MODEL-BASED STOCHASTIC SIMULATION OF SUPER PEER PROMOTION IN P2P VOIP USING GRAPH TRANSFORMATION

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Keywords: Stochastic modelling and simulation, P2P VoIP, P2P VoIP network architecture, Graph transformation.

Abstract: The concept of super peer has been introduced to improve the performance of popular P2P VoIP applications. A super peer is the strongest peer in the network that has the capacity to act as a server for a set of VoIP clients. There is no doubt that by taking benefit of heterogeneity, super peer can do improve the efficiency, without compromising the decentralised nature of P2P networks. The core issue in the formation of super peer based overlay network is the selection of super peer among the participant peers. To solve this problem a number of solutions have been proposed in the literature. Generally, super peer are selected among the best nodes in the network, for example those with the most abondant resources, such as bandwidth, CPU cycles or memory. The next issue is when the peer shall be selected for this extended role either as when peer joins the network, or at any time during the session or promotion to super peer should be subject to requirements. In order to validate these approaches of super peer selection, simulation would be an ideal choice, but most existing simulation approaches cannot cope with unbounded dynamic changes of network topology.

We propose an approach to modelling and simulation of P2P systems based on graph transformations, a visual rule based formalism that has recently been supported by facilities for stochastic modelling and simulation. We are considering a P2P VoIP applications such as Skype, we model three alternative solutions to the problem of peer promotion to super peers and evaluate these through simulation.

1 INTRODUCTION

Today's P2P network (Milojicic, 2002) contains several new and attractive features, which makes them quite different from classical P2P networks. Although, network may still contain millions of peers. As a result of this massive number of peers, they are subject to high degree of churn (Stutzbach and Rejaie, 2006), with a number of peers joining and departing from the system respectively. This massive size of network and geographically diverse location of peers pose several challenges for developers and researchers. As immediate affect its impossible to use either a central authority or fixed communication topologies. Instead a dynamically changing overlay topology is used, where control is kept decentralised. The respective overlay topology is maintained by cooperation of virtual links between peers, these virtual connection are created and removed based on the requirements of particular application.

Peers in P2P networks are always in full control of their local resources and can therefore, any time impose new policies regarding use of their shared resources in the network (Ji, 2004). A peer may even suddenly stop its cooperation by behaving selfish (Gupta and Somani, 2004). Due to this lack of global control and unreliability of the P2P architecture, these systems are prone to dependability problems. Classical P2P applications were lagging the enforcement of particular overlay topology. The direct result of this approach was adaptation of inefficient scheme such as flooding. However, now researchers and developers have realised the importance of building and maintaining appropriate topologies for development of efficient robust P2P Systems (Zhao, 2003; Rowstron and Druschel, 2001; Montresor, 2004; Dabek, 2001; Yang and Garcia-Molina, 2003).

Along with many file sharing applications (Montresor, 2004), P2P VoIP is also considering the implementation of structured topologies by distinguishing peer and super peer (Guha et al., 2006). One of the successful VoIP application developed based on structure topology is Skype (Skype, 2011; Guha et al., 2006). The topology is composed of two-level hierarchy: Nodes with powerful CPU, more free memory

Khan A. and Heckel R..
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DOI: 10.5220/0003526000320042
In *Proceedings of the International Conference on Data Communication Networking and Optical Communication System* (DCNET-2011), pages 32-42
ISBN: 978-989-8425-69-0

and greater bandwidth take on server-like responsibilities and provide services to a set of client peers. This approach allows decentralized overlay network to run more efficiently by exploiting heterogeneity and distributing load to machines that can handle the burden. It has also overcome the flaws of the client server model, because of multiple separate points of failures, thus increasing the health of the P2P overlay network (Guha et al., 2006; Yang and Garcia-Molina, 2003).

Building and maintaining a super peer based overlay topology is not simple. Rapid architectural changes in both ordinary and super peers requires robust and efficient protocols, capable of self-reconfiguring, in spite of both controlled and selfish events like joining, leaving or crashing nodes. In case the P2P is used for VoIP traffic, the network needs to reconfigure fast enough so that Quality of Service (QoS) is not affected (Khan et al., 2009).

Several questions arise for the design of network protocols: Which peer should be promoted to super peer? Can selection of super peer based on the higher bandwidth is enough? Is promotion at the time of registering is better from promotion at any time during session? How about promotion whenever there is a need for super peer? The performance of such a protocol can be measured by answering the question: How many peers are connected, how many are present but yet to be connected, how many super peers are there, how many clients are happy in the network, how many VoIP calls have been supported?

Various solutions have been proposed to these problems, e.g. (Zhao, 2003) suggested the deployment of super peers directly managed by content service providers; (Lo et al., 2005) proposed three different approaches for selection of super peers; (Montresor, 2004) presented a super peer overlay topology algorithm and validated his approach using the Psim simulator; (Oneil et al., 2000; Biondi and Desclaux, 2006; Guha et al., 2006) proposed super peer promotion based on bandwidth, *CPU* power and time spent in the network; (Gupta and Somani, 2004) proposes that an incentive should be given to intermediate nodes and resource owners; (Heckel, 2005) proposes to maintain redundant links between peers, (Ren et al., 2006) propose an autonomous system-aware peer-relay protocol called ASAP; (Lysne et al., 2005) proposes solutions based on changes in routing strategies; (Yang and Garcia-Molina, 2003) discussed general design issues however, their focus is on centralized design of such networks.

However, peer dynamics and complexity of P2P network makes it difficult and expensive to validate these solutions through testing of real networks. Geographical distribution of peers, network dynamics and lack of central control make testing difficult and costly. The simulation of network reconfiguration as well as dynamic role assignment is not easy, as existing simulators do provide very limited support for networks with dynamic topology (Heckel, 2005; NS2, 2008; Khan et al., 2010).

We propose to model dynamic role assignment and associated architectural reconfigurations in P2P VoIP networks by means of graph transformation systems and use a new approach to the stochastic implantation of such systems to evaluate the performance of network protocols. We consider the P2P network architecture as a graph, in which network nodes are represented by graph vertices and graph edges represent network connections. Reconfiguration in such a network can naturally be modelled by graph transformation, in a visual and rule-based formalism (Khan et al., 2009; Heckel, 2005; Khan et al., 2010). Stochastic simulation techniques for validation have been developed in (Khan et al., 2009).

In this paper we are going to present a case study based on the popular VoIP application Skype and discuss four different variations for promotion of peer to the role of the super peer.

2 CASE STUDY: SKYPE NETWORK

Skype is P2P VoIP application developed in 2003 by the owners of KaZaA. Skype claims that it has currently more than 200 million users. Real statistics shows that averagely 25 million users always remain on-line. It currently contributes more than 8% traffic to VoIP applications over internet.

Skype architecture of its operation is getting popularity in P2P research community and Telecom operators, and detailed valuable studies can be seen in the literature (Baset and Schulzrine, 2006; Biondi and Desclaux, 2006; Idrees and Khan, 2008). But due to proprietary protocol, sophisticated nature and anti reverse engineering techniques (Biondi and Desclaux, 2006), many interesting question still remains unanswered. Since, its first release many versions have been released but the core architecture of the network still remain the same. Skype network consists of Skype Clients and Super Peers (Baset and Schulzrine, 2006). The networks nodes are heterogeneous in terms of network locality, CPU power, memory, and most importantly network bandwidth and type of IP. Skype client with sufficient resources in terms of bandwidth can promote to the new role of Super peer, while at the same time, it continues primary role of the client for the physical user (Baset and Schulzrine, 2006; Oneil et al., 2000). Super peers form an overlay network among themselves whereas each client has to first register with central registration server and then subsequently select one of the super peer as their local host or local serving point of contact (Baset and Schulzrine, 2006). The client will use super node for searching on-line buddies as well as for actual traffic routing if client is behind *NAT* or firewall. The registration server is the only server in the Skype network, responsible for storing user information including passwords, authenticating users at login, and provide client with super peer address for local host connection. All relevant information regarding on-line users, there current status is stored in distributed fashion, which does improve the scalability and stability, although these information can sometime be out of date.

The Skype call cost in terms of the bandwidth is not clear due to the proprietary nature but analysis in (Baset and Schulzrine, 2006) suggests that call cost is between 56 Kbps to 16 Kbps whereas (Adami et al., 2009) states that cost in terms of bandwidth is 25 Kbps when call is active. The client and super peer also sends each other keep alive messages which consume bandwidth as well. The actual cost is not clear but (Guha et al., 2006) states that 5 Kbps to 8 Kbps is the cost of keeping connection with super node or super node to super node overlay. These keep a live messages enable both the client and super peer to self-organise, if the client or super peer has left the network. The super peer based on the local awareness of bandwidth may accept or reject overlay topology formation or local host request. Both client and super peer can leave the network either by crashing or by using cooperative exit procedure provided by the application GUI.

3 A GRAPH BASED MODEL FOR SKYPE PROTOCOL

We use graph transformations to model the structural evolution of the Skype network. As one of the most basic models for entities and relations, graphs are a representations of structural models. Formally, a graph consists of a set of vertices V and a set of edges E such that each edge $e \in E$ has source and target vertex $s(e)$ and $t(e)$ in V, respectively. More advanced notions allow for nodes and edges to be attributed with textual, boolean or numeric data (Lara, 2007). Graphs occur at two levels: type level and instance level. A type-level graph is comparable to a class or ER diagram containing the types of nodes and edges, declarations of attributes, etc. Instance graphs represent the states of the system, typed over the type graph. With graphs as states, transformation rules provide state changing operations (Heckel, 2005; Khan et al., 2009).

The type graph TG in Fig 1 represents a model of Skype architecture as described in before. It defines the types of registration server (RS), super peer (SN), Skype client (SC) and their common super type. The node LK is used to model the link between SC and SN while OV represents the connection between existing SNs, to model the overlay topology. The Call and RouteCall is used to model two different call types. The edges type *offline* is used to show that users is registered but not currently in the network, *online* is used to show when user is successfully authenticated, *overlay-index* is used to model the connection between the SN and RS, *route* show when call is routed through the SN, *caller* is used to show the caller of the call and *callee* is used to model the recipient of the call.

In the model whenever a new SC joins the network it first get registered with *RS* and subsequently user get connected with *RS* through *offline* edge. A registered user when joins the network, a client status *SC* is assigned and *offline* edge is replaced by *online*. After this stage the current *SC* has to select one of the *SN* as local host. The *SN* will be responsible for the client request such as queering network and if *SC* is behind NAT or firewall, the actual call packets will be relayed through SN.

In the model SC is promoted to SN based on three alternate choices. In Fig 2 we have shown these promotion constraints and other mandatory features of the protocol as feature tree. In the feature tree we have shown that SC can promote to the role of the SN based on three choices. First, as soon as it gets on-line and bandwidth is more than 1.5 Mbps, we call this approach as static protocol. Second, SC can promote any time during session in the network, still bandwidth will be considered, we call this approach as dynamic protocol. Third, SC can promote to SN only whenever there is need for a new SN, we call this as need based protocol.

In order to model VoIP calls traffic, we assume a codec costing 60 Kbps of bandwidth during active call. The model supports direct calls in P2P architecture as well as relayed calls. We also randomly update the bandwidth of the nodes in order to model the behaviour of the shared bandwidth and background traffic. In the model if SN leaves the network either by crashing or cooperative exit, the model is capable to reconfigure the client back to connect to a new SN.

In Figure 2 feature tree shows the mandatory fea-

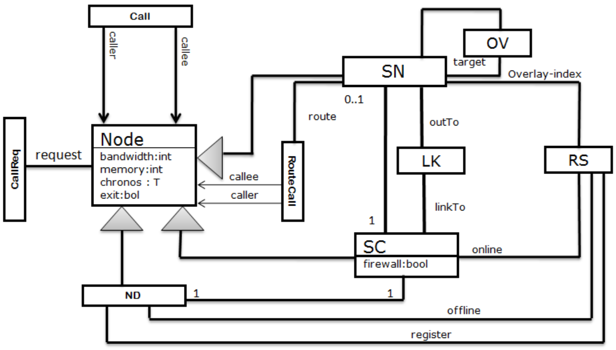


Figure 1: Type graph.

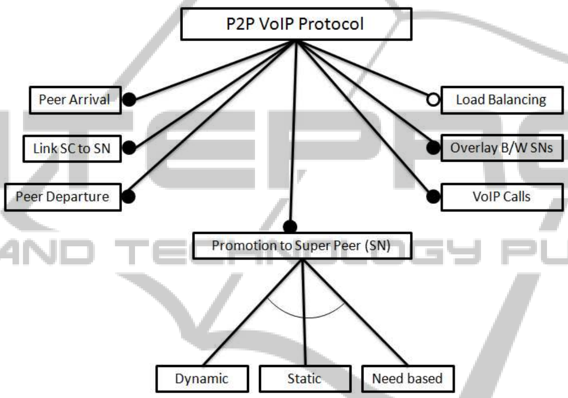


Figure 2: Feature tree.

tures such as go on-line, link SC to SN, load balancing by simply disconnecting, down grade to client (*if SN is having bandwidth less than the minimum 256 kbps*), VoIP call respectively. In the model an SC can exit by crashing as well as exit through a cooperative procedure. All these features are kept constant whereas the promotion strategies are changed in order to check the performance of different SN promotion strategies.

The objective of modelling these three variations of promotion of SC to SN is to evaluate different approaches of promotion against each other. This will also provide information that which approach has scaled the network to what number of SNs and SCs. In addition to that the model will provide information regarding the performance of each approach based on the number of connected, unconnected and happy nodes.

4 SC PROMOTION TO SN AS GRAPH TRANSFORMATION

A graph transformation rule $p : L \longrightarrow R$ consists of a pair of TG-typed instance graphs L, R such that the intersection $L \cap R$ is well defined. The left-hand side L represents the pre-conditions of the rule whereas the right-hand side R describes the post-conditions. Their intersection represents the elements that are required, but not destroyed, by the transformation (Heckel, 2005). Graph transformation rules also use negative application conditions *(NACs)*. A NAC assures that the rule will only be applied if the pattern specified *NAC* does not match the graph (Heckel, 2005; Khan et al., 2009). Graph transformation rule also use *attributes*. The use of attribute enable the rule impose constraints on the attribute value, which will make sure that attribute constraint is evaluated in the pre-condition and attribute value is updated as postcondition of the rule.

We are now going to introduce a set of transformation rules based on a simple client promotion to the extended role of super peer(SN) scenario. Here, due to limitation of space we are not introducing the rules for connecting *SC* to *SN*, crashing, controlled exits, whereas the connection rules are already published in our previous work (Khan et al., 2010). However, in the simulation all these rules are provided in order to give results on the complete model.

Rules in Figure 3: Promote a Newly Joining SC to Extended Role of SN. In this rule a recently on-line

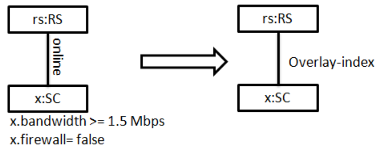


Figure 3: SC promotion to SN at start of joining.

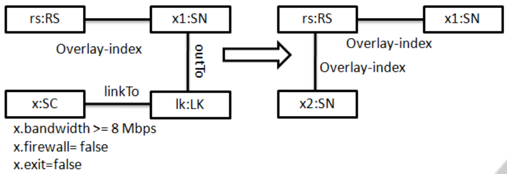


Figure 4: SC promotion to SN at any time.

Skype client *x* will be promoted to its extended role as super peer *SN*, only if the client *x* current bandwidth is more than 1.5 Mbps and is connected to the internet without firewall. After promotion to the new role, SN client *x* will still retain the primary role of the client for the owner user and at the same time will serve other SCs. Once successfully promoted, client *x* informs the server *RS* about the recent changes to its role in the network. The static protocol use this rule to promote SC to SN.

Rules in Figure 4: Promote an SC to Extended Role of SN at Linked State. In this rule a client which is already using the network, can be promoted to the extended role as SN. In this rule a client *x*, is having currently sufficient bandwidth and is connected to the internet without firewall. Further *x* is not leaving the network as cooperative exit procedure. This client *x* can be promoted to SN. The attribute *exit* informs regarding the intended departure of the client *x*. Once selected for promotion, client terminates its dependency connection with the SN and inform the server *RS* regarding the new assignment through edge $overlay-index$. The dynamic protocol use this rule along with rule in Fig 3 to promote SC to SN. This protocol has the capability to promote at the start as well as during active session of the client.

Rules in Figure 5: Promote an SC to Extended Role of SN at Linked State if there is Need for SN. In this rule an *SC* which is currently linked with one of the SN. This linked client can be promoted to the extended role of *SN* if the current host SN bandwidth is less than *1 Mbps*. This SN can select client *x* from dependent clients and promote the same to the extended role of SN. The client *x* will be selected based on the bandwidth attribute value which must be greater than 1.5 Mbps. The need based protocol only use this rule to promote SC to SN.

Rules in Figure 6(a): Assign Random Value to SN Bandwidth Attribute. In this rule *SN* bandwidth

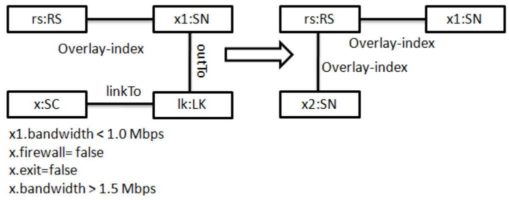
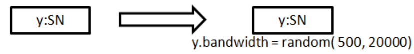
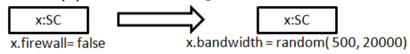


Figure 5: Need based promotion at *linked* state.



(a) Bandwidth usage and release of *SN*.



(b) Bandwidth usage and release of *SC*.

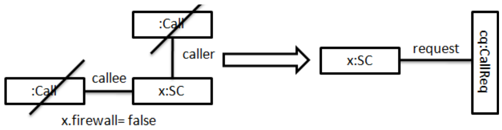
Figure 6: Bandwidth use and release in *SC* and *SN*.

attribute will be assigned random value between 500 Kbps and 2 Mbps. The core purpose of this rule is to change the value of the bandwidth to model other traffic apart from VoIP in the network. Since, P2P are highly dynamic most action in the network including roles of peer depend upon the available bandwidth at certain time in simulation. So due to this a client may start network participation with SN but may end up with a role of SC.

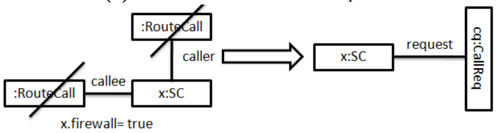
Rules in Figure 6(b): Assign Random Value to SC Bandwidth Attribute. In this rule *SC* bandwidth *attribute* will be assigned random value between 500 Kbps and 2 Mbps if the *SC* is connected with internet through public IP. In the same way this model the traffic that client computer is generating apart from VoIP application. This is one of the reason that dynamic protocol use both the rules as in Fig 4 and Fig 5 to promote at both the stages (Start and linked states).

Rules in Figure 7(a): VoIP Call Request Initiated by SC with Public IP. In this rule an *SC* with public IP connection with the internet can request for establishment of VoIP call with other on-line network participant. Before the call request is generated, SCs existing calls status is checked by using the *NAC* conditions. The *NACs* assure that requesting client *SC* must not be involved in already directly established P2P calls as *caller* or *callee*. We have used the node *Call* to model the direct P2P call.

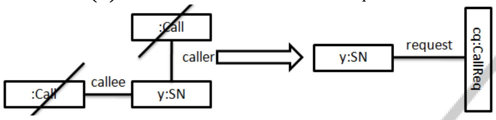
Rules in Figure 7(b): VoIP Call Request Initiated by SC Behind Firewall. In this rule an *SC* behind port restricted firewall can request for establishment of VoIP call with other on-line network participant, The on-line participant can either be *SN* or *SC*. The callee of the call can be either connected through public IP or behind firewall. In all cases the VoIP traffic from this SC will be relayed by the respective host SN of the requesting SC. The *NACs* will make sure that



(a) *SC* with Public VoIP call request.

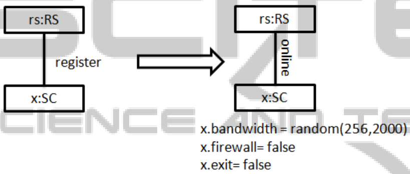


(b) SC behind firewall VoIP call request.

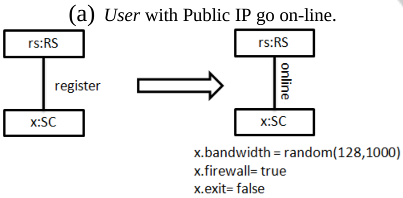


(c) SN VoIP call request.

Figure 7: VoIP call request initiated by *SC* and *SN*.



(a) *User* with Public IP go on-line.



(b) *User* behind firewall go on-line.

Figure 8: User go on-line.

requesting SC is not involved in VoIP calls. We have used the node *Routeall* to model relayed call through SN.

Rules in Figure 7(c): VoIP Call Request Initiated by SN. In this rule an *SN* initiate request for VoIP call with other network participant. As stated earlier *SN* are clients with extended role as Super peer. An SN can not only request calls but also relay traffic for other SCs such as Fig 7(b). The call procedure for SN is the same as that of in Fig 7(a).

Rules in Figure 8(a): User Go On-line with Public IP. In this rule a registered user joins the network. This rules considers that a user is connected to the internet through public IP. Further, based on static IP the user is assigned a random bandwidth between 256 Kbps to 2 Mbps. This new user can be considered as candidate for the extended role of SN if it fulfils the bandwidth requirement.

Rules in Figure 8(b): User Go On-line Behind a Firewall. In this rule a registered user joins the network. This rules considers that a user is behind a restricted firewall. Further, the *firewall* attribute is set to true and this user is assigned a random bandwidth between 128 Kbps to 1 Mbps. This user can not participate in the election process of SN as it has firewall enabled, which restrict this user to communicate or coordinate directly in the network. This user will always be dependent on one of the SN.

5 STOCHASTIC SIMULATION OF GRAPH TRANSFORMATION SYSTEM

Stochastic graph transformation (Heckel, 2005) can be used to model architectural reconfiguration (Khan et al., 2009) and non-functional properties such as performance and reliability. In simple words stochastic graph transformation system (SGTS) is a graph transformation system(GTS) where each transformation rule is associated with a positive real number representing the rate of the exponentially distributed delay of its application to the model. It is important to note that graph transformation can not only model these dynamic networks but it does support a number of validation and verification techniques such as model checking based on CSL and stochastic simulation based on translation to Markov chains (Heckel et al., 2006). Model checking seem the right choice only when abstract properties has to be verified but this approach become hard to use when model is much more complex and dynamic. On the other hand, Monte-Carlo stochastic simulation is typically based on the execution of particular processes, which are selected probabilistically by means of a random number generator (RNG).

Let us consider that a $\mathcal{S}_\mathcal{G} = \langle \mathcal{G}, \mathcal{F} \rangle$ is a *generalised stochastic graph transformation system* whenever $\mathcal{G}$ is a GTS and $F : E_\mathcal{G} \to (\mathbb{R} \to [0,1])$ is a function which associates with every event in $\mathcal{G}$ a general cumulative probability distribution function. We assume that $F(e)(0) = 0$ (*null delay* condition) (Khan et al., 2009). Moreover, the probability distribution is dependent on the event (*rulename and match*) rather than just the rule. The concept of the *SGTS* is explained (Heckel et al., 2006) in detail. Our interest in stochastic graph transformation systems is closely associated with their simulation, where the stochastic aspect is useful in order to resolve the non-deterministic character of ordinary GTSs.

We simulate our model using GraSS (for Graph-based Stochastic Simulation), a tool introduced in (Torrini et al., 2010) and used for our previous work

(Khan et al., 2010). The tool has been developed in Java-Eclipse, as plugin of a graph transformation engine called VIATRA. VIATRA (Bergmann et al., 2008) relies on a RETE-style implementation of incremental pattern-matching, in which precomputed matching information is stored and updated as transformation proceeds. The architecture of the tool is shown in Fig 9. Essentially, the stochastic engine receives the set of enabled rule matches (i.e.the active events) from the transformation engine, turns them into timed events, by assigning to each of them an expected time value, randomly determined on the basis of the probability distribution which is associated with the event type, and sends the events that has been scheduled first back to the transformation engine for execution.

In GraSS a GTS is represented as a VIATRA model, consisting of the model space with the current graph and the transformation rules. Moreover, GraSS takes as input an XML file with the definitions of the distributions associated with transformation rules and respective rates. In order to collect statistics the rules with empty post-conditions that are used as *probes* as can be seen in the architecture Fig 9. Each probe rule returns the number of its matches in the current graph for each state of the transformation system. We used a number of probe rules for collection of simulation results in this paper for the purpose of reader understanding we are presenting one probe rule in both graphical representation in Fig 10 as well as VIATRA code translation shown below:

```
gtrule SC_linked_SN()=
{
precondition pattern lhs(SC,SN,LK)=
{
SN(SN);
SC(SC);
LK(LK);
find LK_LinkTo(LK,SC);
find LK_OutTo(LK,SN);
}
action
{
println("SC linked with SN");
}
}
```

This probe is used to find average number of SCs that are linked with SNs. Additional parameters needed for a simulation run are provided to GraSS as part of the VIATRA model (see (Torrini et al., 2010)). The textual output of a simulation experiment consists of SSJ *TallyStore* class reports (L'Ecuyer et al., 2002).

In this experiments we are going to compare three different protocols for SC promotion to SN. We have used time based simulation, in order to produce comparative results. We have taken simulation unite as 24 hour for these experiments. We use exponential distribution as will as normal distribution. Although due to limitation of space it is not possible to present complete set of rules along with respective probability distribution and necessary justification. We have used normal distribution for all those rules for which implementation boundaries are known. Based on this we have used normal distribution for rule that terminates VoIP calls by keeping in mind that experiments in (Guha et al., 2006) confirms that minimum call duration observed was 2 minutes(m) and 50 seconds(s) and maximum observed duration was 3 hour(h) and 26 m whereas the average as per there experiments was 12m and 53s. Further, we have used exponential distribution for rules presented in this paper at Fig 8. As we are not sure that when a user will go on-line so the rules that makes user to go on-line have been assigned exponential distribution. Similarly, the promotion rules use same exponential distribution.

We performed two different experiments in which we run several simulations of each of the three protocols based on 24 hour simulation time. In the first experiment we varied the rate of the rule that make client on-line by$\{$ *3,4,5,6,7*$\}$. In this experiment a single rule is used which makes user to go on-line. This rule also randomly set the firewall to true and false. The aim is to simulate such behaviour that model does not have any control on the population behind firewall and with static IPs. Due to limitation of space this rule is not presented in the paper. In the second experiment we have fixed the rate of the rule in Fig 8(a) at *5* and we varied the rate of rule in Fig 8(a) by$\{$ *3,4,5,6,7*$\}$. The aim behind this is to reduce the number of candidate for SNs and increase the number of SCs behind firewall. The rates for these rules either in experiment 1 or 2 are selected based on the assumption that if Skype user gets on-line every 8 hour over 24h period then rate based on this principal would be 3, and so on. The aim behind these varying rates is to produce same number of average as that of real Skype network (SkypeStatistics, 2011) in the 24h period in order to confirm the confidence of the model.

In order to comment on the performance of these three protocols, we have used several probes rules such as to find numbers of SCs, SNs, linked SCs, and number of clients that are happy with existing SN. In our experiments an SC is happy with current SN if its current host is having more than 1 Mbps available bandwidth.

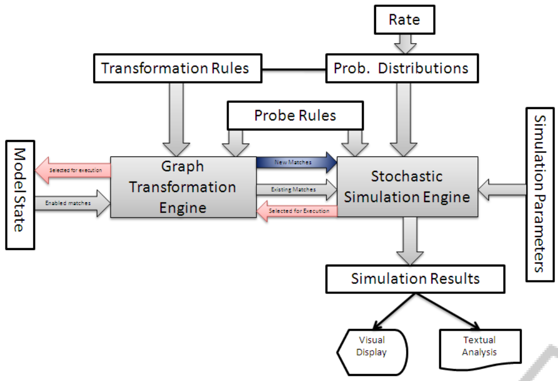


Figure 9: Graph based stochastic simulation (GraSS).

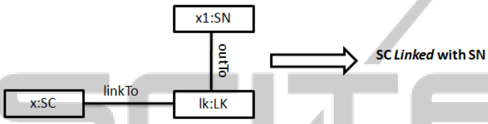


Figure 10: Probe rule *SC* connected with *SN* graphical view.

6 SIMULATION RESULTS

In the first experiment we compared three approaches of client promotion to extended role of SN. Each approach/protocol is represented by a model and set of transformation rules. All three models were tested through 5 variations of rate x for the clients joining the network. The rates assigned were ranging { 3,4,5,6,7}. For each variation we performed 9 runs with a time unit of 4.8 hour. The results are presented as graphs in Fig 11. In this experiment the number of SCs with static IPs and behind firewall randomly joins the network and their ratio is not known.

The graph in Fig 11(a) presents the percentage of the happy peers in the network. As the graph shows that dynamic VoIP protocol for super peer selection approach is performing better as compared to other two at the joining rates 3 and 4. The graph in Fig 11(a) further shows that at joining rate 5, the number of happy peers are almost the same but at rate 6 and 7 we observe that, as the number of clients in the network increases, the need based approach tends to produce good results as compared to other two approaches.

The graph in the Fig 11(b) presents the average number of *linked* clients. From graph we see that dynamic protocol has resulted in well connected network. Although, the other two protocol also managed to link a good number of SCs to SNs. We further see that in terms of SC connection to SN the other two approaches are almost performing at the same level.

The graph 11(d) shows the average number of SNs promoted in this course of simulation. The graph results shows that there is not a big difference between the SNs promoted by all the three protocols. Although, need based approach promoted less number of SNs but in terms of the happy peers the results are not much different from other two protocols approach.

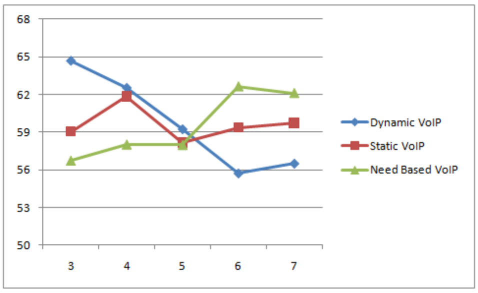
The third graph in the Fig 11(c) further clarify that that all the protocol managed to result in network nodes between 250 and 500. Although need based protocol has slightly less number of clients in the network but the difference in comparison with other two protocol is not much. During the simulation the total number of clients were fixed at 1000.

Th results of first experiment shows that at joining rate 5, all the three approaches managed to produce results which are close enough. In order to further confirm these results we split the single rule of joining into two as presented in Fig 8. Now we have fixed that rate of the rule Fig 8(a), which makes clients online and clients are enjoying static internet connection without enabled firewall. The aim is to fix the number of candidate for SN and we varied the rates for the rule Fig 8(b) by { 3,4,5,6,7}. The aim is to increase the number of users behind firewall. The results of this experiment are presented as graphs in the Fig 12.

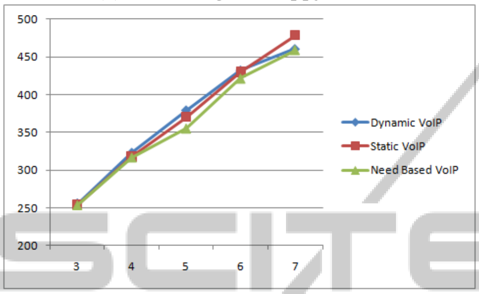
The graph in the Fig12(a) confirms that need based protocol is performing much better as compared to other two approaches when the number SCs behind firewall increases in the model. Although we do observe that in the start the dynamic approach is performing better as compared to other but at joining rate 5 where both rules are ruining at the same rate, the results are changed and need based continues to produce better results as the number of SCs behind firewall increase. The second graph in Fig12(b) the average number of SCs behind firewall in each rates, which are almost close to each other in all the three approaches.

7 VERIFICATION AND VALIDATION OF SIMULATION MODEL

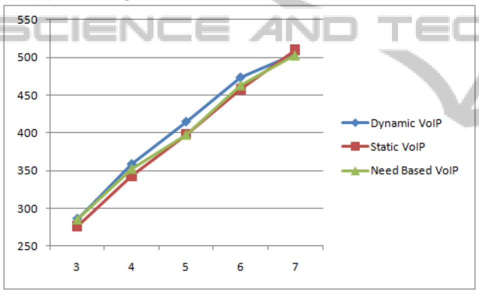
Verification and validation of the simulation models are the core entities in the modelling and simulation study. There are various methods used in the literature for this purpose (Sargent, 1996; Whitner and Balci, 1989). In this study we have used static and dynamic verification techniques in order to make sure that model has rightly been converted to VIATRA model and the associated transformation rules are also rightly converted. The static technique has ensured



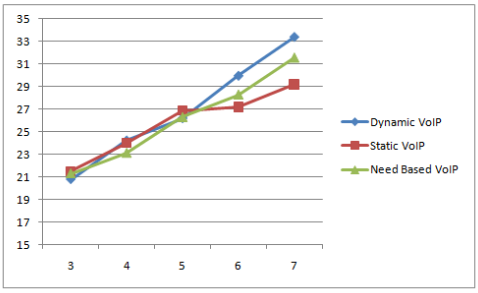
(a) Percentage of happy SCs.



(b) Average numbers of SCs *linked* to SNs.



(c) Total average number of clients.

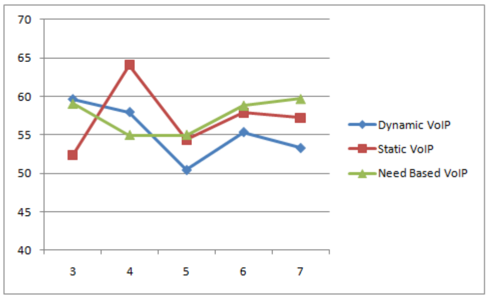


(d) Total average number of SNs.

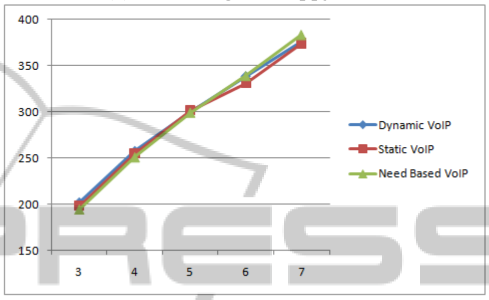
Figure 11: Simulation results first experiment.

that model and rules are error free and right syntax has been followed. The dynamic verification technique has ensured that each rule does the task that it was suppose to do.

Similarly for the purpose of model validation we have used face validation, event validation as well as compared the model with real system (Sargent, 1996; Whitner and Balci, 1989). Fore comparison with real system we have used the study in (Guha et al., 2006) and Skype statistics from (SkypeStatis-



(a) Percentage of happy SCs.



(b) Average numbers of SCs *behind firewal*.

Figure 12: Simulation results second experiment.

tics, 2011; SkypeJournal, 2010). The author analysis in (Guha et al., 2006) is based on real system study, their results shows that over a period 24 hour, the average number of clients that join the network varies by over 40% of the active registered accounts, whereas the average number of super node in this period is below 24% of the active population. Similarity, the author in (SkypeJournal, 2010) states that in December, 2010 there were 1.4 million super peer/nodes and 25 million user were on-line, when Skype network crashed due to non availability of super peers. This figure confirms that almost 6% of the on-line user were super peers and still Skype network crashed. This further confirm that Skype super peer population must be over 6% of the on-line users, in order to make networks stable. Based on these results we have examined our model simulation results obtained over a period of 24 hour simulation. The simulation results confirms that super node population in our model is less than 24% which confirms the results of the (Guha et al., 2006) and are over 6% which further confirms the statement of the author in (SkypeJournal, 2010). Our model further confirm the overall average number of population with real statistic of (SkypeStatistics, 2011; Guha et al., 2006). Once we have obtained these results we have fixed the rates of rules and varied the rates of the rules where we have to make sensitivity analysis.

8 CONCLUSIONS

In this paper we have outlined our simulation approach based on stochastic graph transformation. We used it to model and simulate some aspects of P2P VoIP network protocols, and we have performed our experiments with the GraSS/VIATRA tool (Torrini et al., 2010). We have compared three client promotion to exteded role protocols. The dynamic protocol does not seem to be performing better as compared to the need based and static protocols in terms of happy peers, especially when the number of clients in the network increases over 400. The results further confirmed that increased in super peer does not mean that network may also result in good number of happy clients. We further observed that need based protocol is most suited when there are greater number of clients, as super peer promotion costs in terms of bandwidth use.

As future work, we are planning to extend the model and include spatial information regarding notions of jitter, packet loss and echo, along the lines of (Khan et al., 2009), and to compare a number of different design solutions to problems such as routing, load balancing, selfish exit, and cooperative exit from the network, in order to investigate their tradeoffs and benefits.

ACKNOWLEDGEMENTS

This research is supported by University of Malakand, Pakistan through faculty development programme of HEC, Govt. of Pakistan.

REFERENCES

Adami, D., Callegari, C., Giordano, S., Pagano, M., and Pepe, T. (2009). A real-time algorithm for skype traffic detection and classification. In *Lecture Notes in Computer Science*.

Baset, S. A. and Schulzrine, H. G. (2006). An analysis of the skype peer-to-peer internet telephony protocol. In *25th INFOCOM'06*.

Bergmann, G., Őkrős, A., Ráth, I., and Varró, G. (2008). Incremental pattern matching in the VIATRA model transformation system. In *GraMoT'08*.

Biondi, P. and Desclaux, F. (2006). Silver needle in the skype. Technical report, EADS Corporate Research Centre, Suresnes, France.

Dabek, F. (2001). Building peer-to-peer systems with chord, a distributed lookup service. In *Proceedings of the Eighth Workshop on Hot Topics in Operating Systems*. 8th HotOS, IEEE Computer Society.

Guha, S., Daswani, N., and Jain, R. (2006). An experimental study of the skype peer-to-peer voip system. In *5th IPTPS06*.

Gupta, R. and Somani, A. K. (2004). Pricing strategy for incentivizing selfish nodes to share resources in peer-to-peer (p2p) networks. In *12th (ICON04)*.

Heckel, R. (2005). Stochastic analysis of graph transformation systems: A case study in p2p networks. In *ICTAC05*. Springer-Verlag.

Heckel, R., Lajios, G., and Menge, S. (2006). Stochastic graph transformation systems. *Fundamenta Informaticae*.

Idrees, F. and Khan, U. A. (2008). A generic technique for voice over internet protocol (voip) traffic detection. *International Journal of Computer Science and Network Security*.

Ji, L. C. (2004). Computation in peer-to-peer networks. Technical report, Department of Computer Science, University of Saskatchewan, Canada.

Khan, A., Heckel, R., Torrini, P., and Rth, I. (2010). Model-based stochastic simulation of p2p voip using graph transformation system. In *17th Int. Conf. Analytical and Stochastic Modeling Techniques and Applications, ASMTA (2010)*.

Khan, A., Torrini, P., and Heckel, R. (2009). Model-based simulation of voip network reconfiguration using graph transformation system. In *Vol.17,(ICGT)EASST*.

Lara, J. D. (2007). Attributed graph transformation with node type inheritance. *Theor. Comput. Sci. In Fundamental Aspects of Software Engineering*.

L'Ecuyer, P. L., Meliani, L., and Vaucher, J. (2002). SSJ:a framework for stochastic simulation in Java. In *Winter Simulation Conference*.

Lo, V., Zhou, D., Liu, Y., Dickey, C. G., and Li, J. (2005). Scalable super node selection in peer-to-peer overlay networks. In *2nd HOT-P2P Workshop*.

Lysne, O., Montaana, J., Pinkston, T., Duato, J., Skeie, T., and Flich, J. (2005). Simple deadlock-free dynamic network reconfiguration. In Boug, L. and Prasanna, V., editors, *High Performance Computing - HiPC 2004*, volume 3296 of *Lecture Notes in Computer Science*, pages 277–316. Springer Berlin / Heidelberg.

Milojicic, D. S. (2002). Peer-to-peer computing. Technical report, HP Labs, Palo Alto.

Montresor, A. (2004). A robust protocol for building super peer overlay topologies. Technical report, University of Bologna, Italy.

NS2 (2008). *The Network Simulator-NS2, http://www.isi.edu/nsnam/ns/*.

Oneil, H., Jeong, J. K., and Kwon, D. (2000). Transport layer identification of p2p super node. In *hhhhhhhh*.

Ren, S., Guo, L., and Zhang, X. (2006). Asap: an as-aware peer-relay protocol for high quality voip. In *26th Int. Conf. on Distributed Computing Systems (ICDCS'06), Lisbon, Portugal*.

Rowstron, A. and Druschel, P. (Nov. 2001). Pastry:scalable, decentralized object location and routing for large-scale peer-to-peer systems. In *8th Int. Conf. on Distributed Systems Platforms*. Heidelberg, Germany.

Sargent, R. G. (1996). Verifying and validating simulation models. In *Winter Simulation Conference*.

Skype (2011). *The Skype Ltd, http://www.skype.com*.

SkypeJournal (2010). Skype sn crashed. http://skypejournal.com.

SkypeStatistics (2011). Skype statistics. http://aaytch.com.

Stutzbach, D. and Rejaie, R. (2006). Understanding churn in peer-to-peer networks. In *Proceedings of the 6th ACM SIGCOMM conference on Internet measurement*.

Torrini, P., Heckel, R., and Rath, I. (2010). Stochastic simulation of graph transformation systems. In *FACE2010*.

Whitner, B. and Balci, O. (1989). Guidelines for selecting and using simulation model verification techniques. In *Winter Simulation Conference*.

Yang, B. and Garcia-Molina, H. (2003). Designing a super-peer network. In *19th Int. Conf. on Data Engineering (ICDE),Bangalore, India*.

Zhao, B. (2003). Tapestry: A resilient global-scale overlay for service deployment. *IEEE Journal on Selected Areas in Communications*.