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Analysis of the Scalability of the Overlay Skype System

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Abstract **—In this paper, we study the Skype system in order to determine the presence of bottleneck for its scalability. In fact, as the Internet grows and the number of Skype users increases constantly, the Skype traffic carried by network links grows and also the load of Skype supernodes increases, mainly due to the need of relaying calls for users behind NAT restrictions. We model both the underlay Internet at the inter-Autonomous System level and the overlay Skype system. We let the two systems interact through a computer implementation of their respective models and, by varying critical parameters such as the size of the underlay Internet, the number of Skype users, the per-user traffic and the percentage of users behind NAT restrictions, we calculate the sensitivity of link load and of the supernode load to these parameters. In this way, we are able to determine in which conditions link load and supernode load may become a bottleneck for the scalability of the Skype system**

I. INTRODUCTION

During the last few years telephony has experienced a radical change with the advent of the peer-to-peer (p2p) VoIP technology, which has determined the passage of telephony from a classical model in which it has been a service exclusively supplied by operators to a less structured model in which users can place telephone call directly through the Internet. The popularity of p2p telephony comes from the smaller costs for end-users. The expansion rate of p2p telephony is constantly increasing and telephone operators are currently studying business strategies to cope with p2p telephony.

The most popular p2p telephone system is Skype. Skype users download a VoIP p2p client, whose first version dates back to 2003. Skype allows users to place free telephone calls through the Internet and it offers economic interconnection with traditional Public Switched Telephone Networks and mobile telephone networks at competitive rates. Since its birth, the success of Skype has grown more and more and currently there are more than 195 million registered users.

The remarkable success and impact of Skype has started an intense research activity with the purpose of identifying the main characteristics of Skype (Skype is a closed product, its features are not disclosed and its source code is not available to the public).

Schultzrinne in [1] analyzed key Skype functions including NAT/firewall traversal features and supernodes' relay functions. Guha et al. in [2] studied the population of online clients, the number of supernodes, their traffic characteristics, providing experimental data useful for future design and modeling of p2p VoIP systems. Yanfeng Yu et al. in [3] study the Skype system in order to identify P2P traffic and carry out measures on the Skype overlay network with a special ad-hoc tool.

An important issue concerning the Skype system is its potential scalability. The main question is whether Skype has the chance of growing without limits or if there are intrinsic bottlenecks that may prevent an unlimited growth. The purpose of this paper is to study and analyze the interaction of the overlay Skype network with the underlay Internet, with the objective of understanding how Skype utilizes the network and system resources. We identify two main potential factors that may impair the scalability of Skype: the traffic carried by network links and the number of supernodes required for performing the call-relaying service with adequate quality. In fact, the traffic carried by network links increases as the size of the Internet and of the Skype population grows and also the rate of call-relaying requests rises if network size, number of users and per-user traffic grow.

In order to determine if traffic requirements or supernode load can affect the scalability of Skype, we have built an accurate model of the underlay Internet at the inter-Autonomous System level, moreover, we have modeled the Skype overlay network. With a computer implementation of these models we let the two systems interact and, by growing the size of the Internet, the number of Skype users, the traffic per user, the percentage of users behind NAT restrictions and other critical parameters, we measure the system performance and we determine if Skype is able of scaling in all conditions.

The rest of the paper is organized as follows. In section II the Internet topology is analyzed, and its power-law statistical characteristics are presented. Furthermore, we describe the *Positive-Feedback Preference* (PFP) [4] algorithm, one of the most precise and complete Internet topology generators currently available. In Section III we give a brief description of the Skype topology and of its model in our numerical model. In section IV the simulation algorithm is described by illustrat-

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ing its phases: (1) underlay network generation using the PFP algorithm; (2) overlay network generation; (3) computation of carried traffic; (4) accounting for the call-relaying by supernodes for users behind NAT restriction; (5) measurement of performance metrics. In section V the results of the simulative analysis is presented. Conclusions are given in section VI.

II. UNDERLAY INTERNET MODELIZATION

In the last thirty years, the Internet has experienced a fascinating evolution, with an exponential growth of traffic and size. Focusing on the Internet topology at the Autonomous Systems (AS) level, which is associated with the Internet BGP traffic routing, Faloutsos [5] reported, in 1999, that the Internet topology is characterized by a power-law distribution of the degree of ASs, showing that in a log-log scale the degree,(i.e., the number of links to external ASs) of an Autonomous System is proportional to its rank in the decreasing degree sequence. Two other power-laws, concerning the complementary cumulative distribution function (CCDF) of the AS degree and the eigenvalues of the adjacency matrix of the inter-AS graph are presented by Faloutsos. This discovery can be considered as a revolution, as previous studies on Internet topology models were based on random graphs [6] featuring a Poisson distribution of degree. In [5] it is also shown how these topological properties are approximately constant, in a time frame of several years.

Zhou and Mondragòn, in 2005, proposed their *Positive-Feedback Preference* (PFP) [4], an Internet topology generator able to produce the three power-laws of Internet graphs as well as other significant topological characteristics such as the rich-club phenomenon and the disassortative-mixing. The disassortative-mixing property of Internet graphs means that low-degree nodes tend to connect with high-degree nodes and rich-club connectivity means that high-degree nodes tend to form a tightly interconnected clique with nodes with equal or higher degrees.

The PFP graph generation algorithm starts with a small random network and, at each time step:

- with probability $p \in [0,1]$ a new node is attached to one host node, and one new internal link appears between the host node and a peer node;
- with probability $q \in [0, 1-p]$ a new node is attached to one host node, and two new internal links appear between the host node and two peer nodes;
- with probability $1-p-q$ a new node is attached to two host nodes, and one new internal link appears between one of the host nodes and one peer node.

For $p = 0.3$ and $q = 0.1$, the PFP model produces the same ratio of nodes to links as the inter-AS graph of the real Internet. An important property of the PFP algorithm is that, when choosing host nodes and peer nodes, it adopts a unique nonlinear preference probability defined as:

$$\Pi(i) = \frac{k_i^{1+\delta \log_{10} k_i}}{\sum_j k_j^{1+\delta \log_{10} k_j}}, \delta \in [0,1] \qquad (1)$$

This is called the Positive-Feedback Preference because a node's ability of acquiring new links increases as a non-linear feed-back loop. From numerical simulations, $\delta = 0.048$ produces the best match between the synthetic graphs and the real inter-AS Internet graphs.

III. OVERLAY SKYPE MODELIZATION

Skype [1] is an overlay peer-to-peer network composed of two types of nodes: ordinary hosts and supernodes. An ordinary host is a Skype client that can place and receive voice calls and send/receive text messages. Each node with a public IP address and enough CPU, memory, and network bandwidth can become a supernode. Supernodes form a higher-level overlay network. Supernodes are used by ordinary hosts to search other users and one of the most important functions performed by supernodes is the relaying of media and signaling for users behind port-restricted Network Address Translators (NATs). The Skype network is also provided with gateways, used by the SkypeIn and SkypeOut services for the interconnection with the traditional PSTN or mobile network.

It is difficult to replay the real distribution of Skype users (ordinary hosts) inside an AS, because the available information on the actual number users per AS as a function of the AS's degree is not available. In this work, we use three kinds of statistical distribution of Skype users inside an AS, namely, *constant*, with the same number of users for all ASs, *gravitational*, with a number of user directly proportional to the degree of the AS, and *uniform*, with a number of users per AS between 0 and twice the value of the constant distribution. We define N_o as the total number of ordinary hosts, N as the total number of ASs, O as the vector whose element O_i is the number of ordinary host of the ith AS and D is the vector whose element D_i is the degree of the ith AS. According to these definitions, the number of Skype users for the ith AS are calculated with equations (2), for the *constant* distribution, and (3), for the *gravitational* distribution. In the case of *uniform* distribution a random number of users is selected.

$$O_i = \frac{N_o}{N} \qquad (2)$$

$$O_i = \frac{N_o \cdot D_i}{\sum_j D_j} \qquad (3)$$

It is also difficult to replay the real distribution of supernodes among the Autonomous Systems. However, it is important to note that an ordinary host becomes a supernode depending on its connection capacity and processing characteristics, independently on the AS importance as a function of its degree. Therefore, we have used a uniform random distribution of supernodes among ASs. Finally, we generate a number of Skype gateways, N_g, equal to 1% of the total number of ASs, and we distribute them randomly among the ASs.

IV. THE SIMULATION ALGORITHM

The first step of our simulation algorithm is the creation of the adjacency matrix V of the inter-AS graph with the PFP Internet topology generator. Secondly, we generate the inter-AS routing matrix R by applying the Dijkstra's shortest path algorithm to the matrix V. The third step consists in the generation of the traffic matrix and, to this purpose, we have identified seven relevant types of calls: A) calls between two Skype users, one with a public IP address and the other one behind NAT restriction; B) calls between two Skype users, both behind port-restricted NAT; C) calls between two Skype users both with a public IP address; D) calls from a Skype user with NAT restriction (SkypeOut service); E) calls from a Skype user with a public IP address (SkypeOut service); F) calls towards a Skype user with NAT restriction (SkypeIn service); G) calls towards a Skype user with a public IP address (SkypeIn service). By defining η as the fraction of Skype users behind port-restricted NATs, we can easily compute the fraction of the different call types as it is shown in Figure 1 and Figure 2, where empty circles indicate a Skype user behind port-restricted NAT, black circles indicate Skype users with a public IP address, black squares indicate a supernode and the empty squares labeled as "G" indicate a Skype gateway. Note that calls between users with public IP addresses do not require a supernode for the relaying of the media, while when at least one of the involved users is behind a NAT restriction, supernode acting as a tandem node is required.

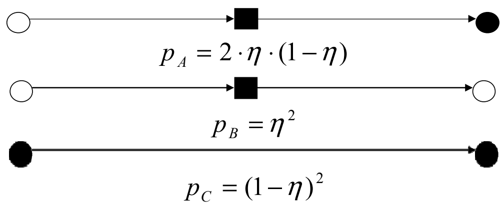

Figure 1. Possible calls between Skype users.

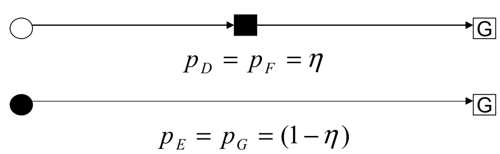

Figure 2. Possible calls from and to Skype users (SkypeOut and SkypeIn services).

We create an end-to-end traffic matrix for calls among Skype users (from now on indicated as the *on-net* service), one for the Skypein service and one for the Skypeout service (for a total of three end-to-end traffic matrices). The end-to-end traffic matrix for the on-net service is generated as follows. The element $A_{on\text{-}net\ i,j}$ is defined as the on-net traffic, in Erlang, from AS *i*th to AS *j*th and it is calculated as follows:

$$A_{\text{on net i,j}} = A_{o,u} \cdot O_i \cdot p_{\text{on net}} \cdot \frac{O_j}{N_o} \qquad (4)$$

where $A_{o,u}$ is the per-user generated traffic (Erlang) and $p_{\text{on\_net}}$ the percentage of traffic generated by Skype users and directed to Skype users. According to the results of Skype traffic volumes published in [7], the ratio of minutes of *on net* traffic to *Skypeout* traffic is 4:1, therefore in our model we set $p_{\text{on net}}$ probability equal to 80%.

Also *Skypeout* (5) and *Skypein* (6) end-to-end traffic matrices are created, and calculate as follows:

$$A_{\text{Skypeout i,j}} = A_{o,u} \cdot O_i \cdot (1 - p_{\text{on net}}) \cdot \frac{G_j}{N_g} \qquad (5)$$

where G is the vector whose elements G_j is the number of gateways of the *j*th AS.

$$A_{\text{Skypein i,j}} = A_{o,u\ \text{ext}} \cdot \frac{G_i}{N_g} \cdot \frac{O_j}{N_o} \qquad (6)$$

where $A_{o,u\ ext}$ is the total amount of traffic received from Skype users which have a SkypeIn number.

After the creation of the three end-to-end matrices, the supernodes traffic relaying effect is accounted for. Depending on the position of caller, callee and their relay supernode in the network, A and B call types can be further divided in other four subclasses, as shown in Figure 3:

I) Both users are inside the same AS and the relay supernode in one other AS.

II) Caller, or callee, and its relay supernode are in the same AS, and the other user is in a different AS.

III) Users and their relay supernode are in three different ASs.

IV) Both users and their relay supernode are in the same AS.

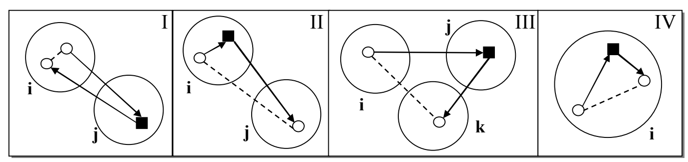

Figure 3. Possible calls between two Skype users and their relay supernode, respect their position in the network (inter-AS level).

It is possible to observe how I e III call types produce an increase of the traffic network transported by network links (solid lines), at inter-AS level, in comparison to the shorter call between two users with public IP address (dashed lines). For reproducing the increasing of network traffic due to supernode relaying two transformations matrices are calculated and added to the $A_{on\ net}$ matrix:

$$\Delta^{(1)}_{\text{i,j}} = A_{\text{on net i,i}} \underbrace{\left(1 - \frac{s_i}{\sum_{k=1}^{N} s_k} \right)}_{p_{\text{Ii}}} (p_A + p_B) \frac{S_j}{N_S - S_i} \qquad (7)$$

Where, p_{Ii} is the fraction of type I of calls with caller and callee located in the AS *i*th, p_A is the probability that at least one users involved in the call is NAT-restricted, p_B is the probability that both users are NAT-restricted, and S is the vector whose element S_j represents the number of supernodes of the *j*th AS. This first transformation, as described in (7), operates on the diagonal elements of the $A_{on\ net}$ matrix, and it redirects a portion of the intra-AS, $p_{\text{Ii}} \cdot (p_{\text{A}} + p_{\text{B}})$, on the *i-j*th relation, proportionally to the probability that the relay supernode is in the AS *j*th. It is worth noting that on the *i-j*th relation both $\Delta 1_{i,j}$ and $\Delta 1_{j,i}$ elements have to be added as described in the following equation (8):

$$A'_{\text{on net i,j}} = A_{\text{on net i,j}} + \Delta^{(1)}_{i,j} + \Delta^{(1)}_{j,i} \big|_{i \neq j} \tag{8}$$

The second transformation operates as follows:

$$\Delta^{(2)}_{i,j} = (p_A + p_B)\left(-A_{\text{on net i,i}} \underbrace{\left(1 - \frac{s_i + s_i}{\sum_{k=1}^{N} s_k} \right)}_{p_{\text{IIIi,j}}} + \psi_1 + \psi_2 \right) \tag{9}$$

where $p_{\text{IIIi,j}}$ is the probability of a type III call with caller located in the AS *i*th, callee in the AS *j*th and supernode in one other AS.

As shown with equation (9), the second transformation removes a fraction of the traffic $((p_A + p_B) \cdot (-A_{\text{on net i,i}} \cdot p_{\text{IIIi, j}}))$ from the *i-j*th relation. Moreover, two contributions are added from the other relations using the relaying services of a supernode:

- From supernodes in AS *j*

$$\psi_1 = \sum_{k=1, k \neq i, j}^{N} A_{on \text{ net i, k}} \cdot p_{III\ i,k} \cdot \frac{S_j}{\sum_{r=1, r \neq k, i}^{N} S_r} \tag{10}$$

- From supernodes in AS *i*

$$\psi_2 = \sum_{k=1, k \neq i, j}^{N} A_{on \text{ net k, j}} \cdot p_{III\ k,j} \cdot \frac{S_i}{\sum_{r=1, r \neq k, j}^{N} S_r} \tag{11}$$

The same procedure is repeated for the $A_{Skypeout}$ and $A_{Skypein}$ matrices and the total end-to-end traffic matrix is computed as $A_{tot\ end\text{-}to\text{-}end} = A_{on\ net} + A_{Skypeout} + A_{Skypein}$.

Users behind NAT restriction depend on a supernode for the traffic relaying service. However, the number of calls that a supernode can simultaneously route is limited. According to the Skype forum information this limit is approximately equal to 12-13 calls. For reproducing this limit of supernodes the $A_{supernodesblock}$ traffic matrix is created. Each element of this matrix is the supernode managed traffic flow on the *i-j*th relation without the application of any call blocking. Concerning the calculation of the call blocking probability due to supernodes, we have adopted two models. The *pessimistic* model assumes that the average traffic per supernode, referred to as *avetraff*, is equal to total amount of traffic divided by the number of supernodes. The supernode's call blocking probability is then calculated as:

$$P_{pessblock} = E_1(\text{avetraff}, \#\max\text{ofcalls}) \tag{12}$$

where *#maxofcalls* is the maximum number of calls that the supernode can simultaneously route. NAT-restricted users contact a randomly chosen supernode for the traffic relay service and, if the selected supernode is already routing *#maxofcalls* calls, the call is blocked. In the *optimistic* model the call blocking probability is calculated as:

$$P_{optblock} = E_1(\text{tottraff}, \#\text{maxofcalls} \cdot N_S) \tag{13}$$

where *tottraff* is the total amount of traffic managed by supernodes and N_s the number of supernodes. Therefore *#maxofcalls* $\cdot N_s$ is the total number of lines available in the network for the traffic relaying service. The optimistic model assumes that a centralized search system is available and, if at least one line is available for relaying on an arbitrary supernode, the system is able to set up the call.

After the computation of the blocking probability, it is applied to each relation of the $A_{supernodesblock}$ matrix, whose elements become the exceeding part of traffic that has to be removed from the elements of the $A_{tot\ end\text{-}to\text{-}end}$ matrix.

The next step consists in the generation of the $A_{erl\ link}$ matrix whose elements represent the total amount of traffic (Erlang) that is transportated by each physical link. For the computation of this matrix, both the $A_{tot\ end\text{-}to\text{-}end}$ matrix and the *R* routing matrix are used. In fact, each *i-j* end-to-end relation has to be mapped on the physical links that it crosses. We assume that physical links are bidirectional and symmetric, therefore, $A_{erl\ link}$ matrix is added to its transposed one, and the resultant matrix becomes symmetric.

We proceed by introducing the quality parameter *P*, defined as the fraction of calls with a bad quality, used to evaluate the minimum capacity (kbit/s) of each link required to satisfy a given performance target. We calculate *P* as:

$$P = \sum_l f_l P_l = \sum_l f_l \left(1 - (1 - p)^l\right) \tag{14}$$

where *p* is defined as the probability that the performance of a single link is low enough to degrade the quality of a call. We dimension the capacity of physical links in such a way that the probability *p* is equal for all links. Given an end-to-end path of *l* physical links, the probability that a call crossing the path is degraded is equal to $1-(1-p)^l$. f_l is the fraction of end-to-end paths composed of *l* links and it is given by:

$$f_l = \frac{\sum A_{tot\ \text{end-to-end}\ i,j} : l_{i,j} = l}{\sum A_{tot\ \text{end-to-end}\ i,j}} \tag{15}$$

Equation 14 gives the probability *P* that a generic call has it quality degraded as a function of *p*. We fix the overall performance target P and, by inverting the equation 13, we obtain *p* that we use to dimension links. We refer to the link capacity consumed by the telephone codec as *v* (bit/s). If a link carries *n* conversations, the link's capacity needs to be greater than nv.

The minimum capacity $C_{i,j}$ of link (*i*, *j*) is obtained by inverting the following equation:

$$p = E_1\left(A_{Erl\_link\_i,j}, \frac{C_{i,j}}{v} \right) \tag{15}$$

V. SIMULATION RESULTS

Firstly, we have analyzed how the Skype system uses link resources at inter-AS level, by varying both the size of the underlay network and the parameters of the overlay networks (number of users, per user traffic and behind NAT port-restricted percentage). Secondly, we have analyzed how the number of supernodes and the way in which they are selected by the Skype application affect the performance and the usage of network resources, with special reference to the traffic relaying service. Both analyses aim at identifying potential bottlenecks of the scalability of the Skype network.

From the results of recent studies [8] and from the analysis of BGP routing tables [5], we can understand that currently there are about 25,000 Autonomous System in the world. Such a wide system is difficult to model mainly because of memory occupancy. Therefore, we have analyzed smaller networks, up to 10,000 Autonomous Systems. We have kept constant the ratio of the number of users to the number of ASs. Currently, there are more than 8 millions online Skype users, at any time of the day, and so we obtained a ratio equal to 320:1. Therefore, in our simulations, 320 thousand users are created for the 1000 ASs model, 640 thousand people for the 2000 ASs and so on. From the number of online users, the supernode population is obtained by keeping a ratio between the number of supernodes and the number of users equal to 1:250 [9]. Other default parameters are the percentage of users behind NAT restriction, which is equal to equal to 75% (according a recent study [9][10] about Internet connection behind NAT restriction) and per user traffic, equal to 40mErlang.

Starting from the average Skype traffic transported by network links, we observe that it remains almost constant (as shown in Figure 4, continuous line), despite the increasing of the network size. The link carrying the largest traffic (as shown in Figure 5, continuous line), always located between the AS with the highest degree and another AS member of the "rich club", it is possible to observe a growth as the size of the underlay network increases. This behavior can be explained by taking into account the topological properties of a Scale Free network. In fact, as a Scale Free network grows, a large number of low-degree ASs is added, connecting preferentially nodes belonging to the "rich-club". In this way, the core of the Internet becomes more and more "hot", as it is crossed by the traffic among a large number of ASs. On the other hand, the average traffic carried by links remains almost constant because the number of links in the core, carrying a very large traffic, is very small relatively to the number of links with low capacity, located at the periphery.

The way in which the average traffic carried by network links and the traffic carried by the largest link change as the size of the underlay network grows, depends strongly on the properties of the underlay level. The same study performed with underlay networks generated by random algorithm (Figure 4 and Figure 5 with dashed line), shows that the average capacity of links does not remain constant as the size of the underlay network grows, and the traffic carried by the largest link is significantly larger for a scale-free network than for a random network.

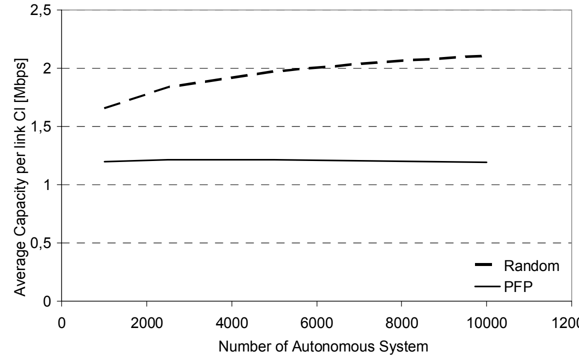

Figure 4. Average capacity of links in the network, varying the underlay level size, which is created with the PFP algorithm or random algorithm.

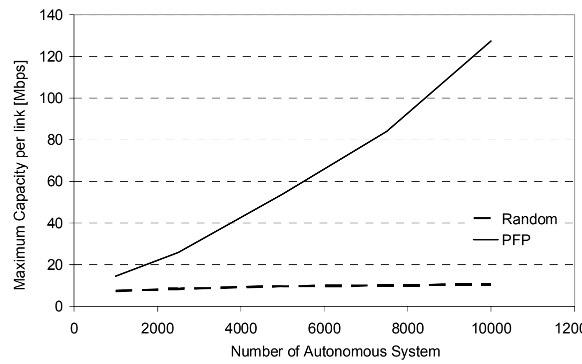

Figure 5. Maximum capacity of a link in the network, varying the underlay level size, which is created with the PFP algorithm or random algorithm.

We conclude that the average Skype traffic carried by network links does not change significantly as the size of the underlay network grows. However, the Skype traffic carried by the largest link increases remarkably.

The next step of our analysis has been to evaluate the sensitivity of the average Skype traffic carried by network links, referred to as C_l and expressed in Mbps, to variations of the parameters of the overlay network, i.e., number of Skype users, per user traffic and percentage of users behind NAT restriction. We consider as a reference metric for our analysis the average traffic offered per network link, referred to as C_0 (expressed in Mbps), and calculated as follow:

$$C_o = \frac{\left(A_{0,u} N_0 \right) r}{N_{link}} \tag{16}$$

where:

- r is the applied bit rate (30 kbit/s) of each call;
- N_{link} is the total number of links in the network;
- $A_{0,u}$ the offered traffic per user expressed in mErl;
- N_0 the total number of ordinary hosts.

It is worth noting that A_{tot}, i.e. the total amount of traffic in the network (expressed in Erlang), is given by:

$$A_{tot} = A_{0,u} \cdot N_0 \tag{17}$$

In this study, the following values $A_{0,u}$ are considered: 20, 40, 60, 80 and 100 mErlang. As a consequence, our approach is to evaluate the scalability of the system intended as the effect of the growth of C_0 on the average traffic carried by network links (C_l). In Figure 6, C_l is plotted as a function of C_0 for different percentages of users behind NAT restriction. It is worth noting that the average traffic carried by network links is always greater than C_0 with a multiplicative coefficient m_{mc} ranging in [3.6, 7].

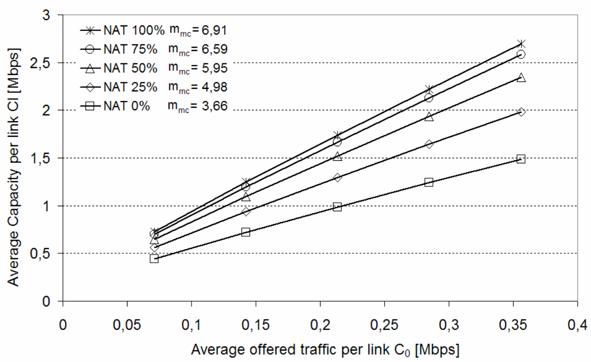

Figure 6. Average traffic carried by network links as a function of offered traffic, for different percentage of user behind NAT.

We have determined that the m_{mc} coefficient can be expressed as:

$$m_{mc} = m_{mr} + m_{mN} \tag{18}$$

where m_{mr} and m_{mN} are two components due to the effect of the following concurrent and independent causes:

- Routing: as the network graph is not fully meshed, more than one link for each traffic relation is loaded;
- NAT restriction: as a user behind NAT restriction needs the relay service of a supernode, an increasing of the NAT percentage causes a growth of the overall traffic in the network.

By setting the percentage of NAT-restricted users to 0, it is possible to evaluate that the effect of the routing amounts to m_{mr} = 3.6. This value represents the average path length between caller and callee (for a uniform distribution of the users) and, as a consequence, it is independent on the NAT percentage variation.

With a percentage of NAT-restricted users equal to 100%, the m_{mc} coefficient approximately doubles the m_{mr} component, meaning that m_{mN} is almost equal to m_{mr}. Such a result is related to the doubling of the average path length caused by the effect of supernode relaying. The growth of carried traffic shows saturation as the NAT percentage increases.

These results can be considered as the product of a worst case analysis in terms of overall traffic in the network, because with the assumption of uniform distribution of Skype users, we do not register ASs without users. On the contrary, in every realistic distribution of users, we have ASs without Skype users (e.g. transit-ASs). While keeping constant the number of users and their offered traffic, an increased number of ASs without users causes a growth of the fraction of intra-AS traffic and a consequent reduction of inter-AS traffic. As a consequence, the overall traffic in the network at inter-AS level is smaller than the inter-AS traffic that we have evaluated with our analysis.

The second type of analysis that we have carried out is relative to scalability of the system as a function of the number of supernodes needed to guarantee the traffic relaying service for users behind NAT restriction. In our model we have assumed that call blocking occurs when no supernodes are available for providing the relay service. Figure 7 shows the maximum ratio ρ of the number of Skype users to the number of supernodes that provides a *target call blocking probability* ≤ 1%. In particular two different cases have been considered: *pessimistic* and *optimistic*. In the *pessimistic* case, when relaying is needed only one supernode is queried and, consequently, if the supernode is not available the call is blocked. In the *optimistic* case, if relaying is needed all supernodes are queried and the call is blocked only if all supernodes are not available.

It is possible to observe that when $A_{0,u}$ grows, a higher number of supernodes is needed, and when $A_{0,u}$>55 mErlang, the current number of supernodes in the real Skype network, with a ratio ρ=250, would not be enough to provide the relaying service with the targeted call blocking probability, even in the optimistic case.

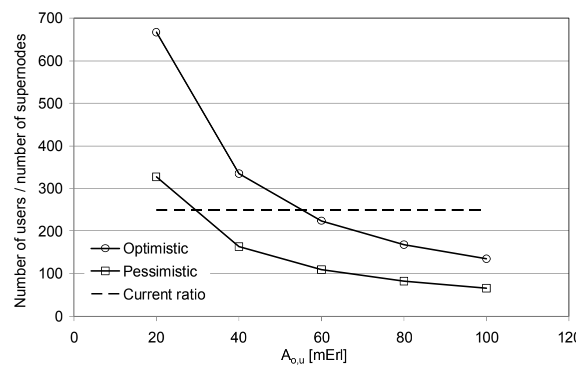

Figure 7. Ratio of the number of users to the number of supernodes versus the per-user traffic. Behind NAT restriction percentage is 75%

It is important to note that this result is strongly dependent on the maximum number of calls that a supernode can handle at the same time, currently set at 12. Therefore we decided to establish how this parameter influences results. Figure 8, plotting ρ versus the maximum number of calls that a supernode can concurrently handle, shows that the previous result (Figure 7) can be consider correct until the number of handled calls is greater than 9. With a smaller relaying capacity of supernodes, even in the optimistic case, the current number of supernodes in the network would not be sufficient to guarantee the targeted call blacking probability. Moreover, it is possible to observe that, with the current conditions of per-user traffic and behind-NAT-restriction users, only if supernodes were able to handle at least 17 calls, at the same time, the *pessimistic* (one shot) method of research could be applied. This is a clear indication that in the current version of Skype the search of supernodes seems to be better than the rough search method that we have assumed in the pessimistic case.

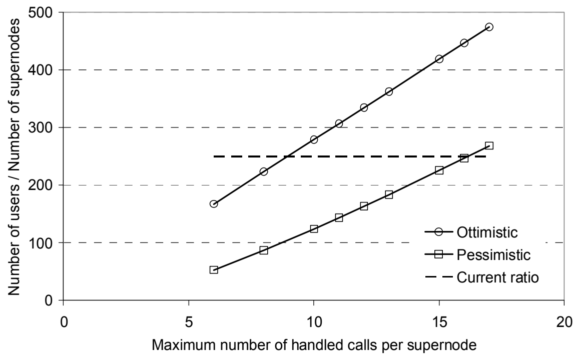

Figure 8. Ratio of the number of users to the number of supernodes versus the maximum number of calls handled by a supernode. Behind NAT restriction percentage is 75%, per user traffic 40 mErl.

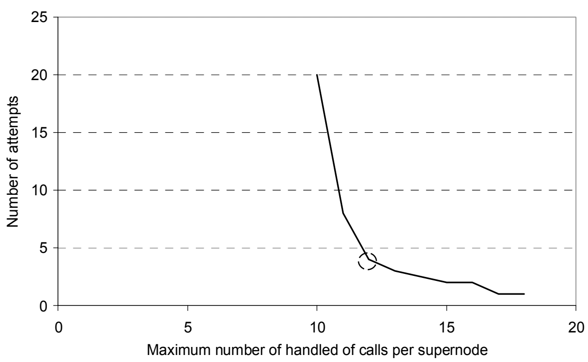

Figure 9. Analysis of the number of supernodes as function of the maximum number of calls that a supernode can handle.

In Figure 9 we plot the number of supernodes that the system needs to interrogate to relay a call. By interrogating 4 supernodes, a call blocking probability smaller than 1% is guaranteed. The system parameters chosen for this analysis are the current estimations of number of nodes, supernodes and users of the Skype system, previously outlined. The computational power and the capacity of the Internet connection of supernodes is a critical feature. In fact, Figure 9 shows that the number of supernodes to be interrogated to assure a given blocking probability decreases significantly as the maximum number of calls that a supernode is able to manage grows.

VI. Conclusions

In this paper we have performed a scalability study of the Skype system based on two different critical features of the network both at the underlay and at the overlay level. In the underlay network, we have considered as a potential scalability bottleneck the traffic carried by network links and, at the overlay level, we have selected the ratio of the number of Skype users to the number of supernodes.

We have studied how the average Skype traffic carried by network links increases as the size of the underlay network grows, and we have found that it is almost constant, as also the number of Skype users increases as the network size grows. However, the traffic carried by the largest link increases dramatically as the network size grows and we have shown that this is a phenomenon due to the scale-free nature of the Internet inter-AS graph and it would not occur if the Internet graph were a pure random graph. We conclude that the scalability issue for Skype, as far as the capacity of links is concerned, is not critical. Only the largest links experience an increase of traffic as the underlay network grows, but this does not seem to constitute a problem, as the capacity of the core Internet links is very large.

The ratio of the number of Skype users to the number of supernodes is more critical. We have found that the relaying service offered by supernodes to NAT-restricted users can be offered with a proper quality (i.e., a small call blocking probability) only if a precise set of requirements is met. Firstly, the number of calls that a supernode should be able to handle concurrently must be greater than 9. Second, the number of supernodes that the Skype system searches to perform the relaying service is critical and it modulates the maximum admissible ratio of users to supernodes. We found that the maximum sustainable ratio ρ between the number of Skype users and the number of supernodes decreases as the overall traffic in the network increases. This is an important scalability issue, as supernodes are carefully selected computers with high performance, high availability and a large access bandwidth. The number of supernodes is relatively small, therefore supernodes drive the maximum size that the Skype network can reach.

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