.0		
α_{ON}	Uniform	0.0008	0.001		
β_{ON}	Uniform	α_{ON}	0.0015		
α_{BK}	Deterministic			0.0007	

As stated in Section II it is assumed that the $f_A(t)$ parameters can be estimated correctly and α_{BK} is known. We measure the estimation accuracy by calculating the mean absolute error (MAE) of the p and p_{CCA} and the mean percentage error (MPE) of the ξ and σ estimation.

As the number of idle period samples affects the time needed for continuous sensing and in our case even gives the number of iterations of the optimization algorithm, it is one of the main design parameters to be considered. Therefore we evaluate the parameter estimation performance for 10^3 and 10^4 idle period samples and iteration steps. In addition, to evaluate the effect of the size of the state space of the discrete optimization we alter the granularity of the discretization of the state parameters $\{\xi_i, \sigma_i, p_i\}$ between 10^{-1} and 10^{-5}, while bounding them within the respective intervals given in Table I.

Figure 3 compares the estimation accuracy of $\{\xi, \sigma, p, p_{\mathrm{CCA}}\}$ under exhaustive search and with stochastic optimization with early termination. Considering the number of integrated samples, we can see that the increased number of samples improves the estimation accuracy under exhaustive search. At the same time, an increased state space does not necessarily lead to better estimation accuracy. The estimation accuracy may increase with increased state space for a while, in this interval the minimizer is found, and the increased granularity means lower MSE. However, as the state space is further increased, the minimizer can not anymore be discovered in the limited number of iterations, and therefore the estimation accuracy drops. Therefore, the state space size has to be selected carefully, taking the expected number of samples into account.

The results show that the performance of the proposed algorithm is comparable to the one of the exhaustive search. A number of samples in the range of 10^4 and parameter granularity of $10^{-3} - 10^{-4}$ gives an estimation accuracy that is sufficient for the cognitive control as it was shown in [9], while it allows acceptable sensing times, and a state space size that is implementable on sensor devices with limited memory.

VI. Discussion

In the heterogeneous networking environment of the the open ISM band the prediction of the availability of the wireless resources is a key enabler for the design of energy efficient wireless networks. In this paper we considered the issue of WLAN and WSN coexistence. In this case WSN transmissions suffer from WLAN interference, because the WLAN carrier sensing does not detect the low power, narrow band WSN transmissions. The sensor network can avoid this interference, if it can characterize the channel occupancy, and tune its transmission parameters accordingly.

We described a semi-Markovian model of the WLAN channel occupancy, as observed by the individual sensor nodes and proposed a discrete stochastic optimization based algorithm to estimate the parameters of the idle time distribution in the Laplace domain. We showed that the proposed solution can achieve the required estimation accuracy by sequentially integrating the measured idle period samples and by simultaneously searching for the optimal parameter vector. We can conclude that the required idle time sample size allows limited sensing times and the parameter granularity can be low enough for the algorithm to be implemented in resource limited sensor nodes. Therefore the proposed algorithm can support the development of cognitive medium access control and routing in WSNs.

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BME-Infokom is working on implementing marketable ideas

SUCCESS RemoteSense, the wireless sensor network attracted many inquirers at Techshow

"We implement marketable innovative ideas by using our highly-qualified research and development resources" – highlighted dr. Sándor Szabó, managing director of BME-Infokom Innovátor Nonprofit Ltd. on the occasion of winning the Hungarian Innovation Techshow Best Presentation Award with RemoteSense system.

– What capabilities helped RemoteSense to achieve this success?

RemoteSense is wireless sensor networking on a whole new level. Utilizing radio technology and modern protocol the system is capable of transmitting information to a central node in self-organized, energy- and cost-efficient way, while the propagation properties are significantly better then other similar technologies'. RemoteSense provides an outdoor range of 2-3 kilometers and an indoor range of 3-5 floors operating on a pair of traditional AA batteries for years. The solution is highly suitable for smart metering purposes like remote measurement of water, gas or electricity. For environmental and healthcare monitoring the longer range and energy efficient operation with state of the art security solutions opens new possibilities. Two-way communications, and easy, automatized installation makes RemoteSense an ideal solution of data collection and device control for intelligent homes, and industrial or medical facilities. For more information about this new system, visit www.remotesense.eu.

– What other innovations are related to BME-Infokom?

One of our high priority projects nowadays is the iParking system, that manages homogeneous car-distribution in parking-lots, helps find free parking spots, and leads drivers back to their cars. I would also like to mention our RFID-based manufacturing process tracker system, and our indoor positioning systems, which relies on various technologies, for instance the Bluetooth.

– How are you related to the university?

The BME-Infokom Innovátor Nonprofit Ltd. is a technical research and development business organization, partly owned by the university. That means a close relationship, as several university researchers and students are working on our projects, and our doctoral students are actively taking part of the education at the university. The BME-Infokom was founded in 2009 with the goal to work on innovative R&D projects which will result in marketable intellectual properties, products, technologies, applications and services. Our previous and recent partners are national businesses, mostly from the telecom sector, system- and solution-oriented companies, and other IT enterprises. We collaborated with Nokia Siemens Networks, TcT Hungary Ltd., Bird Telecom Ltd., National Media and Infocommunications Authority as a partner. Since 2012, we ha've expatended beyond our nine main development fields, what means weto provide a wider spectrum of services and industrial solutions to our future partners. Since 2012, we've extended beyond our nine main development fields, what means we provide a wider spectrum of services and industrial solutions to our future partners.

– How do you inspire students to contribute?

We are open to new ideas. Guided by this attitude we organized a student brainstorming conference in collaboration with BME Department of Telecommunications. We also offer positions in internship programs, project laboratories, and part-time jobs. Our main strengths that attract students are flexible project management, and highly-qualified employees. Our team has already fifty young, highly-trained members, like application programmers, software developers, electrical and computer engineers, economists, PhD researchers and doctoral students, who create competitive technical solutions reacting to the main challenges of the market.

Our work presented in this article was supported from the KMOP-1.1.2-08/1-2008-0001 project by the BME-Infokom Innovátor Nonprofit Kft.

The main development fields of BME-Infokom:

1. Info-communication network design and analysis models
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6. Intelligent transport applications
7. Radio Frequency Identification
8. Tracking and location-based services
9. Intelligent Environment

Source: BME-Infokom

(x)

The team of BME-Infokom with their new innovation at the national Techshow

Extending QoS support in the IP multimedia subsystem: Mobility-aware session reconfiguration

Ognjen Dobrijevic, *Member, IEEE,* and Maja Matijasevic, *Senior Member, IEEE*

***Abstract*—This paper presents an approach to controlling service continuity and quality of service (QoS) in the 3rd Generation Partnership Project (3GPP) IP multimedia subsystem (IMS). The approach extends existing support to manage QoS by dynamically producing multimedia session configurations with respect to different mobility types, which is referred to as *mobility-aware session reconfiguration*. We first give a state-of-the-art overview regarding IMS support for service continuity and QoS management. Then, we propose to enhance QoS support in IMS by introducing a Session Initiation Protocol (SIP) application server that is responsible for (1) generating QoS specifications which conform to the mobility-induced constraints and (2) reconfiguration decision-making. Moreover, we describe several SIP signaling procedures which enable the session reconfiguration, and demonstrate the application of our approach in an IMS laboratory prototype, which also serves for a performance evaluation of the solution.**

***Index Terms*—Multimedia communication, Quality of service, Mobile communication.**

I. Introduction

WITH the widespread use of multimedia services, deployment of network architectures for their provisioning has gained momentum. One of such architectures is the Internet Protocol (IP) multimedia subsystem (IMS) [1]. A critical aspect of service provisioning is quality of service (QoS) support, which enables service-level negotiation of QoS parameters and transport-level control of network resources based on the agreed service parameters [2]. By deploying IP networks that adhere to the Next Generation Network (NGN) concept [3], including IMS, the providers are committed to enable users to communicate and access services independently of changes that may stem from different mobility types, e.g. *terminal mobility* and *session mobility*. The former provides uninterrupted communication when a user terminal switches between access networks of different technologies (called *vertical handover*), while the latter allows to seamlessly move multimedia sessions between user terminals. To facilitate QoS continuity that meets such a prerequisite, management approaches also include means to adapt service to mobility.

The 3rd Generation Partnership Project (3GPP) IMS [4] has been specified as a network subsystem that offers session control for multimedia service provisioning, with the control being based on the Session Initiation Protocol (SIP) [5]. However, current 3GPP specifications reveal some limitations regarding QoS support. The basic IMS specification [4] describes procedures for the session end-points to negotiate session parameters, which is based on the simple offer-answer model [6]. On the other hand, the 3GPP-defined QoS framework [7], which also applies to IMS, focuses on resource allocation and QoS provisioning at transport level in access networks. We argue for an approach to managing QoS that would enable:

- service delivery that is tailored to match user preferences, user terminal capabilities, service requirements, and access network characteristics;
- reservation of network resources that is optimal in, e.g., sharing bandwidth across media components; and
- service adaptation to different mobility types in order to regulate QoS or prevent its degradations.

The goal of QoS management is to apply "the best" session configuration and network resources allocation that maximize QoS parameters in terms of, e.g., bandwidth and delay.

Our approach envisages the following use case scenarios. Users access an IMS network to establish multimedia services, e.g., video-on-demand, or audio-video conferencing, which are hosted by application servers. Each user may utilize different terminals and may move the ongoing sessions between them.

1) During the establishment phase, session parameters are negotiated between user's terminal and the related application server (AS), and include feasible media components, their characteristics, and the needed QoS specification. This leads to the resources reservation.
2) When a user decides to replace her/his terminal, session parameters are reconfigured to meet capabilities of the targeted terminal. The reconfiguration may, e.g., modify media encoding parameters and produce a new QoS specification, thus leading to the adjustment of resources allocation before moving media components.
3) When a user's terminal changes its location, session may be reconfigured by moving its media components to another AS instance in order to maintain QoS. AS instances are deployed for the load balancing purposes and are associated with distinct locations.
4) When a user's terminal changes access network, session parameters are reconfigured to match capabilities of the new access technology. The reconfiguration is completed after allocating network resources, possibly based on a newly negotiated QoS specification.

To the best of our knowledge, there is no a comprehensive IMS-based solution that addresses all these scenarios (e.g., the IMS service continuity specification [8] defines procedures that maintain a service with regards to scenarios 2) and 4), but it does not regulate how to facilitate QoS continuity).

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This paper presents our approach to enhancing QoS support in the 3GPP IMS. By dynamically producing session configurations with respect to session and terminal mobility, we are able to maintain or adapt QoS across different user terminals and access networks. We outline requirements for implementing the session reconfiguration in IMS, which are derived from the use case scenarios and our generic session reconfiguration model (presented in a previous work [9]). We propose to introduce an SIP AS that (1) generates QoS specifications conforming to the mobility-induced constraints and (2) decides on the reconfiguration. This SIP AS, named the Session Configuration Management (SCM) AS, also controls IMS service continuity. The SCM AS can be considered a reusable service offered by the IMS network, which would relieve equipment manufacturers and service providers of implementing its specific functionalities. We design several SIP procedures, which are based on the 3GPP specifications, to enable the session reconfiguration. We also demonstrate the application of our approach in an IMS laboratory prototype, which also serves for a performance evaluation of the solution.

The remainder of this paper is organized as follows. An overview of service continuity and QoS support in IMS is given in Section 2. In Section 3, we outline requirements for implementing the session reconfiguration in IMS. Section 4 presents the SCM AS and the IMS prototype. We demonstrate the approach and analyze its performance evaluation in Section 5, followed by the conclusion section.

II. SERVICE CONTINUITY AND QOS SUPPORT IN IMS

A. 3GPP specifications

The 3GPP IMS is an NGN-compliant architecture, which was designed to be independent of the access network technologies. A simplified view of the 3GPP IMS architecture is given in Figure 1. To control multimedia sessions, IMS defines a number of functional entities. The main entities are:

- IMS User Equipment (UE), which issues requests for session establishment and modification;
- SIP AS, which enables introduction of new services to the IMS network by hosting and executing them;
- Proxy-Call Session Control Function (P-CSCF), which represents the first contact point for a UE on the signaling path towards the rest of IMS core entities;
- Serving-CSCF (S-CSCF), which offers a coordinated SIP interaction among IMS entities and selects an SIP AS depending on the service to be invoked; and
- Home Subscriber Server (HSS), which stores subscription information to authenticate and authorize users, and information related to the user's location and IP address.

The 3GPP specifications consider different aspects of the overall QoS support in IMS. Technical specification (TS) 23.228 "IP Multimedia Subsystem (IMS)" [4] provides a high-level description of SIP procedures for the session end-points to negotiate multimedia session parameters. The specification elaborates on how to determine media characteristics in the session establishment phase or when a session is modified in the context of, e.g., adding a media component or changing bandwidth requirements. The negotiation procedure is based

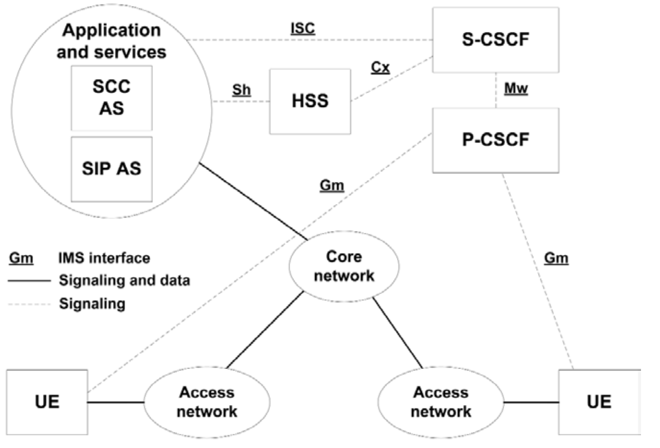

Fig. 1. A simplified view of the 3GPP IMS architecture

on the simple offer-answer model [6], which proposes a mechanism for the end-points to reach a common view of the session. In the model, one end-point offers a set of desired session parameters to the other, while the other end-point answers with the session parameters that are wanted from its perspective. Such a procedure may take multiple negotiation steps until the media characteristics are agreed upon.

TS 23.207 "End-to-end QoS concept and architecture" [7], on the other hand, defines the 3GPP QoS framework, which also applies to IMS. The framework, however, focuses on QoS management and resources allocation at the IP-bearer level and the transport-bearer level of different access networks.

TS 23.237 "IMS Service Continuity" [8] provides a high-level description of service-level (SIP) procedures that transfer a session between different access networks (the specification refers to this process as the *access transfer*, AT) or different UEs (the *inter-UE transfer*, IUT). This specification employs two functional entities to execute the service continuity mechanisms: (1) the SCC AS, which is an SIP AS (Figure 1), and (2) a UE with the associated support. Multimedia sessions started by UEs are anchored at the SCC AS, which uses the 3rd party call control (3pcc) mechanism to facilitate session transfer. The SCC AS is inserted in the SIP signaling path and selected by an S-CSCF to control the AT or the IUT, while the UE initiates the transfer procedures. The AT is triggered based on the criteria such as operator policy, user preferences, and access network conditions, while user input starts the IUT.

B. Other related work

The related QoS research efforts, for which a summary and a comparison are given in Table I, mainly differ in two aspects. The first one regards different degrees of QoS support offered:

1) service continuity is provided, but without any QoS guarantees;
2) resource allocation is performed, but no form of QoS negotiation is included;
3) QoS negotiation is employed solely in a service set-up phase; and
4) QoS adaptation is supported during the course of a service lifetime.

TABLE I
A SUMMARY AND A COMPARISON OF RELATED WORK

Approach	Considered deployment levels	Degree of QoS support	Regarded mobility types	Supported service customization
Y. C. Yee et al. [10]	Service level	Service continuity	Terminal mobility	Move sessions among user terminals, change network attachment point for the terminals, switch media codecs
E. Cerqueira et al. [11]	Transport level	QoS adaptation	Terminal mobility	Remove/add media components of different priority from/to sessions, assign different QoS classes to sessions
K. S. Munasinghe and A. Jamalipour [12]	Service level	Service continuity	Terminal mobility	N/A
M. Rawashdeh and A. Karmouch [13]	Service level	Service continuity	Session mobility	Change video framerate
P. Bellavista et al. [14]	Service level	Service continuity	Terminal mobility	N/A
W.-K. Chiang and P.-C. Kuo [15]	Service level	Service continuity	Terminal mobility	N/A
M. Navarro and Y. Donoso [16]	Transport level	Resource allocation	N/A	Assign different QoS classes to sessions
S.-R. Yang and W.-T. Chen [17]	Service level + transport level	QoS negotiation	Terminal mobility	N/A
T. Renier et al. [18]	Service level + transport level	QoS negotiation	Terminal mobility	N/A
J. Liao et al. [19]	Service level + transport level	QoS adaptation	Terminal mobility	Change point of access network attachment for user terminals
L. Skorin-Kapov et al. [20]	Service level + transport level	QoS adaptation	None	Add/remove media components to/from sessions, switch media codecs, adapt allocation of network resources

While most of the referred solutions do not employ QoS adaptation, those that facilitate it focus on control procedures at transport level and consider a limited set of parameters to be adjusted. The other research efforts' aspect relates to handling distinct mobility types – a majority of the approaches is centered on either terminal mobility or session mobility.

The *Proactive and Adaptive Handover* (PAHO) system [10] is an SIP-based approach that customizes service configuration in the event of network performance degradation caused by terminal mobility. To maintain service continuity, the PAHO, e.g., moves media components among user terminals or switches media codecs. The PAHO, however, does not include QoS negotiation and resources reservation. The *Multi-User Session Control* (MUSC) approach [11] regulates session QoS parameters in response to terminal mobility and network performance variations. The MUSC employs transport-level procedures to coordinate QoS adaptation, which leads to removing or adding "lower-priority" media components, or, assigning different QoS classes to a session. It enables to match session QoS requirements with the available QoS classes, but does not support service delivery that adjusts to user preferences and terminal capabilities, as well as handling of session mobility.

An IMS platform that controls mobility between a Wireless Local Area Network (WLAN) and a Universal Mobile Telecommunications System (UMTS) network is presented in [12]. It offers service continuity with a make-before-break type of handover between the two network types, but does not allow to adjust QoS parameters to the targeted network characteristics nor to allocate necessary resources. A seamless video transfer for session mobility in IMS is described in [13]. It focuses on minimizing the disruption time when moving a video session between UEs, but does not tackle the issue of providing QoS. The IMS-compliant Handover Management AS (IHMAS) [14] is introduced in the IMS network to achieve session continuity upon a vertical handover. This solution proactively triggers SIP signaling with the targeted access network by having UEs predict the handover. A similar SIP AS approach, referred to as the Centralized Service Continuity (CSC), is presented in [15]. The continuity is managed by the CSC AS, which, similarly to the SCC AS and the IHMAS, acts as the session anchor point and performs the 3pcc for session re-establishment. However, the last two approaches do not provide any QoS guarantees.

An IMS-centered enhancement of the 3GPP QoS framework is presented in [16]. Its transport-level approach reassigns QoS classes to the sessions based on the network state and resource availability. Another QoS framework for IMS is proposed in [17]. It alleviates the influence of terminal mobility on QoS by having UEs trigger resources reservation at "neighboring" IMS networks which they may visit during service execution. This way, QoS agreements from the service set-up phase tend to be preserved, but with possibly a large waste of the resources that will not be used. An approach for IMS that offers session continuity in response to vertical handover is described in [18]. It enables delivery of agreed QoS parameters between P-CSCFs that control different access networks, but assumes that network conditions remain the same after the handover.

An improvement of the framework presented in [17] is an enhanced IMS handover mechanism (EHM) [19]. The EHM employs a mobility prediction algorithm to detect a UE's movement between network attachment points. Before the UE moves to a new IMS domain, the EHM chooses the most appropriate access network, which leads to reserving resources in advance. Still, this approach lacks both the means to react to session mobility and parameters such as UE capabilities, service requirements, and budget constraints when customizing service delivery. Our previous work [20] proposes an SIP AS for IMS that hosts a function for matching communication requirements of the parties involved in service establishment and for calculating the reservation of network resources that is optimal in distributing them among the media components. The matching function was designed to assist in the QoS negotiation by producing an offer of feasible media components for a session and their characteristics. However, this work does not consider a service adaptation to mobility.

III. Proposed enhancements to QoS support in IMS

After presenting an overview of IMS support for service continuity and QoS management, we now focus on requirements regarding the session reconfiguration and meeting them in IMS. The session negotiation procedure in IMS resides on a simple matching among capabilities and requirements of multimedia session's end-points, which must agree on session parameters such as type and encoding of media components. In order to provide QoS guarantees by applying controllable values of network performance indicators, the associated QoS support must somehow map session requirements to, e.g., expected network bandwidth and delay.

As mentioned in the previous section, we have proposed a common function, named the QoS Matching and Optimization Function (QMOF), that produces session configurations based on user preferences, user terminal capabilities, multimedia service requirements, and access network constraints [20]. A session configuration is feasible when it meets these criteria:

1) user terminal capabilities comply with the processing requirements of desired media components;
2) access network constraints (e.g., available bandwidth and delay) support the minimum requirements on network performance for desired media components; and
3) user preferences, such as respective relevance of media components, are fulfilled.

The purpose of the matching functionality is to enhance the negotiation procedure by offering a number of potential configurations for a particular session, with all of them meeting the mentioned criteria, but differing in calculated parameter values. Another aspect of the QMOF relates to determining the optimal QoS reservation of network resources across media components of the session (the optimization objective may be formulated in various ways and specified by, e.g., the network operator). The interested reader is referred to our previous work [20] for details on the optimization and the QMOF.

Furthermore, we argue for additional mechanisms that would improve IMS support beyond service continuity and facilitate QoS continuity in the event of session mobility and terminal mobility. Besides offering procedures to control session transfer between different user terminals and different access networks, we believe that IMS support should include the means to decide on how a session is reconfigured in response to mobility-induced changes that emerge during service execution. For instance, if the available access bandwidth decreases after a vertical handover, the reconfiguration could lead to reducing bandwidth demands for all the session's media components or to removing a media component from the session in accordance to user preferences. Moreover, if capabilities of the user terminal improve as a result of session mobility (e.g., replacing a mobile phone with a personal computer), a session could be reconfigured to apply codecs with higher bit rates.

Our generic session reconfiguration model [9] presents requirements and functional entities in the context of NGN [3] that are needed to control session parameters and manage QoS. The model is based on two main concepts, the one of a *mobility event*, and the other of a *reconfiguration primitive*. Mobility events represent changes stemming from mobility:

1) *Change of terminal* – a change of the user terminal due to session mobility;
2) *Change of location* – a change in user terminal's location due to terminal mobility; and
3) *Change of access network* – a change of the terminal's access network due to the vertical handover.

When a mobility event occurs, the associated *mobility event notification* carries context information regarding the change, which is used to direct the reconfiguration process. One of the key functions in the reconfiguration model is the Session Reconfiguration Function (SRF), which analyzes delivered event notifications and decides on the reconfiguration primitive(s) to be executed. A result of this decision may, as well, involve invoking the QMOF, e.g., if new session configurations need to be produced. The reconfiguration primitives are management operations which are executed to regulate session configurations: (1) *start media flow*, (2) *modify media flow*, and (3) *stop media flow* (the reconfiguration model distinguishes among media components, which it refers to as media flows).

3GPP defines the IMS architecture in a way to enable multimedia services to take advantage of common IMS functions and different service enablers via standardized interfaces. To meet the outlined requirements and implement the session reconfiguration in the 3GPP IMS, we exploit the IMS mechanisms currently specified, and propose to combine the QMOF and the SRF on an SIP AS that we name the Session Configuration Management (SCM) AS. The SCM AS would be responsible for producing session configurations during the service establishment phase and for customizing the configurations in order to adapt them to mobility-induced constraints (i.e., changes in UE capabilities and access network characteristics that follow after the mobility). One of its most important functional aspects is calculation of QoS specifications that define required network performance in terms of bandwidth, delay, jitter, and packet loss ratio. Another aspect relates to deciding on how to perform the reconfiguration.

As IMS specifies functional entities rather than network nodes, we also propose to integrate functionalities of the SCC AS [8] into the SCM AS, which would allow it to control the service continuity as defined by IMS. Being accessible over the IMS Service Control (ISC) interface, the SCM AS can be regarded to as a reusable common service offered by the IMS network. This would relieve UE manufacturers and service providers of implementing specific and rather complex QMOF and SRF functionalities.

IV. The IMS laboratory prototype

To demonstrate the application of our approach, we implement an IMS laboratory prototype with the SCM AS (Figure 2). For the demonstration purposes, we design several SIP signaling procedures, which build on the IMS session control procedures (described in [4], [21], [22], [8]), to enable the session reconfiguration. The central part of the prototype

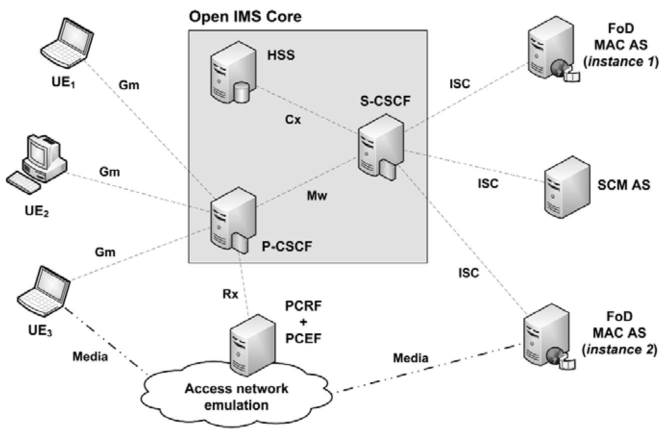

Fig. 2. Architecture of the IMS-based laboratory prototype

is the Open IMS Core (OIC) [23], which is a reference, open source implementation of IMS CSCFs and HSS. We utilize OIC to implement P-CSCF, S-CSCF, and HSS, where the latter is only used for user authentication and authorization.

A. Prototype components

The prototype uses two multimedia applications we developed: Audio-Video Call (AVC) and Football on Demand (FoD). Each of these multimedia services is described with a *service profile*, which specifies its media components and supported encodings, demands on network performance, available transport protocols, etc. AVC offers a conversational service that enables two users to establish an audio-video call. This application incorporates the VLC media player [24] to support live media streaming by the Real-time Transport Protocol (RTP). FoD is a simple Video-on-Demand service for users to watch prerecorded football matches. It is hosted by an SIP AS and also uses VLC for RTP streaming. An FoD feature includes the existence of multiple SIP AS instances, which are organized in a way to serve UEs at different locations. For the demonstration purposes, the location is determined by IP address a UE is assigned to while on the move. These instances are deployed for the load balancing purposes, while moving media flows between them may aid in maintaining QoS.

Each user holds a *user profile*, which specifies her/his preferences (e.g., favored access technology) and capabilities of the associated UEs. For the demonstration purposes, user profile also contains types and predefined characteristics (e.g., available bandwidth, delay, and jitter) of different access networks, which are then signaled in the given use case scenario. A UE enables the user to establish multimedia sessions and to access the offered services. SIP signaling for UE is provided by a signaling application programming interface (SAPI), which relieves application developers from the need to know signaling protocol details. UE also implements modules for VLC streaming control and media reproduction, which are executed for AVC and FoD. FoD is hosted and executed by the SIP AS implementation we refer to as FoD MAC AS (this SIP AS is also built upon the SAPI functionality). As such, it may be the responsibility of either an IMS operator or a 3rd party service provider. FoD MAC AS holds source files of the football matches, adapts their content according to the negotiated session parameters, and streams them by using VLC. In addition, it stores the accompanying service profile.

By implementing the QMOF logic and the SRF logic independently of a specific multimedia service, the SCM AS can be used for different users and multimedia applications. Its inclusion in the SIP signaling path is decided based on service control rules specified at the chosen S-CSCF. The QMOF matches parameters from user and service profiles to recommend feasible combinations of session media flows and their operating parameters, and produces an *optimized service profile*, which contains QoS specification(s) for the resources reservation. One of the most significant aspects of the QMOF is calculation of a Media Degradation Path (MDP). The MDP is a list of an optimal and a number of suboptimal combinations of resources allocations across media flows, which are referred to as MDP configurations and conveyed in SIP messages. They may be used for adapting the resources allocation. The SRF, on the other hand, analyzes delivered mobility context information and controls the reconfiguration.

The Policy and Charging Control (PCC) architecture [25] is responsible for policy control in NGNs and can, thus, be applied in IMS. PCC involves the Policy Control Resource Function (PCRF) and the Policy Control Enforcement Function (PCEF), which are implemented in our prototype [26]. The PCRF decides of the resources authorization and of a suitable MDP configuration to apply in accordance to resource availability, while the PCEF imposes chosen QoS rules. The PCRF interacts with a P-CSCF and the PCEF via Diameter [27]. The P-CSCF selects MDP from a related SIP message and delivers it to the PCRF. The PCRF then invokes the PCEF to identify resource availability and to carry out the authorization/allocation. The PCRF sends a final decision about the allocation back to the P-CSCF. Network resources are emulated by using the Wide Area Network Emulator (WANem) [28]. This tool can be used for emulating multiple network characteristics, including bandwidth, delay, and jitter, which then affect network performance for the media flows.

User and service profiles are organized in an eXtensible Markup Language (XML) format and conveyed among the prototype components in SIP messages. XML was chosen to support modularity and extensibility, but our goal is to introduce the profiles which employ the Session Description Protocol [29], a standardized format for describing multimedia sessions. By then, we are working on the introduction of the XML Document Management (XDM) [30]. XDM enables to manage data stored in XML format on a central file repository. Such an approach would allow to retrieve the profiles past SIP signaling and, thus, to reduce signaling overhead.

B. SIP control procedures

In order to facilitate session reconfiguration and QoS management, we have designed five SIP signaling procedures in IMS for the use case scenarios:

1) *Session establishment* and *Session termination*,
2) *Session reconfiguration upon a change of terminal*,

3) *Session reconfiguration upon a change of terminal's location*, and
4) *Session reconfiguration upon a change of terminal's access network.*

Session establishment negotiates QoS and other session parameters between two end-points, while *Session termination* stops media flows and releases the allocated resources. The procedure applied in response to *Change of terminal* negotiates QoS parameters that conform to the targeted terminal capabilities and adjusts the resources allocation. When a terminal changes location, the associated procedure may result in transferring media flows to maintain QoS. The procedure invoked due to *Change of access network* tunes QoS parameters to the new access characteristics and reserves resources in the network.

1) Session establishment: Figure 3 shows SIP message sequence for establishing media flows between a UE and an MAC AS (e.g., for FoD), which focuses on agreeing upon session parameters and reserving necessary resources.

The sequence assumes that a user and her/his UE are registered to the IMS network. When the user requests a service via its SIP address (i.e. SIP Uniform Resource Identifier, SIP URI), the UE sends an SIP *INVITE* request (step 1, Figure 3) that conveys the user profile to the SCM AS and the MAC AS. The corresponding service profile is delivered to the SCM AS in an SIP *183 (Session Progress)* response (steps 7-8), which triggers the QMOF to generate a *feasible service profile* that comprises an offer of media flows and their parameters (step 9). This profile is then sent to the UE, from which the user chooses among the offered session parameters (step 15).

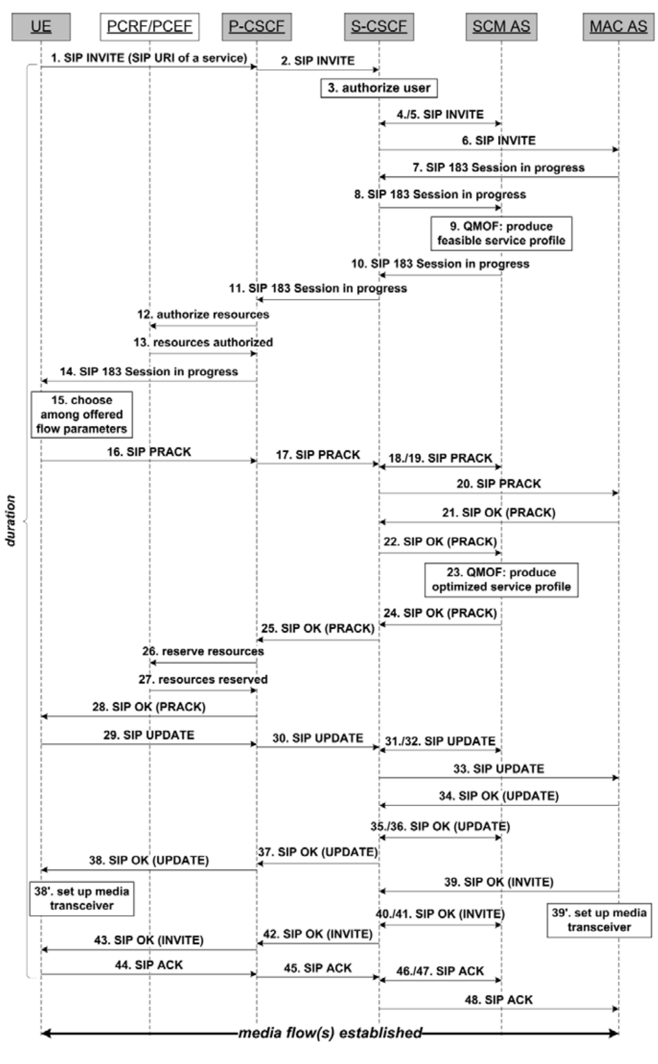

Fig. 3. SIP signaling for *Session establishment*

The resulting service profile is delivered in an SIP *PRACK* request to the MAC AS. When an SIP *OK (to PRACK)* response traverses the SCM AS, the QMOF invokes the *optimization process* to generate an *optimized service profile* (step 23). This profile includes a determined MDP, which is employed for the resources allocation (steps 26-27). If the allocation is successful, the optimized service profile and the applied MDP configuration are forwarded to the UE and the MAC AS (steps 29-38) to start media transmission and establish the agreed flows.

2) Session reconfiguration upon a change of terminal: Figure 4 shows SIP message sequence that negotiates QoS parameters while transferring media flows from, e.g., UE1 to UE2. This signaling sequence assumes that UE1 and UE2 are controlled by the same P-CSCF and the same S-CSCF, but this does not affect its generality. Definition of the procedure is based on the IMS service continuity specification and SIP specification for managing session transfer [31].

In the first part of the procedure, the targeted UE (UE2) is required to establish the current media flows with the MAC AS (thus applying the *start media flow* primitive). This part is identical to the *Session establishment* procedure. To complete the transfer, the flows then need to be terminated between the originating UE (UE1) and the MAC AS. An SIP *REFER* request is employed for delivering the established service information to UE2 (Figure 4, steps 1-6), including address of the used MAC AS, which leads UE2 to invite the MAC AS to establish the flows. The SRF at the SCM AS is invoked (step 20) to examine whether a feasible service profile has already been produced for UE2 and the associated user profile. The latter, together with the delivered session parameters, represent mobility context information for this scenario. If there is no profile produced, the QMOF determines a new service profile offer that takes capabilities of UE2 into account.

After the flows are established between UE2 and the MAC AS (steps 13-59), UE2 sends an SIP *NOTIFY* request to inform UE1 of the transfer. This SIP request triggers UE1 to terminate its participation in the communication (thus applying *stop media flow*), which is initiated by sending an SIP *BYE* request to the MAC AS (steps 72-76). When the MAC AS receives the termination request, it sends an SIP *OK (to BYE)* response to UE1, which is also used for invoking the release of allocated resources (steps 81-82).

3) Session reconfiguration upon a change of terminal's location: If, e.g., by replacing UE the location is also changed, the SRF at the SCM AS may decide upon moving media flows between different MAC AS instances. That way, media flows can be established with the instance that is "closer" to the used UE, which could help in maintaining QoS. In that case,

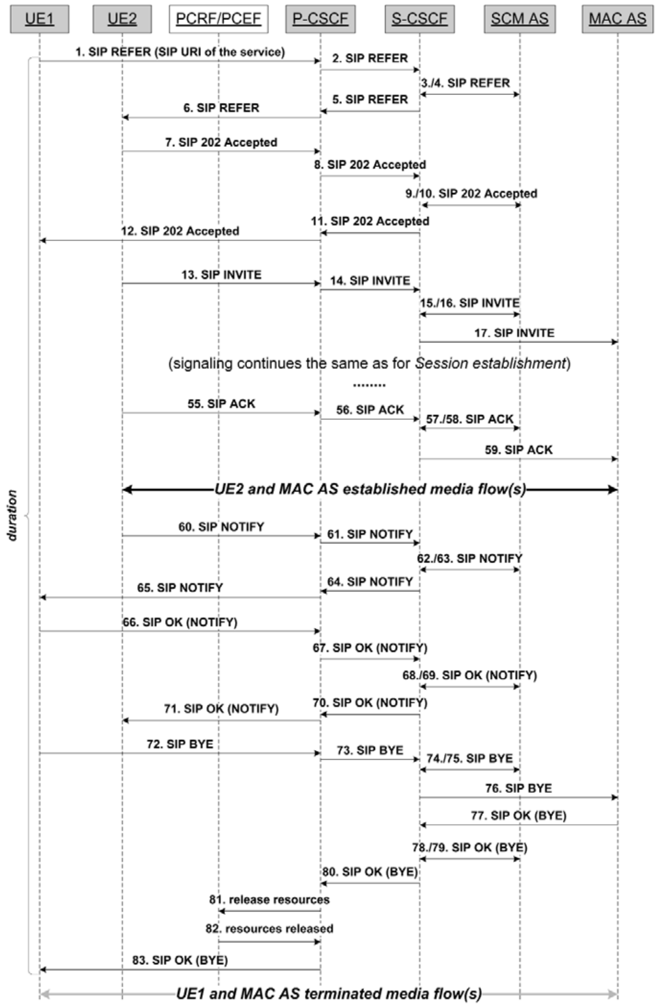

Fig. 4. SIP signaling for *Session reconfiguration upon a change of terminal*

the SCM AS will modify the SIP *REFER* request (step 3 in Figure 4) to target another MAC AS instance (e.g., MAC ASi2 instead of MAC ASi1) by providing its address. This would instruct UE2 to establish the flows with MAC ASi2.

V. Case study and performance evaluation

A. Experimental testbed

Case study demonstration and performance evaluation measurements are conducted in an experimental network shown in Figure 5. Configuration of the nodes which host the components of the IMS laboratory prototype is depicted in Table II. It must be emphasized that the network does not involve any traffic besides the one pertaining to the applied SIP procedures and to media delivery within the prototype services.

B. Case study scenario

The purpose of this case study is to demonstrate application of our approach when QoS for media flows is negotiated during the establishment phase and adapted when a user decides to change the terminal for communication, which also includes a change of location and of access network.

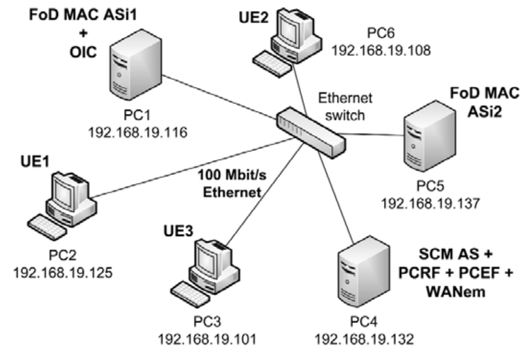

Fig. 5. Topology of the experimental testbed

TABLE II
CONFIGURATION OF THE TESTBED NODES

Node	Oper. system	Configuration
PC1	Linux Ubuntu	Pentium IV, CPU 3.0 GHz, RAM 1 GB
PC2	Linux Ubuntu	Pentium IV, CPU 2.4 GHz, RAM 512 MB
PC3	Linux Ubuntu	Pentium IV, CPU 1.7 GHz, RAM 1 GB
PC4	Linux Ubuntu	Pentium IV, CPU 1.7 GHz, RAM 1 GB
PC5	Linux Ubuntu	Pentium IV, CPU 1.6 GHz, RAM 512 MB
PC6	Linux Ubuntu	Pentium IV, CPU 1.7 GHz, RAM 1 GB

1) Session establishment: In the first part of the scenario, two friends, Alice and Bob, decide to watch a football match together over the IMS network. Their IMS operator offers a FoD service via a 3rd party service provider, which deploys multiple MAC AS instances for the service (e.g., FoD MAC ASi1 and FoD MAC ASi2). Alice is at home. She uses her laptop computer (represented by UE1) over an Asymmetric Digital Subscriber Line (ADSL) connection, which supports a downlink of 10 Mbps and an uplink of 512 kbps, to establish a session with an FoD MAC AS. This session will be referred to as *session1*. Based on the Alice's location, UE1 establishes the session with FoD MAC ASi1, which comprises one audio and one video flow (Figure 6).

At the same time, Bob is traveling home by train. He uses his smarthpone (UE2) over a High-Speed Packet Access (HSPA) connection, which supports a downlink of 3.6 Mbps and an uplink of 384 kbps, to watch the game. Based on his location, UE2 establishes the session with FoD MAC ASi2.

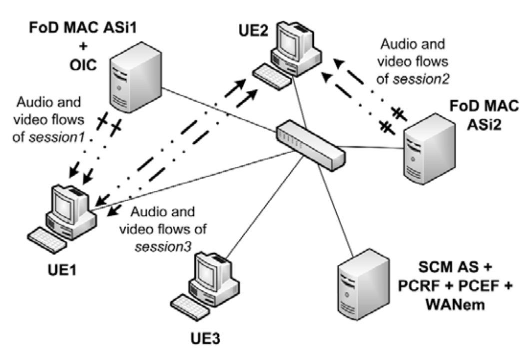

Fig. 6. Flow map after establishing sessions

This session will be referred to as *session2*. Just after the match started showing, Alice invites Bob to an audio-video call, so they can comment on the match together. The latter session, *session3*, comprises two audio and two video flows (Figure 6). An example of the QoS parameters that result from the *Session establishment* procedures is depicted in Table III, with eight media flows established in total. These parameters are used at WANem to reserve necessary resources and provide QoS.

TABLE III
RESULTING QOS PARAMETERS AFTER ESTABLISHING SESSIONS

Media flow	Bandwidth (kbps)	Delay (ms)	Jitter (ms)	Drop (%)
session1: video	1024	200	100	0.4
session1: audio	64	200	100	0.4
session2: video	512	200	100	0.7
session2: audio	48	200	100	0.7
session3: video1	128	100	50	1.0
session3: audio1	32	100	50	1.0
session3: video2	128	100	50	1.0
session3: audio2	32	100	50	1.0

2) Session reconfiguration: After coming home, Bob decides to transfer the communication to his laptop computer (UE3), which is connected to the network via an ADSL connection. For the demonstration purposes, this request also includes a change in location, which is represented by different IP addresses. While processing the SIP *REFER* request (Figure 4, step 3), the SRF at the SCM AS processes the change in location and instructs UE3 to establish media flows of *session2* with FoD MAC ASi1, instead of FoD MAC ASi2. After *session2* and *session3* are transferred (Figure 7), Bob continues watching the match and chatting with Alice on UE3.

To achieve service continuity, SIP messages of the transfer procedure carry information about the elapsed time for the match, which enables Bob to resume watching the game from the right moment. An example of the resulting QoS parameters, which are enforced after the flow transfer, is shown in Table IV. The parameters from the first part of the scenario are improved regarding enhancements in hardware configuration of UE3 and its access network. Transport parameters of *session2* and *session3* are updated to reflect the new UE.

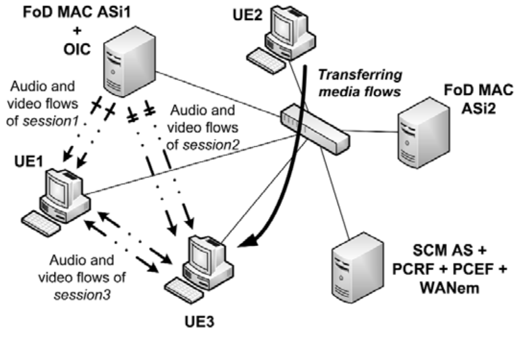

Fig. 7. Flow map after transferring media flows

TABLE IV
RESULTING QOS PARAMETERS AFTER TRANSFERRING MEDIA FLOWS

Media flow	Bandwidth (kbps)	Delay (ms)	Jitter (ms)	Drop (%)
session1: video	1024	200	100	0.4
session1: audio	64	200	100	0.4
session2: video	1024	200	100	0.4
session2: audio	64	200	100	0.4
session3: video1	256	100	50	0.7
session3: audio1	64	100	50	0.7
session3: video2	256	100	50	0.7
session3: audio2	64	100	50	0.7

C. Performance evaluation

A preliminary performance evaluation of our approach in an IMS setting is conducted to analyze delay induced by the reconfiguration procedures, with the focus on examining a scalability of the solution. For the purposes of this work, we define the *duration* performance metric that refers to the time interval required to complete a specific SIP procedure from the user perspective (Figures 3 and 4). This metric is similar to the SIP performance metrics specified in [32]. For *Session reconfiguration upon a change of terminal*, *duration* is the interval between sending *1. SIP REFER* and receiving *83. SIP OK (BYE)* at UE1, while its “reference value” implies procedure completion for a single UE. *Duration* is measured in relation to the number of UEs simultaneously executing a particular procedure with, e.g., an MAC AS. The measurement results for the analyzed SIP procedures are given in Table V. Average *duration* was obtained over 30 test runs.

TABLE V
AVERAGE *duration* FOR THE ANALYZED SIP PROCEDURES

The procedure / **Number of UEs**	**1**	**4**	**7**	**10**
Session establishment [s]	6.13	6.61	7.37	8.32
S. recon. u. a chan. of terminal [s]	7.62	8.13	8.85	9.89
S. recon. u. a chan. of term. locat. [s]	7.66	8.18	8.93	10.00

Signaling load of multiple UEs exchanging SIP messages with an MAC AS instance is achieved by employing the SIP traffic generator called SIPp [33]. SIPp is able to create SIP messages as per user-defined scenarios, and we customize it to send the messages of the applied reconfiguration procedures. The performance, besides the SIP message exchange, is influenced by time duration of the matching and optimization processes at the SCM AS (comparing to the SRF, for which it is negligible). Table VI shows average QMOF processing time.

TABLE VI
AVERAGE QMOF PROCESSING TIME

The QMOF process / **Number of UEs**	**1**	**4**	**7**	**10**
Matching process [s]	0.74	1.10	1.46	1.88
Optimization process [s]	0.19	0.31	0.45	0.61

As user and service profiles constitute a signaling overhead, which affects the overall performance, we alter the SIP procedures in a way that SIP messages only reference the profiles, instead of carrying them along the signaling path. This required all the profiles to be produced in advance and stored at each prototype component that uses them. The measurement results are given in Table VII.

TABLE VII
AVERAGE *duration* FOR THE PROCEDURES WITH PROFILE REFERENCING

The procedure / **Number of UEs**	**1**	**4**	**7**	**10**
Session establishment [s]	5.08	5.55	6.28	7.33
S. recon. u. a chan. of terminal [s]	6.59	7.11	7.84	8.79
S. recon. u. a chan. of term. locat. [s]	6.64	7.15	7.92	8.89

The results show that *duration* increases "slightly faster" than the increase in the number of UEs, and in a non-linear fashion, which does not promise a good scalability. In addition, overall *duration* of the procedures poses a QoS violation itself, by leading to the signaling delays that are, e.g., around a hundred times longer than requested QoS delays. But, the results are encouraging when we compare them to results from [34], where IMS session establishment delay is reported as 3.37 seconds, or to results from [35], where session reestablishment delay due to vertical handover is reported as around 2.5 seconds. Moreover, it can be noticed that *duration* improves by over a second when profile conveyance is removed from SIP messages, which could justify the decision to introduce the XDM management. Different mechanisms will be investigated to mitigate the mentioned effects, with a focus on a notable delay introduced by the SCM AS processes. As the SCM AS represents a bottleneck in the current prototype deployment, several of its instances could be employed to serve different UEs and share the load. In addition, the SCM AS should be realized as the session anchor point, regarding the SCC AS, which would suppress the need for end-to-end signaling and may lead to faster reconfiguration procedures.

VI. CONCLUSION

This paper presents an approach to enhancing QoS support in the 3GPP IMS by dynamically producing multimedia session configurations with respect to session and terminal mobility. We propose to introduce an SIP AS that, based on received mobility context information, steers the reconfiguration in the IMS network and produces QoS specifications conforming to the mobility-induced constraints. We design several SIP signaling procedures, which are built upon the 3GPP specifications, to enable the session reconfiguration. We also implement an IMS laboratory prototype and describe a case study, in which QoS is negotiated during session establishment, and successfully adapted when a user replaces her/his UE and changes its access network. An initial performance evaluation of our solution indicates the signaling delay of a few seconds, which is generally unacceptable, but comparison to the similar research results encourages us to investigate different mechanisms in order to reduce this delay. Future work will include additional performance evaluation to address scalability of the solution in the context of various background traffic conditions.

ACKNOWLEDGMENT

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Performance Analysis of DNS64 and NAT64 Solutions

Gábor Lencse, Gábor Takács

Abstract—**The need for DNS64 and NAT64 solutions is introduced and their operation is presented. A test environment for the performance analysis of DNS64 and NAT64 implementations is described. The resource requirements of the implementations are measured. The performance of DNS64 and NAT64 solutions is measured under heavy load conditions to determine if they are safe to be used in a production environment, like the network of an internet service provider.**

***Index Terms*—IPv6 deployment, DNS64, NAT64, performance analysis.**

I. INTRODUCTION

As the Internet Assigned Numbers Authority (IANA) delegated the last five "/8" IPv4 address blocks to the Regional Internet Registries in 2011 [1], and the depletion of the IPv4 address pool of the RIPE NCC (which is responsible for the IPv4 address allocations in Europe) is expected to happen in 2012 [2], the deployment of the IPv6 became inevitable in Europe, too. Internet service providers (ISPs) must urgently take preparations for both providing IPv6 services and the co-existence of the two versions of IP. (Of course, not only ISPs, but also customers (including both private and business customers) have to manage this complex change carefully [10]; and because of its complexity they have to use some integrated approach for the evaluation of all aspects concerning their activities and networks [11].)

In the past years a lot of research was done in the field of IPv6 and important theoretical results were achieved. However, if an ISP plans to introduce IPv6, it is crucial to test the *performance* and the *stability* of the different published solutions and choose the ones that are proven to be suitable.

The co-existence of IPv4 and IPv6 raises many different issues. In the beginning of the deployment of IPv6, the following situation is found to be the most typical: there will be customers that have IPv6 addresses only, and they want to connect to servers still having IPv4 addresses only. (The case of the IPv4 only clients and the IPv6 only servers will be a typical situation in a later phase of the deployment of IPv6.)

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Even though the use of dual stack by any of the parties (client or server) would solve the problem, it is not a feasible solution for ISPs because, on the one hand, they will not be able to provide their customers with IPv4 addresses as they are running out of them soon, and on the other hand, they cannot force third party server operators to use dual stack instead of IPv4 only.

The techniques that an ISP can use for solving the problem of IPv6 only clients and IPv4 only servers are DNS64 [3] and NAT64 [4]. These well-known methods have several implementations and we have carefully chosen some of them to investigate their performance and stability.

The rest of this paper is organized as follows: first, the operation of DNS64 and NAT64 is introduced, second, the selection of the implementations is discussed, third, our test environment is described, fourth, the performance measurement method of DNS64 is detailed, fifth, the DNS64 results are presented and discussed, sixth, the performance measurement method of NAT64 is described, seventh, the NAT64 results are presented and discussed, and finally, our conclusions are given.

II. THE OPERATION OF DNS64 AND NAT64

To enable an IPv6 only client to connect to an IPv4 only server, one needs *DNS64 service* and a *NAT64 gateway*. The operation of the solution is introduced using the following network as an example.

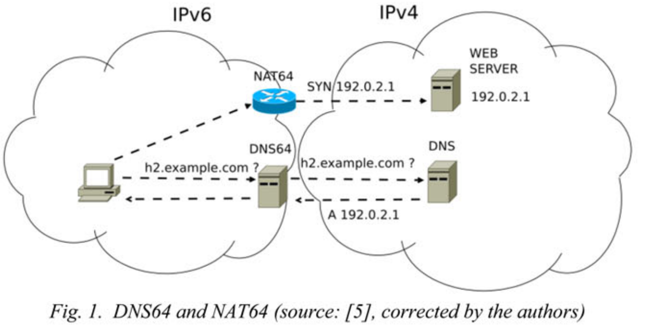

Fig. 1. DNS64 and NAT64 (source: [5], corrected by the authors)

The IPv6 only client (symbolized by a PC on the left side of Fig. 1) wants to connect to an IPv4 only server (symbolized by a web server on the right side of the figure). The *IPv4 only* server means that the DNS system has only an "A" record for the server and no "AAAA" records. A precondition for the operation of the method is that the DNS64 server should be set as the DNS server of the IPv6 only client. When the IPv6 only client tries to connect to the web server, it sends a *recursive query* to the DNS64 server to find the IPv6 address of the web server. The DNS64 server uses the normal DNS system to find out the IP address of the web server.

- If the answer contains an IPv6 address (also) then the DNS64 server returns the IPv6 address as its answer to the recursive query.
- If the answer contains only an IPv4 address then the DNS64 server returns a special IPv6 address; in our example this is the 64:ff9b::/96 prefix plus the 32 bits of the IPv4 address of the web server.

The route towards the network with given IPv6 prefix (in our example, it is 64:ff9b::/96) should be set in the IPv6 only client (and in all of the routers along the route from the client to the NAT64 gateway) to go through the NAT64 gateway.

The IPv6 only client uses the received IPv6 address to set up a connection to the desired (IPv4 only) web server. The client sends a SYN packet to the received IPv6 address. When its SYN packet arrives to the NAT64 gateway, the gateway builds an IPv4 packet using the payload (and some header fields) of the IPv6 packet and it sets the destination address of the IPv4 packet according to the rightmost 32 bits of the destination address of the IPv6 packet. These 32 bits contain exactly the IPv4 address of the desired web server. The source address of the IPv4 packet is set to be the IPv4 address of the NAT64 gateway. The NAT64 gateway sends out the IPv4 packet and it arrives to the IPv4 only server. The IPv4 only server responds the normal way using the source address of the IPv4 packet, that is, the server sends its response to the NAT64 gateway. The gateway receives the IPv4 packet and builds an IPv6 packet using the payload (and some header fields) of the IPv4 packet. The NAT64 gateway sends the IPv6 packet back to the client. (To be able to do this, the NAT64 gateway uses stateful NAT or some other method to track the IPv6 – IPv4 mapping.)

The short example above used the *well-known prefix* described in [6]. In practice, the worldwide use of this prefix has a number of hindrances, see points 3.1 and 3.2 of [6]. For this reason, when implementing a NAT64 gateway, a given size of the subnet is reserved from the actually used IPv6 network. This solution is called *network specific prefix*. In this way, the *Infocommunications Laboratory* of the Department of Telecommunications, Széchenyi István University has got the 2001:738:2c01:8001::/64 network out of which we have reserved 2001:738:2c01:8001:ffff:ffff::/96 as the network specific prefix.

Note that for NAT64, we embed IPv4 addresses into IPv6 addresses: the last 32 bits of the IPv6 address hold the embedded IPv4 address. These kinds of IPv6 addresses are called *IPv4-embedded IPv6 addresses* [6]. There is a further naming convention. Even though their structure and the way of their generation is identical, the IPv4-embedded IPv6 addressed have two further subgroups distinguished on the basis of the purpose of their usage: IPv6 addresses used to represent IPv4 hosts in the IPv6 network are called *IPv4-converted IPv6 addresses*. (They are used in this paper.) The term *IPv4-translatable IPv6 address* is used for an IPv6 address that belongs to an IPv6 host and the purpose of the address translation is to be able to connect to an IPv4 only host. (We are not dealing with this case in this paper.)

III. The Selection of DNS64 and NAT64 Implementations

As BIND, the most widely used DNS implementation, contains native DNS64 support from version 9.8, there was no reason to consider anything else.

As for NAT64 gateways, there are a number of implementations [5]:

- TAYGA is a stateless NAT64 implementation for Linux
- Ecdysis is a NAT64 gateway containing also DNS64
- Microsoft Forefront Unified Access Gateway is a reverse proxy and VPN solution that implements DNS64 and NAT64
- Stateless Network Address Translation 64 runs on Cisco ASR 1000 router
- Stateful NAT64 feature on Juniper MX Series 3D Universal Edge router
- OpenBSD PF packet filter is promised to be NAT64 capable in OpenBSD 5.1

From this seemingly wide selection, finally we were able to test the stability and performance of TAYGA only. Why?

- Ecdysis contains a non official Linux kernel module and it is unfortunately not stable. Ecdysis was tested with version 2.6.32, 2.6.35, 2.6.37 and 3.0.1 kernels and it froze many times. In addition to that, the home page of the project does not reflect any development in the last 15 months [7].
- The Microsoft solution is a small part of a multi function product; the whole product is not at all needed and would be too expensive and resource consuming.
- The solutions running on Cisco or Juniper routers require special hardware that we did not have.
- At the time of our measurements the current OpenBSD release was 5.0.

TAYGA is a free software under GPLv2 license and according to its developers it was intended to provide *production quality NAT64 service* [8]. TAYGA is a *stateless* NAT64 solution. It means that by itself it can create only a one-to-one mapping between IPv6 and IPv4 addresses. For this reason TAYGA is used together with a stateful NAT44 packet filter (`iptables` under Linux): TAYGA maps the source IPv6 addresses to different IPv4 addresses from a suitable size of private IPv4 address range, and from the private IPv4 addresses the stateful NAT44 packet filter performs an SNAT to the IPv4 address of the NAT64 gateway. In the reverse direction, the stateful NAT44 packet filter "knows" which private IPv4 address belongs to the reply packet arriving to the IPv4 interface of the NAT64 gateway.

After the NAT44 translation TAYGA can determine the appropriate IPv6 address using its one-to-one address mapping and then it rewrites the packet to IPv6.

Note that TAYGA is able to store the one-to-one IPv6 – IPv4 address mappings on disk, therefore, in case of a system crash TAYGA can continue using these after restart. On the basis of our experiences with TAYGA we do not think this functionality would be much used.

When configuring TAYGA, a suitably large private IPv4 address range should be provided.

IV. The Test System for DNS64 and NAT64 Performance Measurements

The aim of our tests was to examine the selected programs regarding stability and behaviour under heavy load conditions. (For testing the software, some hardware had to be used, but our aim was not the performance analysis of any hardware.)

A. *The Structure of the System*

A test network was set up in the *Infocommunications Laboratory* of the Department of Telecommunications, Széchenyi István University. The logical topology of the network is shown in Fig. 2. The central element of the network is the DNS64/NAT64 computer. This Linux box played the role of both the DNS64 server and the NAT64 gateway but not simultaneously, but rather one after the other, as we measured the performance of the two systems separately.

For the measurements, we needed a namespace that:

- can be described systematically
- can be resolved to IPv4 only
- can be resolved without delay

The 10-{0..10}-{0..255}-{0..255}.zonat.tilb.sze.hu namespace was used for this purpose. This name space was mapped to the 10.0.0.0 – 10.10.255.255 IPv4 address by the name server at 192.168.100.105. The DNS64 server mapped these IPv4 addresses to the 2001:738:2c01:8001:ffff:ffff:0a00:0000 – 2001:738:2c01:8001:ffff:ffff:0a0a:ffff IPv6 address range.

The IPv6 only workstations at the bottom left corner of the figure played the role of the clients for both the DNS64 and the NAT64 measurements.

During the NAT64 measurements, the TAYGA NAT64 gateway used the 172.16.0.0/12 private IPv4 address range that was SNAT-ed to the 193.224.129.170 IPv4 address.

At the NAT64 gateway, the address of the next hop router towards the 10.0.0.0/8 network was set to 193.224.129.172. (The PC with this IP address responded instead of all of the hosts with IP addresses from the 10.0.0.0/8 network, see more details later on.)

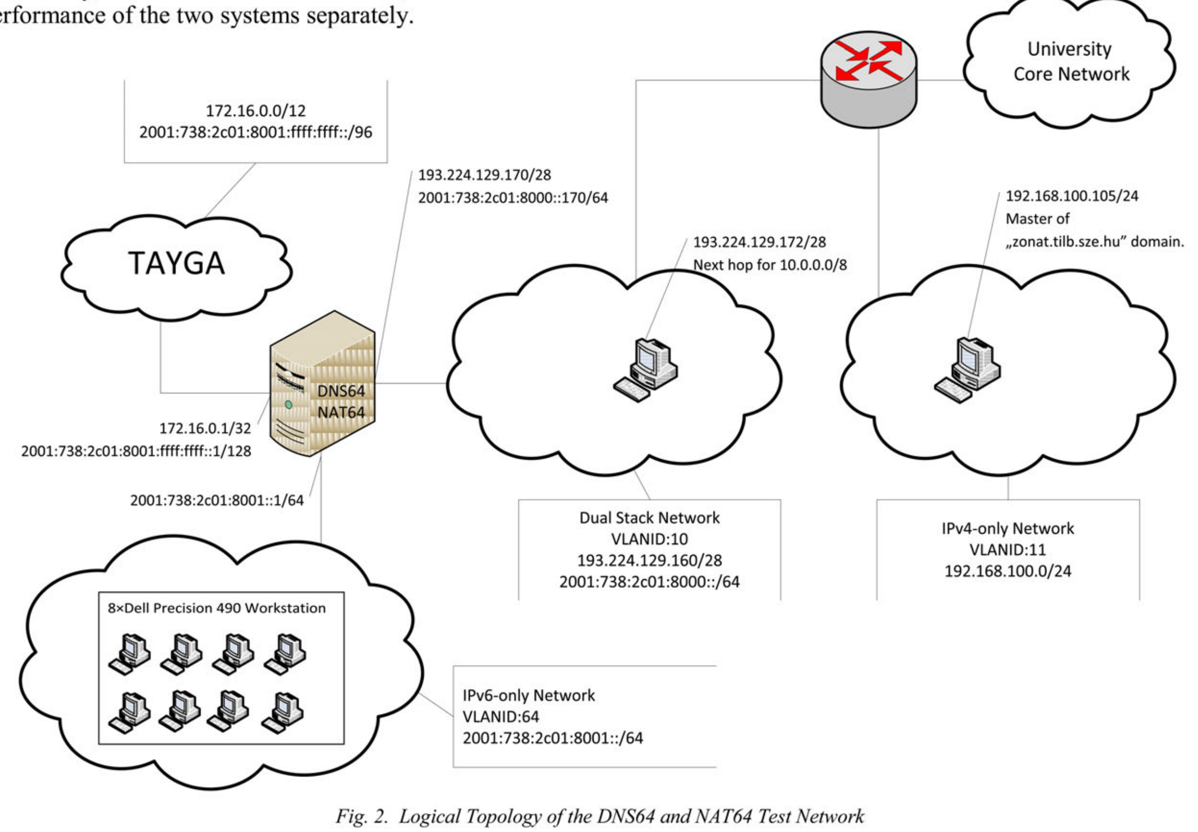

Fig. 2. Logical Topology of the DNS64 and NAT64 Test Network

The physical topology of the system is shown in Fig. 3. It is provided for the purpose that our measurements can be reconstructed and verified. Due to the infrastructural reasons that the client workstations and the DNS64/NAT64 Linux box were in two neighbouring rooms, they were interconnected by two Gigabit Ethernet switches using VLANs. (The VLAN IDs are written to the network in circles.)

B. The Configuration of the Computers

A test computer with special configuration was put together for the purposes of the DNS64 server and the NAT64 gateway in order that the clients will able to produce high enough load for overloading it. The CPU and memory parameters were chosen to be as little as possible from our available hardware base in order to be able to create an overload situation with a finite number of clients, and only the network cards were chosen to be fast enough. The configuration of the test computer was:

- Intel D815EE2U motherboard
- 800 MHz Intel Pentium III (Coppermine) processor
- 128 MB, 133 MHz SDRAM
- Two 3Com 3c940 Gigabit Ethernet NICs

Note that the speed of the Gigabit Ethernet could not be fully utilized due to the limitations of the PCI bus of the motherboard, but the speed was still enough to overload the CPU.

For all the other purposes (the 8 client computers, the IPv4 DNS server and the next hop router towards the 10.0.0.0/8 network) standard *DELL Precision Workstation 490* computers were used with the following configuration:

- DELL 0GU083 motherboard with Intel 5000X chipset
- Two Intel Xeon 5130 2 GHz dual core processors
- 4x1 GB 533 MHz DDR2 SDRAM (Quad Channel)
- Broadcom NetXtreme BCM5752 Gigabit Ethernet controller (PCI Express)

Debian Squeeze 6.0.3 GNU/Linux operating system was installed on all the computers (including the Pentium III test computer, too).

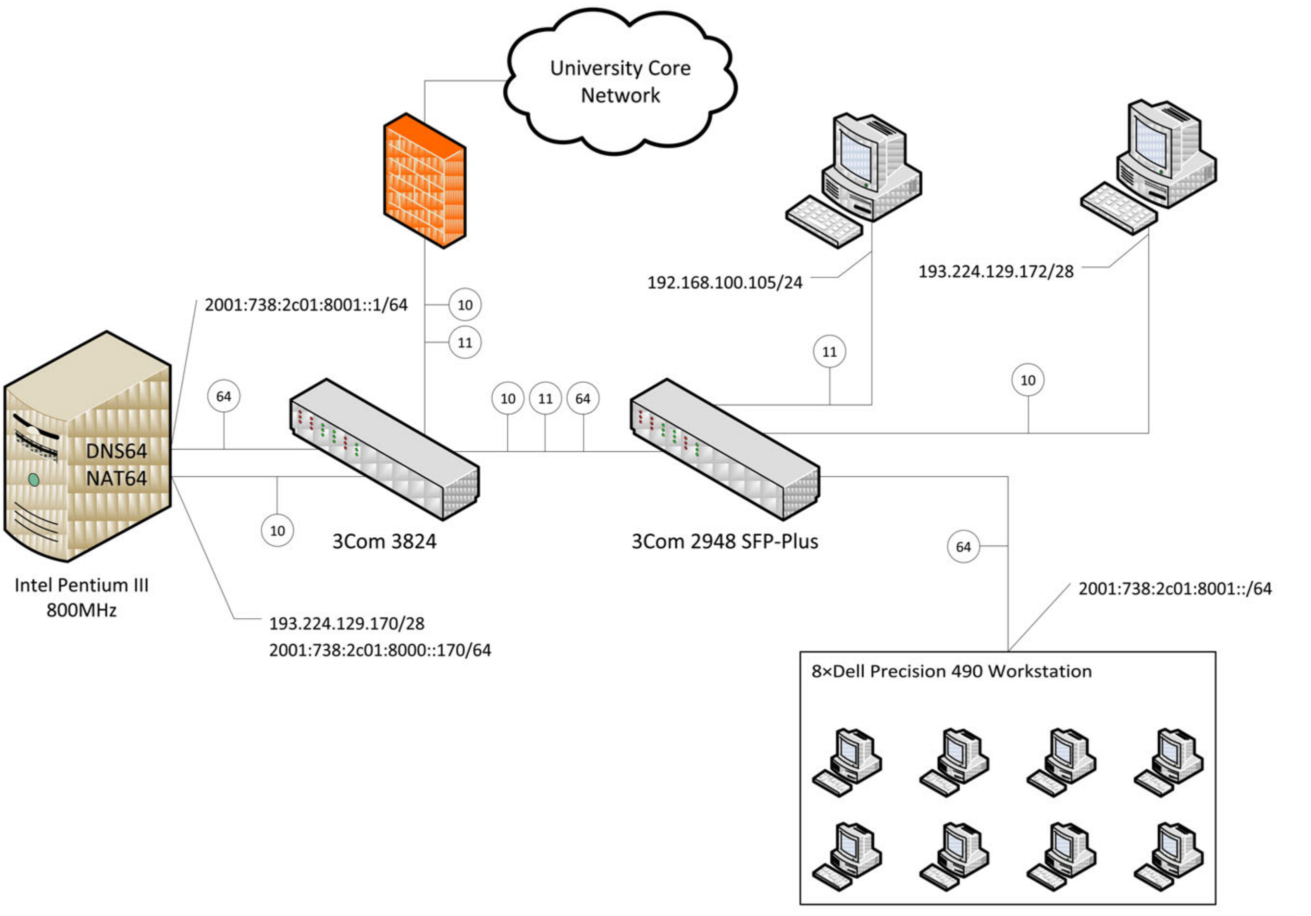

Fig. 3. Physical Topology of the DNS64 *and NAT64 Test Network*

V. DNS64 Performance Measurement Method

A. IPv4 DNS Server Settings

The DNS server was a standard DELL Linux workstation using the 192.168.100.105 IP address and the symbolic name **teacherb.tilb.sze.hu**. The version of BIND was 9.7.3 as this one can be found in the Debian Squeeze distribution and there was no need for special functions (unlike in the case of the DNS64 server).

The 10.0.0.0/16-10.10.0.0/16 IP address range was registered into the **zonat.tilb.sze.hu** zone with the appropriate symbolic names. The zone file was generated by the following script:

```
#!/bin/bash
cat > db.zonat.tilb.sze.hu << EOF
\$ORIGIN zonat.tilb.sze.hu.
\$TTL    1
@ IN SOA teacherb.tilb.sze.hu. kt.tilb.sze.hu. (
                    2012012201 ; Serial
                         28800 ; Refresh
                          7200 ; Retry
                        604800 ; Expire
                             2 ) ; Min TTL

@   86400   IN   NS   teacherb.tilb.sze.hu.

EOF

for a in {0..10}
do
    for b in {0..255}
    do
        echo '$'GENERATE 0-255 10-$a-$b-$ IN A \
             10.$a.$b.$ >> db.zonat.tilb.sze.hu
    done
done

echo "" >> db.zonat.tilb.sze.hu
```

The first general line of the zone file (describing the symbolic name resolution) was the following one:

```
$GENERATE 0-255 10-0-0-$ IN A 10.0.0.$
```

A line of this kind is equivalent with 256 traditional "IN A" lines; the **$GENERATE** directive was used for shorthand purposes.

As it can be seen from the script above and as it has been mentioned earlier, these symbolic names have only "A" records and no "AAAA" records, so the generation of the IPv6 addresses is the task of the DNS64 server.

B. DNS64 Server Settings

The network interfaces of the freshly installed Debian Squeeze Linux operating system on the Pentium III computer were set according to the logical topology shown in Fig. 2.

For the purposes of the DNS64 server, the BIND 9.8 was compiled from source (as the Debian Squeeze did not contain this version yet).

The 2001:738:2c01:8001:ffff:ffff::/96 prefix was set to BIND for the DNS64 function using the **dns64** option in the file **/etc/bind/named.conf.options**.

In order to facilitate the IPv6 SLAAC (*Stateless Address Autoconfiguration*) of the clients, **radvd** (*Router Advertisement Daemon*) was installed on the NAT64 gateway.

The settings in the file **/etc/radvd.conf** were the following:

```
interface eth2
{
        AdvSendAdvert on;
        AdvManagedFlag off;
        AdvSendAdvert on;
        prefix 2001:738:2c01:8001::/64
        {
                AdvOnLink off;
        };
        RDNSS 2001:738:2c01:8001::1 {};
};
```

C. Client Settings

Debian Squeeze was installed for the DELL computers used for client purposes, too. On these computers, the DNS64 server was set as name server in the following way:

```
echo "nameserver 2001:738:2c01:8001::1" > \
     /etc/resolv.conf
```

D. DNS64 Performance Measurements

The CPU and memory consumption of the DNS64 server was measured in the function of the number of requests served. The measure of the load was set by starting test scripts on different number of client computers (1, 2, 4 and 8). In order to avoid the overlapping of the namespaces of the client requests (to eliminate the effect of the DNS caching), the requests from the number **i** client used target addresses from the 10.$i.0.0/16 network. In this way, every client could request 2^{16} different address resolutions. For the appropriate measurement of the execution time, 256 experiments were done and in every single experiment 256 address resolutions were performed using the standard **host** Linux command. The execution time of the experiments was measured by the GNU **time** command. (Note that this command is different from the **time** command of the bash shell.)

The clients used the following script to execute the 256 experiments:

```
#!/bin/bash
i=`cat /etc/hostname|grep -o .$`
rm dns64-$i.txt
do
    for b in {0..255}
    do
        /usr/bin/time -f "%E" -o dns64-$i.txt \
           -a ./dns-st-c.sh $i $b
    done
done
```

The *synchronized start* of the client scripts was done by using the "Send Input to All Sessions" function of the terminal program of KDE (called **Konsole**).

The **dns-st-c.sh** script (taking two parameters) was responsible for executing a single experiment with the resolution of 256 symbolic names:

```
#!/bin/bash
for c in {0..252..4} # that is 64 iterations.
do
    host 10-$1-$2-$c.zonat.tilb.sze.hu &
    host 10-$1-$2-$((c+1)).zonat.tilb.sze.hu &
    host 10-$1-$2-$((c+2)).zonat.tilb.sze.hu &
    host 10-$1-$2-$((c+3)).zonat.tilb.sze.hu
done
```

In every iteration of the **for** cycle, four **host** commands were started, out of which the first three were started asynchronously ("in the background") that is, the four commands were running in parallel; and the core of the cycle was executed 64 times, so altogether 256 host commands were executed. (The client computers had two dual core CPUs that is why four commands were executed in parallel to generate higher load.)

First, a test was performed with one client only. After a while, the *conntrack table* of the *netfilter* of the test computer running DNS64 became full and the name resolution stopped functioning. As DNS64 does not require *netfilter*, the *netfilter* module was removed from the kernel of the computer. After this, the test was completed with no errors.

As a production system may require the presence of **iptables** for security reasons[1], a different solution may be needed. One may increase the size of the *conntrack* table (it is necessary to increase the value of the *hashsize* parameter proportionally, too), or decrease the value of the timeout. As the first one has a resource (memory) requirement, the second one was chosen. The timeout for the UDP packets was decreased from 30s to 1s. The exact name of the changed kernel parameter is:

```
/proc/sys/net/netfilter/nf_conntrack_udp_timeout
```

With this setting, the test was completed with no errors even with 8 clients (that produce much higher load).

Next, the number of clients was increased from one to eight (the used values were: 1, 2, 4 and 8) and the time of the DNS resolution was measured. The *CPU and memory utilization* were also measured on the test computer running DNS64. The following command line was used:

```
dstat -t -c -m -l -p --unix --output load.csv
```

VI. DNS64 PERFORMANCE RESULTS

The most important results were summarized in Table 1. The first row of the table shows the number of clients. (The load of the DNS64 server was increasing in the function of this parameter.) The second row shows the execution time of one experiment (that is the execution of 256 **host** commands). Even though the results showed little deviation, the standard deviation is included in the third row.

[1]Firewalls often use *stateful packet inspection* today (that requires a *conntrack table*); however this solution is highly susceptible to DDoS attacks. The proliferation of DDoS attacks reported in [12] may require the use of the *stateless packet inspection* instead of the *stateful* one.

Rows number four and five show the CPU utilization and the standard deviation of the CPU utilization, respectively.

Row number six shows the estimated memory consumption of DNS64. (This parameter can be measured with high uncertainty, as its value is quite low and other processes than DNS64 may also influence the size of free/used memory of the Linux box.) Fortunately, it can be seen, that its value was always really low.

The size of the *conntrack table* was also logged during the experiments using the **nf_conntrack_count** kernel parameter. Its maximum is displayed in row 7.

The number of DNS64 requests per second, served by the test computer, was calculated using the number of clients (in row 1) and the execution time values (in row 2) and it is displayed in the last row of the table.

TABLE 1. DNS64 PERFORMANCE RESULTS

1	Number of clients	1	2	4	8
2	Exec. time of 256 **host** commands [s]	1,195	1,746	3,643	7,287
3	(standard deviation)	0,074	0,040	0,031	0,050
4	CPU utilization [%]	66,9	97,0	100,0	100,0
5	(standard deviation)	3,8	2,0	0,0	0,0
6	DNS64 memory consumption [MB]	0,5	1,4	2,0	2,4
7	Maximum size of the conntrack table	2347	4989	6474	7493
9	Number of requests served [request/s]	214	293	281	281

On the basis of the results above, we can state that:

- The increase of the load does not cause serious performance degradation and the system does not at all tend to collapse due to overload. Even when the CPU utilization is about 100% the response time increases approximately *linearly* with the load (that is, with the number of clients)
- We cannot give an exact estimation for the memory consumption of DNS64, but it is visibly very low even for extremely high loads.
- It can be seen from the last row of the table that the maximum of the number of requests served was achieved using two clients. The further increase in the number of clients caused only increase in the response time, but the number of requests per second could not increase. It was so, because the test program did not send a new request until all the four **host** commands (running in parallel) received an answer.
- The maximum size of the *conntrack table* increased with the load; it means that one must be aware of the size of the *conntrack table* in the case of a production system.

The results presented above are very important, because they show that the behaviour of the DNS64 system complies with the so called *graceful degradation* [9] principle; if there are not enough resources for serving the requests then the response time of the system increases only *linearly* with the load.

To compare the performance of the DNS64 to the performance of a *caching-only DNS server*, the same series of experiments were performed with the only difference that the DNS64 was switched off in the BIND, so the test computer was functioning as a caching-only DNS server. The results showed that DNS64 needs only a very little more computing power than a caching-only name server. E.g. the DNS64 test with 8 clients (each clients executed 256 **host** commands) lasted 7,287s (std. dev.: 0,050), and the same experiment with the caching-only DNS server lasted 7,009s (std. dev.: 0,036).

VII. NAT64 Performance Measurement Method

A. NAT64 Gateway Settings

The TAYGA system was installed on the Pentium III test computer and it was configured as a NAT64 gateway. The following modifications were done in the **/etc/tayga.conf** file:

```
tun-device nat64
ipv4-addr 172.16.0.1
dynamic-pool 172.16.0.0/12
prefix 2001:738:2c01:8001:ffff:ffff::/96
```

The next hop address for the 10.0.0.0/8 IPv4 network was set to 193.224.129.172.

B. The Settings of the 'Responder' Computer

For the testing of the NAT64 gateway, 'someone' had to answer in the name of the IPv4 only hosts. A DELL computer (the same configuration as the clients) was used for this purpose. This host had the 193.224.129.172 IP address. At the DELL computer, the packets towards the 10.0.0.0/8 network were redirected to the computer itself by the following **iptables** rule:

```
iptables -t nat -A PREROUTING -d 10.0.0.0/8 \
  -j DNAT --to-destination 193.224.129.172
```

As the DELL computer had much more computing power than that of the Pentium III test computer, it was able to answer easily instead of the IPv4 only computers. E.g. in the case of the **ping6** test with 8 clients, the CPU utilization of the DELL computer was under 4%.

C. Client Settings

The DELL computers were used as clients. No DNS server was used, but the clients prepared the necessary IPv4-converted IPv6 addresses for themselves by concatenating the 2001:738:2c01:8001:ffff:ffff::/96 prefix and the appropriate IPv4 addresses from 10.0.0.0/8. At the client computers, the next hop towards the 2001:738:2c01:8001:ffff:ffff::/96 network was set to the IPv6 address of the NAT64 gateway:

```
route add -A inet6 2001:738:2c01:8001:ffff:ffff::/96 \
  gw 2001:738:2c01:8001::1
```

D. NAT64 Performance Measurements

The following script was executed by 1, 2, 4 and 8 clients:

```
#!/bin/bash
i=`cat /etc/hostname | grep -o .$`
for b in {0..255}
do
    rm -r $b
    mkdir $b
    for c in {0..255}
    do
    ping6 -c11 -i0 -q \
      2001:738:2c01:8001:ffff:ffff:10.$i.$b.$c \
      >> $b/nat64p-10-$i-$b-$c
    done
done
```

Using the **ping6 -c11** command, eleven *echo request* ICMPv6 messages were sent to all of the generated IPv6 addresses.

During the preliminary tests the kernel of the NAT64 gateway sent "Neighbour table overflow" messages. (The *neighbour table* is the IPv6 counterpart of the IPv4 ARP cache.) The different limits for the size of the neighbour table were raised as follows:

```
cd /proc/sys/net/ipv6/neigh/default/
echo 4096 > gc_thresh1
echo 8196 > gc_thresh2
echo 16384 > gc_thresh3
```

VIII. NAT64 Performance Results

The results can be found in Table 2. Row 1 shows the number of clients that executed the test script. Rows 2, 3 and 4 show the packet loss ratio, the response time, and the standard deviation of the response time, respectively. The following two rows show the CPU utilization of the test computer and its standard deviation. Row 7 shows the number of packets per seconds that was calculated from the average traffic arriving to the test computer from the direction of the clients measured in bytes. (In the calculations, the size of the messages carrying the ICMPv6 echo requests was always 100 bytes.) The last line show the memory consumption measured at the test computer.

TABLE 2. NAT64 PERFORMANCE RESULTS

1	number of clients	1	2	4	8
2	packet loss [%]	0,025	0,020	0,008	0,025
3	average response time of **ping6** [ms]	0,36	0,37	0,52	1,11
4	(std. deviation)	0,20	0,06	0,11	0,20
5	CPU utilization [%]	26,1	49,6	79,2	93,9
6	(std. deviation)	4,1	5,0	5,6	3,0
7	measure of the traffic [packets/s]	2207	4322	6804	7491
8	NAT64 memory consumption [MB]	4,6	6,2	5,8	3,1

Evaluation of the results:

- Though packet loss occurred even for a single client, the packet loss ratio was always very low (under 0.03 percent).
- The response time showed no increase while the CPU utilization was far from 100% (0.36 and 0.37 for one and two clients, respectively). Later it started growing and for eight clients it was nearly twice as much as it was for four clients.
- The behaviour of the response time is in a good agreement with the changes that can be seen in the number of packets: when the number of clients was increased from one to two, the number of packets were also nearly doubled, the next doubling of the number of clients could cause only 50% increase in the number of packets, and for 8 clients the number of packets grew only a little from 6804 to 7491 as TAYGA could not serve more packets due to the lack of CPU capacity.
- The memory consumption does not show correlation with the load, but it is very low again.

To sum up the findings above, we can lay down that TAYGA performed well, its memory consumption was found to be very low and its response time started growing at high CPU utilization but still remained proportional only with the load, that is, TAYGA also complied with the *graceful degradation* principle.

IX. CONCLUSIONS

A test environment and the methods for the performance analysis of DNS64 and NAT64 solutions were described.

The resource requirements and the performance of the DNS64 support of BIND 9.8 and of the stateful NAT64 solution achieved by the combination of the stateless TAYGA plus iptables NAT44 were measured.

It was found that these implementations are stable even under heavy load conditions and their performance complies with the graceful degradation principle under serious overload.

We conclude that they are safe to be used in a production environment like the network of an internet service provider.

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Dynamic Log Analysis

András Lukács, Zsolt Nagy

Abstract—**This article reviews a new log analysis solution based on multidimensional data cubes. It introduces the process of dynamic log analysis, including real-time log compression during the log management, a new log parsing language, and the efficient regular expression processing engine for this language. The article assesses the new online analytic processing (OLAP) tool implemented for log analysis. The algorithm and software technology developments appearing in the resulting system are creating a new class of real time log processing and analysis tools.**

Index Terms—**log analysis, real-time compression, regular expression, OLAP, bitmap index**

I. Introduction

The article introduces the log analysis tool developed by KÜRT Co. and its technology. This is a professional software system capable of analyzing log data generated by large complex IT systems.

Companies and organizations put an increasing emphasis on the protection of their data and information systems. Log analysis is a technology-intensive method of IT security services. Log files describing the operation of computers, networks and applications are constantly collected. Logs tell the user what has happened to which system or device and when the event occurred. Logs data allow important events to be recognized like attacks to the network, policy violence, fraud, and technical defaults can also be detected and predicted. Last, but not least, logs may be suitable for detailed tracking of business processes.

As logs are generated in large quantities even in middle sized IT systems (they record as many as a million events per second), the collection and the storage of logs for longer than a few days can pose a serious problem. Processing the accumulated data is a long lasting and resource intensive challenge. Generally, only a hundred logged events out of a billion hold information about an incident, and only two of these require intervention; thus an analysis system is required that is able to process this large amount of logs automatically by efficient pattern recognition and filtering.

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Current log analysis systems are based on monitoring the frequency of events (incidents), and the co-occurrence of particular previously defined events and simple rules. It often takes several hours to produce alerts and analysis results because of the limited speed of log processing methodology hindering timely intervention and damage prevention or even making them completely impossible. To solve these problems, we designed a completely novel concept of log analysis compared to previous solutions. The presented algorithm and software technology developments appearing in our system lead to a new class of real time log processing and analysis tools.

The concept of dynamic log analysis bridges rule based analysis tools of the past decade and future solutions based on fully automatic pattern recognition and semantics. The main idea of dynamic log analysis places human professionals in the center of decision making and supports them by (semi)automatic tools in every task.

Implementing the concept of dynamic log analysis involves many technological challenges. The first problem is the immense amount of log data which serve as the basis of the analysis. It is not uncommon to see an IT system producing more than 100 TB of log data every year. Processing such a large amount of data is clearly a *big data* problem [1], [2]. The second issue is how to read the state of the IT system from the logs during the (risk-) analysis. A further important criterion is the openness of the log analysis system. The analysis system has to be capable of receiving and understanding various types of log data, while on the other hand it has to be able to represent the new risks universally. It is also important that the reports and graphs must be understandable by professionals, and they have to be easily added to the existing processes of risk analysis.

The second section of the article describes the necessary steps for log collection and log normalization, which include real time compression, a new language for log processing with its general regular expressions engine effectively processing complex events composed by multiple log lines. The third section introduces the online analytic processing (OLAP) solution used for the analysis, which – besides the ordinary functions – is able to retrieve the original log lines belonging to the queried cell due to a multi-level indexing technique.

A unique feature of the developed OLAP engine is the extremely fast query response time for the regular OLAP operations, and for the log line retrieval too. The fourth section deals with analysis techniques based on the data stored in the OLAP cube. Finally in the fifth section we sum up the novelties of the log analysis tool, and we mention its use cases.

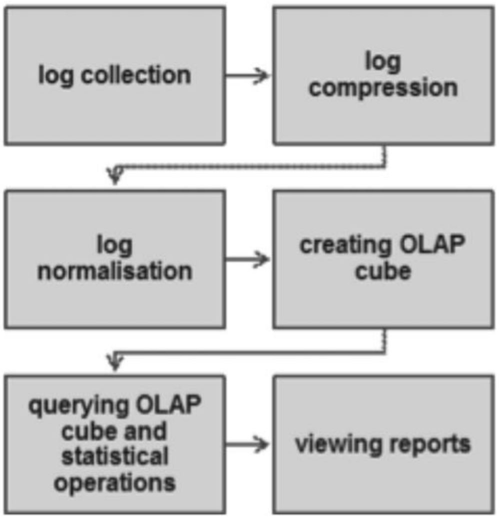

Fig. 1. Proposed process of dynamic log analysis.

II. Log collection and log normalization

The collection process of logs generated by IT systems is supported by many well developed solutions [3]. Thus the first step in log processing is the long term storage of collected logs while maintaining efficient data access. Most log collection systems apply lossless compressors, for example *gzip*, to store the log files in a compressed format. Although such solutions can compress the log data to the tenth in size and so the reading speed from disk can increase with an order of magnitude, the data access speed of a log analysis tool also depends on the speed of decompression. Since compression rate and (de)compression speed are basically inversely proportional, it is a non-trivial problem to find the optimal solution for data compression and implementation, respectively.

The data access speeds can be well characterized with the achieved acceleration for data access speed of compressed logs compared to the data access speed of uncompressed logs. We have examined several real-time, extreme high-speed compressors, and *lz4* [4] showed the best results, which is optimized for data access based on a Lempel-Ziv algorithm. This comparison also included standard *gzip* (version 1.3.12) [5], *lzop* (version 1.02rc1) based on Lempel-Ziv-Oberhumer algorithm to optimize decompression [6], and *pigz* [7], which utilises the gzip parallel multi core architectures effectively.

Collected and stored logs are generated by heterogeneous devices (different operation systems, network devices, applications), thus the log formats are not standard; there are thousands of different log types to work with. On the other hand, logs are usually simple text files with information pieces separated by a character; they generally do not hold any deeper structures (like nested parentheses). This means that log parsing is typically a text processing problem. During this process, it has to be decided whether an input log matches a previous format or not, and then the needed data must be retrieved from the log for further processing. Because of the heterogenity of logs, the log analysis system must be easily expandable with new log formats, and log format descriptors created in an environment must be easily transferable to other similar environments.

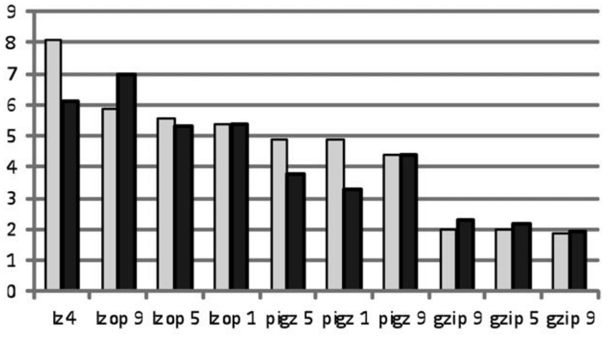

Fig. 2. Data access speed achieved by the use of different compressors evaluated on two data sets (light and dark columns). The height of columns show the achieved acceleration rate on compressed logs compared to the data access speed of uncompressed logs. Numbers behind gzip, lzop and pigz indicate the applied compression ratio (1 is the smallest and fastest; 9 is the largest and slowest). The compression speed of lzop 9 és gzip 9 is substantially worse than the results for other compressors.

Empirical evidence shows that log formats can be described with regular grammars like formal languages [8]. Furthermore it is known that a deterministic finite automaton (DFA) can be created for every regular language, and this automaton allows words belonging to a certain language to be recognized in linear time [9]. This theoretical solution also ensures that these automata are actually feasible in practice, and that they scale adequately in case of grammars describing log formats. The next step of pre-processing involves a further difficulty, since the data actually needed for the analysis must be retrieved from the log lines. In that phase, the filtering and transformation steps determined by professionals give the opportunity to introduce available risk analyst knowledge into the log analysis tool. Although the recognition of defined patterns can be tackled by the DFA-based algorithms, the extraction of a recognized pattern in itself surpasses the boundaries of DFA-based approaches. Therefore, the empirical testing of the available implementations for evaluating regular expressions was necessary.

Examining multiple regular expression processing softwares, we came to the following conclusions. The *flex* (Fast Lexical Analyzer) [10] is well suited to log format recognition. Flex provides the user with the opportunity to build a DFA from several regular expressions, which enables quick recognition. However, a technical problem is posed by the limited size of useable DFA, and the code does not support UTF8 coding, therefore the source code might require some modifications. A further problem of the conditional rules applicable to the extraction of patterns required to the analysis is that these rules reduce the speed of processing drastically.

Google *RE2* [11] is a regular expression matching library based on a highly efficient automaton theory. It offers high speed DFA and NFA (Non-deterministic finite automaton) based analysis. In case of NFA analysis, it is possible to retrieve parts from the input. In DFA mode, the matching algorithm reads through the input once, pacing the automaton by characters. If we would like to retrieve the data, the DFA matching will not be appropriate. In that case NFA matching will be done, but the speed will be reduced to such an extent that it will be inadequate for log analysis.

The RE2 offered a good basis for the development of a prototype suitable for matching numerous samples simultaneously. The Set interface of RE2 made it possible to match multiple regular expressions with one DFA simultaneously. The speed of matching several thousand patterns simultaneously with such a DFA was similar to the speed of recognizing only one pattern. Although the size of the automaton can grow exponentially with the number of patterns, the size of a DFA built from almost a hundred thousand patterns still remains under 1 GB.

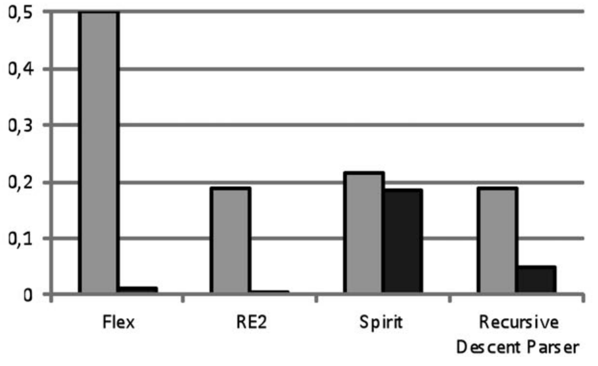

Fig. 3. Comparison of solutions applicable during the log normalization step. Lighter columns show the speed of pattern recognition, darker columns show the speed of pattern retrieval measured in GB/s when processing the test data set.

The *Spirit* [12] template metaprogram library developed for text parsing is part of the Boost program library [14]. With the help of Spirit, parsers can be generated in compile time for environment independent and regular language classes. An advantage of that is the possibility to optimize the parser during the translation, which may result in a significant increase in speed. The Spirit uses the recursive descent parser strategy [15], which - complemented by a state of the art programming approach – proved to be efficient in log format recognition and retrieval, too.

As text parsing rules are defined by a professional, a C++ translation step is required in case of Spirit for creating an actual text parsing program; this is not acceptable because the special (e.g.: legal) requirements of log analysis demand a high degree of stability. Although the compilation of the parser is feasible during run-time, optimizations during the translation were not possible in that case, so the speed decreased significantly. This led to the re-implementation of the recursive descent parser used by Spirit. The processing speed fell to the quarter, however, the resulting speed of text parsing and aggregation – measured in EPS (events per second) unit used by log analysis tools – was around 100 000 EPS on one core evaluated on a test data sample. This performance is comparable with the performance of the fastest log analysis tools available on the market. At the same time the log analysis tool has its own modular descriptive language designed for log analysis; highly efficient analyzers can be compiled during run-time.

III. Means of achieving the data model

The starting point of the data model used in the log analysis is the multidimensional data cube applied in the OLAP (online analytical processing) tools [16]. Log records are represented in an aggregated form in the data cube. The log lines are represented with numbers in the data cube. Each number shows the corresponding counts of log lines where all special fields are the same. This comes from the experience of log analysis professionals; counts and other derived data based on counts (like statistics) provide excellent input for efficient analysis. Compared to the amount of log lines, there are generally much less nonzero numbers in the data cubes, so the data cube is a significant compression of the parsed log data. An important advantage of the multidimensional data model is the hierarchy of dimensions, which appears naturally in most cases, and can be exploited during the analysis. An example for this is the date, where the structure is the following: month, day, hour, minute etc. An extension compared to the ordinary OLAP implementations is that for each cell the corresponding log lines can be searched and retrieved due to a multiple inverse indexing.

Although the idea to use multidimensional data cubes and OLAP tools for special types of log lines occurred at the end of the second millennium especially in case of web server logs [17] [18], the OLAP has not been used yet for general purpose log analysis. The main reason for that might be the insufficient scaling of the available OLAP engines on big data. Nevertheless an early and sporadic source mentioning the use of OLAP tools in log analysis is worth noting [19].

In course of implementing the multidimensional data model of the log analysis tool, we posed and examined the question of applicability for several existing OLAP tools. We examined the applicability of OLAP implemented with relational database manager (ROLAP), and the use of memory resident OLAP (MOLAP) in log analysis. The examined ROLAP solution was an application executing the required OLAP functions needed for log analysis. This ROLAP solution is based on a MySQL [20] relational database manager. During the tests we used the *icCube* MOLAP tool [21]. As the performance of the tested tools was not sufficient for using it in a dynamic log analysis tool, a new OLAP engine implementation made it possible to fully meet the diverse needs (Fig. 4.)

The implemented OLAP engine (*Colap v1*) can quickly answer the count cube queries with the help of so-called

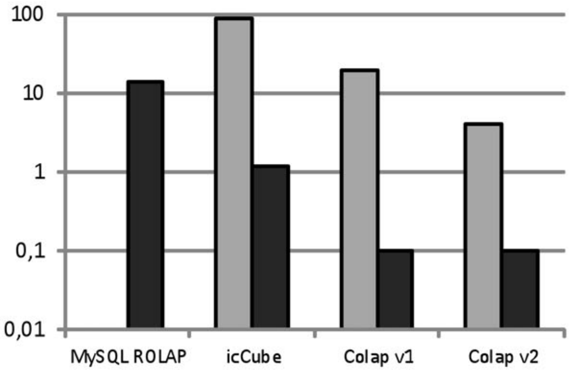

Fig. 4. Comparison of performances measured with different OLAP engines. The test data set was a log file with 6.5 million lines, it was 1.5 GB in raw size. Light columns indicate the loading speed of the gzip compressed test data from the memory to the OLAP tool and the building time of the data cube measured in seconds. Dark columns indicate the average time of OLAP operations on the built-up data cube measured in seconds. In case of MySQL ROLAP, the data loading time is not applicable.

compressed bitmap indexes. On one hand, the bitmap index is an incidence vector built upon all of the possible values of all the dimensions as coordinates; this vector describes one particular cell of the data cube by giving the values of the defining dimensions. On the other hand, the value of the corresponding cell also appears as the last element of the vector. This model is called indexed count cube model [21]. The bitmap index is a true representation of the cube. Every OLAP query can be traced back to the product of AND operations on the bit vectors of indexed count cube and of additions on the count values of cube record set. Vectors can be stored in a compressed format, AND operations can be done effectively even in this compressed format.

There are numerous methods for compressing bitmap indexes. Most solutions are based on run-length encoding, where sequences of the same values following each other in the bit vector are given with two data; the element of the sequence and the number of repetitions. The Byte-aligned Bitmap Code is used in most relational database management system. Another option is the application of Word-Aligned Hybrid (WAH) coding [22]. In case of particular fields like time, columns based bitmap vector encoding increases speed significantly [23]. This solution is the base of the OLAP engine version implemented by us, which could be further improved in terms of operation time by fine tuning the applied encoding (*Colap v2*).

IV. Analysis

Aggregated data cubes created from logs are the basis of the next steps in the analysis. Because of the diversity of log source systems, other tools were needed for the expansion and merging of the data cubes. Several operations available on multidimensional data cubes were implemented for the final phase of the analysis. Some of these are regular OLAP operations, others are statistical, event pattern comparison tools. What these tools have in common is that they create a low (two or three) dimension aggregate of the data cube storing the most detailed information, which thus can be visualized. OLAP queries can be done in a language similar to MDX (MultiDimensional Expressions) query language [25].

The log analysis tool can be managed via a web page, which is also the locus of displaying generated reports and visualizations (Fig. 5.). The number of feasible reports are inexhaustible, essentially all the information demanded by the analysts can be supplied by the system in real time. With the aid of the reporting system, one can configure other services starting from alerts to the periodic summary reports through the interface for defining reports.

The attained OLAP engine of the log analysis tool makes interactive analysis possible even when dealing with large data sets. Operations on the data cube are finished typically under one second. If the analyst wishes to look at the original log lines from the count cube cells, the indexing technology makes it possible to reach them under one second no matter how large the data set is.

Fig. 5. A typical screen image from the analysis module of the system: With color mapping, the heat map visualizes the frequencies of log patterns given in the data cube in hourly breakdown. The lighter (originally red) color indicates large number of occurrences, the darker (originally blue) indicates low occurrence rate or no occurrences. The sample represents information from logs collected about a week long period.

V. CONCLUSION

In the article we introduced a new log analysis solution based on multidimensional data cubes. We described the structure of the log analysis tool and explained the new technologies in modules attaining some steps in log management. According to that, we described in detail the real-time compression, the new log processing language able to represent professional knowledge for log normalization with the corresponding regular expressions, and the engine processing complex events described by multiple log lines. A new solution is the online analytic processing (OLAP) used for the analysis. Important features of the developed OLAP engine is the extremely fast response time for the queries, for regular OLAP cube operations, and also for data retrieval.

The presented log analysis tool offers effective monitoring technology for all companies using information technology extensively. Possible use cases cover multiple areas from traditional security monitoring, through real time exploration of operational risks, to business intelligence analysis of the monitored IT devices, helping to enhance efficiency.

Methods and algorithms developed during our research might be utilized in many areas outside log analysis. One of the promising alternative use cases is the processing of large amount of short text messages, or the analysis of extreme large data sets, which allows us to analyze the general behavior of many systems outside the field of informatics.

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Call for Papers
Special Issue on Applied Cryptography

This special issue will focus on the area of applied cryptography, bringing up selected papers from SantaCrypt 2012. Santa's Crypto Get-Together (SantaCrypt) started in 2001 as the first annual Czech and Slovak workshop aiming to facilitate closer cooperation of professionals working in the field of applied cryptography and related areas of security. This get-together of experts is organised in order to foster exchange of information and ideas on past, ongoing, and also future projects. The special issue of the Infocommunications Journal will focus on the following areas:

- Applied Cryptography
- Practical Cryptanalysis
- Cryptographic Protocols
- Security Mechanisms Deploying Cryptography

Submissions to Santacrypt 2012 should be mailed to matyas AT fi.muni.cz, and clearly marked with Subject “SantaCrypt 2012”. The manuscripts should follow the MKB format (http://mkb.buslab.org/cfp.htm.en) with the maximum length of 10 pages (corresponds to 8 pages in the IEEE format). The final deadline for the submissions is 1st October 2012. Submissions will be evaluated by the program committee and authors will be informed about the evaluation results by 29th October. Camera-ready versions for the workshop proceedings have to be delivered by 12th November. The workshop takes place in Prague, Czechia, on November 29-30, 2012.

No more than 4 papers from the workshop shall be then selected for the special issue of the Infocommunications Journal, and authors of these papers will have the opportunity to revise their papers (including typesetting in the IEEE format) after the workshop – final versions for the special issue will be due December 9.

Guest Editors:

VÁCLAV (VASHEK) MATYÁS is a Professor at the Masaryk University, Brno, Czech Republic, and serves as a Vice-Dean for Foreign Affairs and External Relations, Faculty of Informatics. His research interests relate to applied cryptography and security, publishing over a hundred peer-reviewed papers and articles, and co-authoring six books. He was a Fulbright Visiting Scholar with Harvard University, Center for Research on Computation and Society, and also worked with Microsoft Research Cambridge, University College Dublin, Ubilab at UBS AG, and was a Royal Society Postdoctoral Fellow with the Cambridge University Computer Lab. Vashek was one of the Editors-in-Chief of the Identity in the Information Society journal, and he also edited the Computer and Communications Security Reviews, and worked on the development of Common Criteria and with ISO/IEC JTC1 SC27. Vashek is a member of the Editorial Board of the Infocommunications Journal. He received his PhD degree from Masaryk University, Brno and can be contacted at *matyas@fi.muni.cz.*

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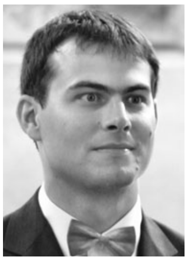

PETR SVENDA is an Assistant Professor at the Masaryk University, Faculty of Informatics, in Brno, Czech Republic. He received his PhD degree from Masaryk University, working in the area of the cryptographic protocols for restricted environments such as wireless sensor networks, with focus on automatic generation of cryptographic protocols with the help of evolutionary algorithms. In 2008, he worked at TU Dresden on secure logging for the AN.ON anonymity service. He is also interested in practical aspects of security in cryptographic smartcards and their resistance against side-channel attacks and properties of random number generators available on smartcards and mobile devices. Petr can be contacted at *svenda@fi.muni.cz.*

Call for Papers
Special Issue on Quantum Communications

Although quantum computers are going to be the applications of the far future, there are already a few algorithms to solve problems which are very difficult to handle with traditional computers. Quantum computing is based on various quantum effects in physics and offers revolutionary solutions for different problems e.g., prime factorization, searching in unsorted database, key distribution and information coding. The power of quantum parallelism allows us to solve classically complex problems, and the quantum entanglement leads to quantum communication algorithms like teleportation and super dense coding. The quantum cryptography provides new ways to transmit information with unconditional security by using different quantum key distribution protocols. The special issue of Infocommunications Journal will focus on quantum communications with the following scope:

- Quantum Communications
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- Quantum Cryptography
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- Quantum Programming Languages and Semantics

Authors are requested to send their manuscripts via electronic mail to László Bacsárdi at *bacsardi@hit.bme.hu* until 30 September, 2012. The manuscripts should follow the IEEE format with maximum length of 8 journal pages.

Guest Editors:

OZGUR B. AKAN (M'2000–SM'2007) received the B.S. and M.S. degrees in electrical and electronics engineering from Bilkent University and Middle East Technical University, Ankara, Turkey, in 1999 and 2001, respectively, and the Ph.D. degree in electrical and computer engineering from the Broadband and Wireless Networking Laboratory, School of Electrical and Computer Engineering, Georgia Institute of Technology, Atlanta, in 2004.
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LÁSZLÓ BACSÁRDI obtained his MSc degree in computer engineering from Budapest University of Technology and Economics (BME) in 2006. He holds an assistant professor position at the University of West Hungary. He wrote his PhD Thesis on the possible connection between space communications and quantum communications at the BME Department of Telecommunications. His current research interests are in mobile ad hoc communication, quantum computing and quantum communications. He is Secretary General of the Hungarian Astronautical Society (MANT), which is the oldest Hungarian non-profit space association founded in 1956. He is member of the board of a Hungarian scientific newspaper ('World of Nature') and he is the publisher of a non-profit Hungarian space news portal ('Space World'). Furthermore he is member of IEEE and HTE. He has joined the Space Generation Advisory Council (SGAC) as well, and is currently active as the Hungarian National Point of Contact.

SÁNDOR IMRE was born in Budapest in 1969. He received the M.Sc. degree in Electrical Engineering from the Budapest University of Technology (BME) in 1993. Next he started his Ph.D. studies at BME and obtained dr. univ. degree in 1996, Ph.D. degree in 1999 and DSc degree from the Hungarian Academy of Sciences in 2007. Currently he is carrying his activities as Professor and Head of Dept. of Telecommunications. He is chairman of Telecommunication Scientific Committee of the Hungarian Academy of Sciences. He participates the Editorial Board of two journals: Infocommunications Journal and Hungarian Telecommunications. He was invited to join the Mobile Innovation Centre as R&D director in 2005. His research interest includes mobile and wireless systems, quantum computing and communications. Especially he has contributions on different wireless access technologies, mobility protocols, security and privacy, reconfigurable systems, quantum computing based algorithms and protocols.

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Format of a book reference:

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Figures should be black-and-white, clear, and drawn by the authors. Do not use figures or pictures downloaded from the Internet. Figures and pictures should be submitted also as separate files. Captions are obligatory. Within the text, references should be made by figure numbers, e.g. "see Fig. 2."
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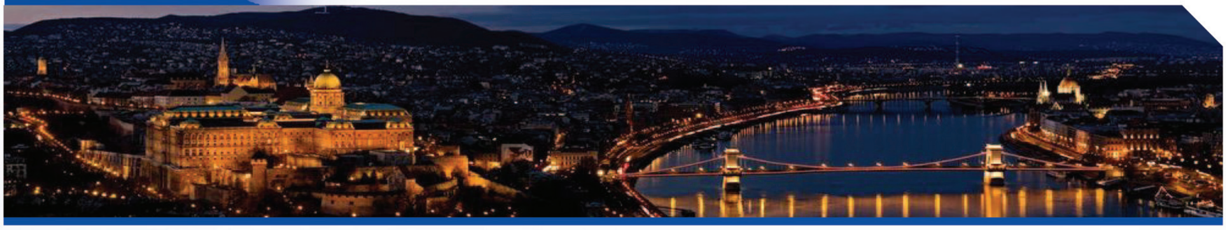

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SCIENTIFIC ASSOCIATION FOR INFOCOMMUNICATIONS

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Founded in 1949, the Scientific Association for Infocommunications (formerly known as Scientific Society for Telecommunications) is a voluntary and autonomous professional society of engineers and economists, researchers and businessmen, managers and educational, regulatory and other professionals working in the fields of telecommunications, broadcasting, electronics, information and media technologies in Hungary.

Besides its more than 1300 individual members, the Scientific Association for Infocommunications (in Hungarian: HÍRKÖZLÉSI ÉS INFORMATIKAI TUDOMÁNYOS EGYESÜLET, HTE) has more than 60 corporate members as well. Among them there are large companies and small-and-medium enterprises with industrial, trade, service-providing, research and development activities, as well as educational institutions and research centers.

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