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P2P IP Telephony over wireless ad-hoc networks

A smart approach on super node admission

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Abstract This paper presents a new strategy to form P2P IP Telephony overlay for wireless ad-hoc networks. In the proposed strategy a structured P2P system is considered where some nodes, called super-nodes, with higher capacity form the overlay and provide registry and call routing services. As selection and admission of new super-nodes in wireless ad-hoc networks is more challenging than backbone networks, we define the strategies to select and admit new super-nodes into the overlay. On one hand, scarce resources and fluctuating link quality demand additional criteria than just node computing resources for super-nodes selection. On the other hand, the indiscriminate increase in super-node number can raise the call session setup delay and degrade the quality. This is due to the relaying of packets across multiple wireless links. In this paper, we first define the criteria to select super-nodes and then the major part of the paper is dedicated to defining the required strategies to admit new super-nodes. Our admission strategies add new super-nodes to the system whenever they are required. Since the strategy does not simply admit all eligible super-node candidates, this ensures control over the number of super-nodes and keeps the session setup delay within to the required service level threshold. We define a queuing network to model our system and evaluate the efficacy of our admission strategies with intensive simulations. Furthermore, we have implemented a P2P IP Telephony system that operates on wireless ad-hoc networks and validated the performance of our admission strategies on this real platform.

Keywords IP Telephony · P2P · Wireless ad-hoc networks · Distributed hash table

1 Introduction

Providing support of IP Telephony services over wireless ad-hoc networks enables interesting use-case scenarios. For instance, in-campus ad-hoc IP Telephony is an attractive use-case. Colleagues and classmates in a campus share the resources of their laptops or PDAs' to organize a peer-to-peer (P2P) overlay on an ad-hoc network and benefit from various free services such as: voice call, instant messaging, presence and video conferencing. Two interesting aspects of such systems can be mentioned as follows. (i) The intra-campus traffic with only local relevance stays local; and (ii) no infrastructure is required.

To provide IP Telephony services, the first step is to establish a *service control system*. This is the signaling infrastructure part of an IP Telephony system which localizes the clients, processes the queries and routes them to the correct destinations [1, 34].

The service model designed to work on top of wireless ad-hoc networks needs to be distributed, self-organized, scalable, dynamic and reliable. Tradition-

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ally, IP Telephony systems follow the client/server model (e.g., H323 [18] and Session Initiation Protocol (SIP) [32]) which contradict these requirements. P2P systems, according to their architectural analogy with wireless ad-hoc networks, could be an appropriate service model solution. Nevertheless, in a wireless ad-hoc network it cannot be expected that all the nodes participate in creating a P2P service overlay. This is mainly because the devices which take part in the network vary in computational power, storage capacity and battery life. Some P2P systems, such as Skype follow a *multi-layer P2P overlay* model [4]. In this model, a certain number of nodes (instead of all the nodes) are selected to form a P2P overlay [20]. These nodes are typically called *super-nodes* (SN). Super-nodes share their computing/memory capacity to (i) host the distributed services, (ii) provide a distributed lookup service and (iii) route a query from source to destination. Hierarchical P2P systems have been shown to be suitable solutions for wireless ad-hoc networks [10]. However, with scarce computing resources and fluctuating link quality in wireless ad-hoc networks, the problem of super-node selection and overlay formation become more challenging.

Super-nodes selection for the P2P applications which are designed to work in backbone networks, such as the Internet, with high speed routers are less challenging. For instance, in Skype, a node with sufficient processing resources which is not behind a Network Address Translator (NAT) and has been active for a certain period of time will be selected to serve as super-node [4]. In wireless ad-hoc networks, there are other important criteria for selecting super-nodes, such as, physical location, relative distances of super-nodes, the distribution of key space and finally the accessibility of super-nodes. Indeed, the overlay should be aware of the underlaying network condition in order to work efficiently.

There is a good deal of research on the formation of network-aware overlays [2, 19, 29, 39] in the Internet, where the topology in terms of open links between different super-nodes or between super-nodes and clients is optimized based on the location information or bandwidth limits. In these works, any node with enough computing capacity will become a super-node and the strategy is to have as many super-nodes as possible. Since there are always enough super-nodes in the system, no super-node admission strategy is defined. Moreover, the optimizations in most of these research efforts are based on the requirements of large scale networks (i.e., Internet) for file sharing applications. For overlays design in wireless ad-hoc networks, super-nodes should be admitted in the overlay whenever the performance of the system falls below an acceptable level. In a wireless ad-hoc network, admitting as many super-nodes as possible in the overlay before they are needed may not improve the performance but also increase the look-up delay. This means that there should be an admission strategy that (i) detects the degradation of the system performance and (ii) run the appropriate strategy to admit new super-nodes.

The criteria that defines the performance of an overlay depends on the application. In the proposed P2P IP Telephony overlay, we consider the *session setup delay* as our main performance criterion. In this paper, we show that if we do not detect why the system performance is degrading and just select a new super-node based on its computing capacity, the system performance will not necessarily improve. We study the different factors that can lead to performance degradation of an overlay, and seek to find solutions that will provide super-nodes with the capability of detecting these causes of performance degradation. This is vital for a distributed system in the absence of a global system performance monitor. Hence, each super-node is able to evaluate its performance. Then, in the event of performance degradation, (i) the affected super-node can determine the main contributor factor of its performance degradation and (ii) trigger appropriate strategies to admit the new super-nodes that have the potential to alleviate the influences of that specific factor.

We will show that by applying these super-node admission strategies, with less admitted super-nodes as compared to a straightforward admission approach, an overlay can achieve an equal or even better performance level. We implement this system and show its efficiency via experimental results in addition to simulation validations.

The rest of the paper is organized as follows. Section 2, provides the problem statement and the main idea of the proposed solution. In Section 3, we present the overall of our strategy for the IP Telephony P2P overlay formation, suitable for wireless ad-hoc networks. Section 4, introduces our analytical model to calculate session setup delay. Section 5, details our super-node admission strategies. Simulation and Implementation results are respectively presented in Sections 6 and 7. Section 8, describes the related work and Section 9 concludes the paper.

2 Problem statement and main idea

An important issue for adapting and tuning P2P systems, like Skype, for wireless ad-hoc networks is defining the criteria for selection of new super-nodes. In this section, we first present in detail the challenges and then we introduce the main idea of our solution for super-node selection and admission.

2.1 Problem statement—challenges of super-node selection

In Skype, a node with a public IP address and sufficient computing capacity which has been active for a certain period of time will very likely be selected as super-node [35]. However, there are other important criteria for selecting super-nodes in wireless ad-hoc networks. Figure 1 shows three undesirable scenarios that may happen if super-nodes are selected based on the nodes capabilities only.

2.1.1 Unreliable connection

The selected super-nodes should have redundant connections to the other super-nodes and the rest of the overlay; otherwise, with a link break, the super-node may be isolated from the rest of the network. Figure 1a illustrates the case of unreliable super-nodes. As shown, the number of one-hop neighbors can be considered as a factor to measure the accessibility and connection reliability of a super-node in a wireless ad-hoc network.

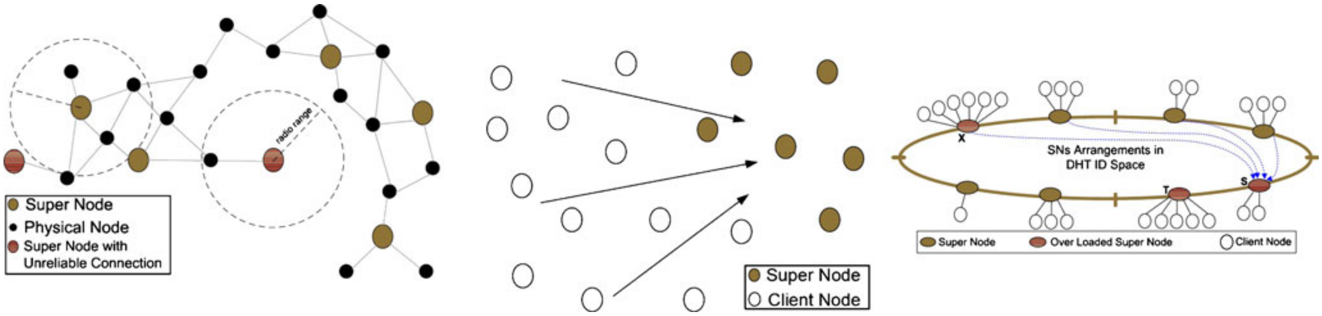
2.1.2 Geographical aggregation

Another factor that impacts the efficiency of P2P overlays is the arrangement of super-nodes in the underlying wireless network topology. Super-nodes should not be aggregated in a certain area of the network; otherwise, as Fig. 1b shows, requests converge toward certain parts of the network and create congested areas.

2.1.3 Unbalanced load

Finally, the load on each super-node has a significant effect on the system performance. The concept of load on each super-node in P2P IP telephony is different from the load in P2P file sharing applications. In file sharing, the size of each object is considerable and the load limit is defined by the number and size of the objects that are kept in a super-node and the bandwidth limit [39]. In contrast, in P2P Telephony, object size is not an issue since the objects are the identifications of the users (SIP URIs) with very low data volume. In P2P Telephony systems, particularly in wireless ad-hoc networks, the hit rate of a super-node indicates the load on it. The hit rate of a super-node is the number of times that this super-node is referred to by the lookup process per unit time. The number of clients of a super-node and the number of neighboring super-nodes that point to a super-node specify the hit rate of that super-node Fig. 1c. Unbalanced hit rate on super-nodes results in some super-nodes becoming bottlenecks and degrade the performance of the system. In structured P2P systems where each object is indexed, the distribution of super-nodes in the ID space should be in a manner that their hit rate stay balanced. It should be noted that uniform hit-rates are not necessarily achieved by the uniform distribution of super-nodes in ID space Fig. 1c.

Knowing the challenges of super-node selection in wireless ad-hoc networks, the next key question to answer is: *Should we admit as many super-nodes as possible in the overlay? More precisely, given M nodes which satisfy the required criteria to become a super-node, should we admit all of them in the overlay, or should we be selective?* In [14], it has been shown that having more super-nodes don't necessarily lead to better performance in a P2P IP Telephony overlay. Assuming N is the number of super-nodes in an overlay, increasing the number further increases the lookup



(a) Unreliable Connection to the Rest of the Network

(b) Geographically Aggregated super-nodes

(c) DHT Overlay with unbalanced hit rate

Fig. 1 Three undesirable situation of SNs arrangement hops while decreasing the load per super-node. The main contribution of this paper is to define efficient and simple strategies for super-node admission, adapted to wireless ad-hoc network conditions, by answering the following questions:

1. What is the *performance criterion* that can reflect the quality of a P2P overlay for IP telephony systems?
2. Among the different possible "factors", which is(are) the one(s) that caused the overlay to not perform adequately?
3. Among the client nodes, which are legitimate candidates to become super-nodes?
4. Among the existing candidates, which is the best candidate to be admitted to alleviate the stress on the overloaded super-node and most effectively address the degradation?

2.2 Main idea—super-node selection based on session-setup-delay

Session setup delay (also known as *post-dialing delay*) is the time elapsed between the last dialed digit and receiving ring-back. This criterion is basically influenced by the signaling protocol and lookup strategy, and quantifies the performance of a telephony system [12]. The ITU-T E.721 standard [17] recommends an average delay of not more than 3.0, 5.0 and 8.0 s, for local, toll and international calls, respectively. We also take the session setup delay as our performance criterion which indicates how good our P2P overlay is serving its clients. All mentioned challenges in Section 2.1 are reflected in this criterion. Unreliable links, congested network and over-loaded super-nodes, all influence the session setup delay.

Letting Δ denote the session setup delay threshold, the P2P IP Telephony overlay is deemed to have degraded below acceptable levels when $\Delta_{SE} > \Delta$, where Δ_{SE} is the average delay of session setup in the system. The goal is to keep $\Delta_{SE} < \Delta$ with the minimum number of admitted super-nodes. To this end, whenever the session setup delay crosses the threshold, we admit new super-nodes to improve the performance of the system. The super-node admission process first determines the cause of the performance degradation, and then applies a specified strategy to select the best candidate to become a super-node to alleviate the cause of high session establishment delay.

According to the analysis that is presented in Section 4, the following factors has been identified for having the greatest effect on the session setup delay of a P2P lookup system deployed in a wireless ad-hoc network:

(1) Node transmission range;
(2) Mean lookup hop count;
(3) Path length between super-nodes;
(4) Reliability of super-node connection;
(5) Super-node hit rate.

Node transmission range is a network configuration parameter. In wireless ad-hoc networks using IEEE 802.11 Medium Access Control (MAC) technology, the increase in transmission range, on one hand, decreases the physical hop count while, on the other hand, increases the collision probability [6]. The adjustment of this parameter depends on the nodes densities and is a MAC layer strategy [6]. We consider the optimization of this parameter beyond the scope of this paper since we have no control over this parameter in the overlay configuration level.

Mean lookup hop count depends on the semantic of the deployed lookup system in the overlay. Hence, it is not directly dependent on the strategy of super-node selection. In non-structured P2P systems, lookup hop count is in the order of $O(N)$ and in Distributed Hash Table (DHT)-based structured P2P systems, it is in the order of $O(\log N)$ where N is the number of super-nodes in the overlay [21]. Mean DHT lookup hop count can be reduced by the deployment of clustering strategies, and this has been well studied in the literature [10, 13]; we also consider to have clustering based on nodes vicinity in our overlay structure.

Among these factors, the last three (path length, connection reliability and hit rate) are tightly dependent to the strategy of super-node selection. Defining the strategies to optimize the path length between the super-nodes has been the focus of several research efforts and well addressed in the current literature [19]. Therefore, we do not focus on this aspect in this paper.

The reliability of a super-node's connectivity is a factor that should be considered as an important policy criteria for admission of a new super-node. In [15], this factor has been explored in detail, and a *connectivity index* that realistically reflect the connectivity of a super-node to the rest of the overlay has been defined. In this work, the index was used to identify legitimate client nodes that can become a super-node when required by the system. Each serving super-node creates a *candidate list* based on the hardware capabilities (CPU speed, storage capacity, power consumption etc.), active time and finally connectivity index.

The last factor, that is the super-node hit rate, is the main focus of this paper. We define the hit rate as the number of times that a super-node receives a lookup request per time unit. The overloaded super-nodes with high hit rates become the bottlenecks in the system and cause delays in session establishment. A super-node receives two kinds of requests: (i) the requests which are terminated at this super-node, and (ii) the enroute requests which should be forwarded by this super-node. The lookup requests arrive at a super-node from other super-nodes which have it in their neighbor list. We call them *pointing super-nodes*. Therefore, the numbers of clients which are registered with a super-node as well as the number of associated pointing super-nodes influence its hit rate. We define a dedicated strategy for new super-node admission for each of the factors influencing the hit-rate.

3 General architecture and overlay formation strategy

In this paper, we consider a *structured P2P* system to organize our service overlay. Structured P2P systems index the objects and define some rules to store the information using DHT [20]. DHT provides a scalable lookup system and is used in many P2P systems such as [11, 26]. In comparison to non-structured P2P systems, structured P2P systems are more complicated in terms of management but more efficient in lookup [21]. We exploit structured P2P for this lookup efficiency. Moreover, DHT allows the deployment of a self-organized P2P system that is suitable for wireless ad-hoc networks.

Despite the advantages, P2P systems using DHT suffer from inefficient routing. To locate and reach a destination, the request passes through a certain number of super-nodes which are spread across the network, even when the destination node is physically in the vicinity of the source node. In the Internet where high speed routers form the network infrastructure, this may be tolerable. But, in wireless ad-hoc networks with limited energy resources and scarce bandwidth, it incurs considerable overheads.

There are some research efforts that consider the location of nodes as a factor, in addition to hash key, when organizing the super-nodes, e.g., [10, 13] and [38]. They basically follow the idea of proximity based clustering and form a two (or multi) level overlay. For instance, in [13] the proposed scheme uses two levels of overlay: Global Overlay and Local Overlay. One copy of the information will be stored locally in the physically closest super-node (cluster head) and another one is stored globally independent of the location in a super-node based on DHT rules. The Global Overlay is a DHT-based overlay which keeps track of each user/resource according to its key which is obtained by a hash function. In this paper, we also consider a two-layer cluster/DHT overlay to cope with inefficient routing issue.

3.1 General architecture

3.1.1 DHT technology

We utilize Chord [36] as our DHT algorithm. However, Chord can be replaced by any other DHT implementation without affecting the architecture since our design is not dependent on the Chord signaling.

3.1.2 Two node classes

We consider the existence of Client Nodes (CNs) and super-node. Super-nodes participate in overlay networks formation as well as process and route service queries.

3.1.3 Bootstrap super nodes

We assume the existence of three bootstrap super-nodes. The assumption of bootstrap super-nodes is not a vital assumption and it just guarantees a minimum level of service in the initial phase.

3.1.4 Clustering

The network is divided into the clusters according to the proximity of the nodes. Each cluster is managed by a *Cluster Head* (CH) which is also a super-node. Besides registering with the super-node with closet ID (based on DHT rules), all the CNs within a cluster register with their CH. This is called *local registration*. In the bootstrap phase, all the bootstrap super-nodes become CHs.

3.1.5 Location information

The mechanisms used for estimating the location are called localization protocols. In our system, we assume the existence of a localization system that allows each node to estimate its position in the network. Existence of a location estimation system is not vital and is a complementary assumption that we consider. We evaluate how such kind of information can enhance the performance of the overlay network. To justify the feasibility of this assumption, we should indicate that there is not necessarily a need for the deployment of an extra system such as GPS (Global Positioning System) for the purpose of localization. There have been extensive research efforts on localization in sensor networks [16] and one class of approaches is known as *range-free* localization mechanisms. In these mechanisms, some powerful nodes (typically called anchors or reference nodes) are aware of their location and they periodically beacon their location information to their neighbors. Using this information, the listening nodes estimate their location with some simple procedure, e.g., partition the environment into triangular regions between beaconing nodes (anchors) and determine if a node is inside or outside a triangular area [16]. Then, by narrowing the triangular area, the center of gravity (CoG) is assumed to be the node location.

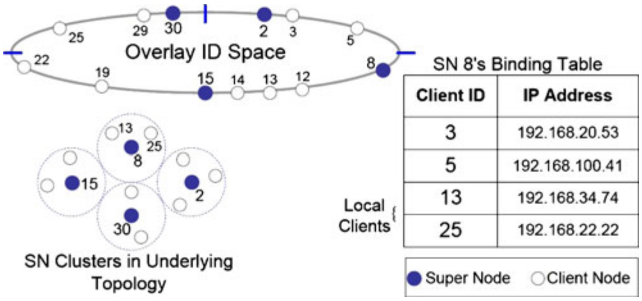
3.2 Overall overlay formation strategy

3.2.1 Initialization

The bootstrap nodes start sending advertisement messages to initiate overlay network construction. The advertisement messages are used for two purposes: (i) discovering other bootstrap nodes to create the DHT overlay, and (ii) informing client nodes to join the cluster. Upon receiving an advertisement message, every bootstrap node updates its *finger table* and the links between overlay members will be created. On the other hand, the client nodes which desire to use the service of this overlay join the cluster of the nearest bootstrap node and register with the bootstrap supernode which has the closest ID in DHT-ID-space.

3.2.2 Localization (registration)

All the users of our P2P IP Telephony system are identified by a contact-ID which may be a SIP URI [5] in the form *user-name@domain-name*. The overlay system should store the binding between a user contact-ID and the node IP address (Fig. 2). Then, by hashing on the contact ID a unique key is provided in the overlay ID space for each user. On the other hand, by hashing on the IP addresses of super-nodes, each super-node is also assigned an ID in the overlay ID space. A super-node registers all the keys smaller than its ID and greater than its predecessor's ID. All the users whose keys fall in this range register with this super-node. This is called *global registration*. In our two-layer overlay approach, a user key is also registered with the Cluster Head (local registration). Figure 2 shows a binding table sample with local and global registration items. In this example, the keys 13 and 25 belong to local clients and keys 3 and 5 are registered with $SN8$ (short for Super-node #8) according to the DHT rules (global registration).



SN 8's Binding Table

Client ID	IP Address
3	192.168.20.53
5	192.168.100.41
13	192.168.34.74
25	192.168.22.22

● Super Node ○ Client Node

Fig. 2 Localization and binding table

3.2.3 Query processing (call setup)

In order to establish a call, the caller should obtain the IP address and port number of the other party. This information is provided by the DHT overlay for the clients. A caller submits a request containing the key of the callee (SIP INVITE) to the local super-node (i.e., CH) with which it has locally registered. If the callee resides in the same cluster or has globally registered with this super-node, the CH returns the requested IP address to the caller and the lookup process is finished. If not, the CH starts the lookup process in the DHT overlay and forwards the request to its finger with ID that is nearest to the callee's key. This process continues until the query arrives at a super-node with which this key is registered. To see how this system works, consider the example in Fig. 3 where Alice wants to communicate with Bob@int-edu.eu. Then, Alice should find the super-node with which Bob has registered. In the first step, Alice hashes Bob's ID to obtain the key and submits it to $SN6$. $SN6$ is the nearest super-node in Alice vicinity. In this example, $Key_{\text{Bob}} = H('Bob@int\text{-}edu.eu')$ gives $f6e2$ which is among $SN6$'s finger list. This request points to $SN9$ which has the closest ID to Key_{Bob}. Subsequently in $SN9$, the closest finger to Bob's key is $SN7$ and the request is transferred to it. This process repeats until the request finally arrives at $SN4$ with which Bob has registered.

3.2.4 Creating the super-node candidates list

Each super-node looks for legitimate nodes among the CNs of its cluster to create a super-node candidate list.

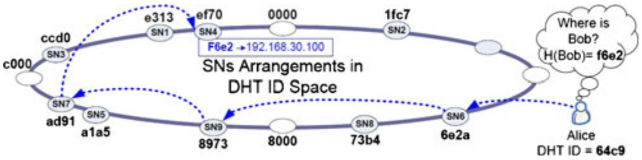


Fig. 3 Call routing in DHT overlay (Alice calls Bob)

This list is formed based on the computing capability of a node as well as the connectivity index (CI) which measures how well a node is connected to the rest of the network. In [15], a connectivity index which derives its idea from the field of molecular chemistry has been proposed. Randic [27] proposed the "Randic Connectivity Index (CI)" to express the branching (or connectedness) of a molecular structure. The Randic Connectivity Index definition is extended for general weighted undirected graph as follows:

$$\chi(G) = \sum_{u,v \in V(G)} \frac{w_{u,v}}{\sqrt{d(u).d(v)}} \tag{1}$$

where $0 \leq w_{u,v} \leq 1$ is the weight of the edge adjacent to vertices u and v, and $d(u)$ and $d(v)$ represent the degree of vertex u and v respectively, while $V(G)$ represents the vertex set of the graph. The Randic index is defined to represent the connectedness of a whole graph; however, the index proposed in [15] can quantify connectivity of a node to the rest of the network. There is a compromise between accuracy and cost to quantify the node level connectivity. The most cost-effective but imprecise way of estimating a node's connectivity is to count the first-hop neighbors and consider the number as the connectivity index. The approach adopted in [15] is to associate a 2-hop sub-graph to each node, and find the Randic CI of the sub-graph. Since our index should also be able to reflect the connectivity due to link-quality, we have to consider the link-qualities of links in the CI calculation. Link quality is considered as the weight of the corresponding arc in a weighted graph, and packet delivery ratio (PDR) can be considered as an example of such weights for the links. The CI of a node v is calculated as shown in Eq. 2.

$$CI(v) = \sum_{j \in \text{2-hop neighborhood of } v} \frac{w_{v,j}}{\sqrt{d(v).d(j)}} \tag{2}$$

where $w_{v,j} \in [0, 1]$ represents the weight of the link (v, j). A weight of 0 corresponds to a non-existent link where as 1 represents a link with maximum quality.

In Fig. 4, the actual network graph represented as a labelled graph with 12 vertices and 12 edges. In the same figure, the sub-graphs corresponding to node A and node B is presented.

After we assign a sub-graph to each node, we calculate the CI of the sub-graph. For example, for the sub-graphs of Fig. 4 with ideal links ($w_{v,j} = 1$), the CIs are:

$$CI(Node\ A) = 1.73205$$

$$CI(Node\ B) = 2.94338$$

More details about this CI and its efficiency can be found in [15].

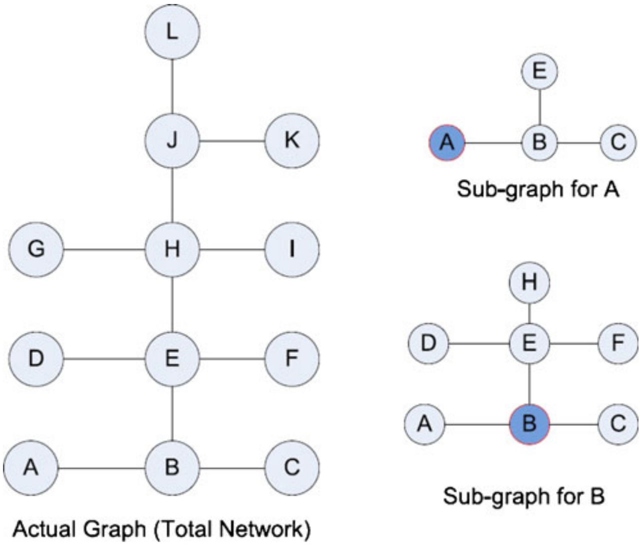


Fig. 4 Illustration of connectivity index calculation

4 Performance metrics analysis

In this section the factors that influence the *session setup delay* in the proposed IP Telephony P2P overlay is discussed.

Session setup delay is one of the standard metrics that shows the performance quality of a telephony system. We analyze the average session setup delay to evaluate the performance of the proposed P2P overlay. This metric is denoted by Δ_{SE}. As explained in Section 3.2.3, in a P2P overlay, before a call setup request reaches the destination, it passes through a number of intermediate super-nodes. For simplicity, we assume that the path as well as the end-to-end transmission time for the both directions between any source and destination is same. With this assumption we formulate Δ_{SE} as $\Delta_{SE} = 2\Delta_{Tr} + \Delta_W$ where Δ_{Tr} is the *mean end-to-end transmission time* from source to destination, and Δ_W is the *mean waiting time* spent in all intermediate super-nodes in route toward destination. Continuing, we calculate and analyze each of Δ_{Tr} and Δ_W.

4.1 Mean E2E transmission time (Δ_{Tr})

The value of Δ_{Tr} depends on the mean number of intermediate physical hops between sources and destinations. We note that one hop at the overlay level between two neighboring super-nodes may contain several physical hops at the network level. So, if we let L denote the mean number of intermediate hops between two super-nodes at the physical network level, M denote the mean number of visited super-nodes (at the overlay level) before arriving at the destination, and τ_{tr} denote the mean transmission time between two physical neighbors, then we have $\Delta_{Tr} = M \times L \times \tau_{tr}$ We now discuss each of the parameters in this equation in detail.

(i) L is the mean number of intermediate physical hops between two super-nodes. This parameter is very dependent on the arrangement of super-nodes in the physical network. Hence, at the selection time of super-nodes, there should be some strategy to avoid selecting distant nodes. Moreover, in a wireless ad-hoc network the transmission range of nodes can affect L. Longer transmission range decreases the value of L. However, increasing the transmission range will not necessarily lead to a reduction in session setup delay. In fact, as we we analyze τ_{tr} later, we will see that increasing the transmission range can degrade τ_{tr}.

(ii) M is the mean number of visited super-nodes in a lookup process before arriving at the destination. M represents the number of overlay hops. The order of M depends on the semantic of the lookup routing. In the most DHT technologies, such as Chord, CAN and Pastry, the lookup hop count is in the order of $O(\log N)$ where N is the number of super-nodes in the overlay. However, it is hard to derive an abstract formula to calculate the exact value of M. In [14], the value of M for the special case of load balanced Chord has been derived, as shown in Eq. 3.

$$M = 1 + \frac{\log N}{2} \tag{3}$$

Clustering and the use of proximity information can decrease the value of M and speed up the DHT lookup process. To evaluate how and when clustering can improve the lookup process, we define the *absorption probability* ($P[A]$). For a given SN_i, we define this probability as the probability that the sought contact is found in this super-node and we denote it as $P[A_i]$. Higher values of $P[A]$ for the super-nodes of an overlay leads to shorter lookup path and smaller M. Denoting the number of client nodes that have registered their IP Telephony contact information with SN_i through the DHT rules as n_{D_i} and the total number of clients in the P2P overlay system as n, then:

$$P[A_i] = \frac{n_{D_i}}{n} \tag{4}$$

Clustering can improve the absorption probability of super-nodes. If super-nodes organize and manage clusters, they will get additional registered clients through the local registration (i.e., local clients). In this case, if n_{C_i} local clients are in the cluster C_i managed by SN_i, the probability that a contact be found in this super-node is given by Eq. 5, as follows:

$$P[A_i] = \frac{n_{D_i} + n_{C_i} - n_{\{C_i \cap D_i\}}}{n} \tag{5}$$

where $n_{\{C_i \cap D_i\}}$ is the number of local clients of SN_i that have also registered with this super-node based on DHT rules.

As shown in Eq. 5, with a given number of cluster members (n_{C_i}), clustering will be most effective means to increase the absorption probability when $n_{\{C_i \cap D_i\}} = 0$, which refers to the case where there is no overlap between the local clients of a super-node and clients that registered with this super-node through the DHT process.

(iii) τ_{tr} is the time interval from the moment that a request is ready to be sent until it is successfully received by the next physical hop. This parameter is tightly dependent on the MAC protocol.

In this work, we adopt the IEEE 802.11 MAC technology for considered wireless ad-hoc network. IEEE 802.11 is a contention-based medium access scheme. Therefore, a node needs to compete with other nodes to access to the medium and sends its packet. There is also the probability of packet collisions during transmission. Hence, τ_{tr} is directly related to the time waiting to access the medium and the number of collision before a successful transmission. In [6], it has been shown that the number of contending nodes exponentially affects the waiting time before transmission in an 802.11 wireless ad-hoc network. The number of contending nodes in a wireless ad-hoc network depends on the density of the nodes and their transmission range. In a network with a fixed node density, increasing the transmission range has two contradicting effects on the session setup delay. On one hand, it leads to a longer channel access waiting time while on the other hand, the hop count decreases. Therefore, there is a trade-off between these two effects to reach the minimum session setup delay. Optimization of transmission power is a MAC and PHY layer strategy, which is beyond the scope of this paper.

4.2 Mean waiting time in intermediate super-nodes (Δ_W)

In order to calculate Δ_W, The P2P overlay is modeled by an open queueing network. An example of the queueing network model with three super-nodes is shown in Fig. 5.

In this model, each super-node is considered as an $M/M/1/K$ queue. SN_i has the service rate of μ_i and queue capacity of $K_i - 1$ (which means the maximum number of jobs in the super-node is K_i). These two parameters are tightly related to the computing and storage capacity of a super-node.

The incoming request rate in each super-node comprises two main elements, namely, new arrivals and forwarded requests from other super-nodes (Fig. 6).

New arrivals come from the local clients in the super-node's cluster. We assume the arriving request traffic is distributed uniformly over all super-nodes. Hence, if the whole incoming request rate in the overlay system is λ, the new arrival rate in SN_i will be equal to $n_i\lambda/n$ where n_i is the number of CNs in the SN_i's cluster and n is the total number of clients in the system. To obtain the arrival rate from other super-nodes, following probabilities is considered: q_{ji} which is the probability that a request arriving at SN_j is to be forwarded to SN_i, and q_{ii} which is the probability that the incoming request at SN_i has reached its destination. It is clear that

$q_{ji} = 0$ if $SN_i \notin \{SN_j\text{'s fingers}\}$.

Then, by defining

$$Q = \begin{pmatrix} q_{11} & \cdots & q_{j1} & \cdots & q_{N1} \\ \vdots & \cdots & \vdots & \cdots & \vdots \\ q_{1i} & \cdots & q_{ji} & \cdots & q_{Ni} \\ \vdots & \cdots & \vdots & \cdots & \vdots \\ q_{1N} & \cdots & q_{jN} & \cdots & q_{NN} \end{pmatrix}_{N\times N},$$

$$D_Q = \begin{pmatrix} q_{11} & 0 & 0 & 0 & 0 \\ 0 & \ddots & 0 & 0 & 0 \\ 0 & 0 & q_{ii} & 0 & 0 \\ 0 & 0 & 0 & \ddots & 0 \\ 0 & 0 & 0 & 0 & q_{NN} \end{pmatrix}_{N\times N}$$

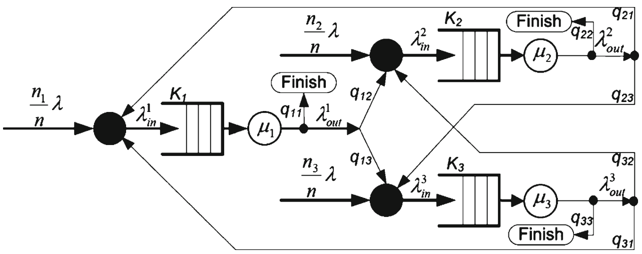


Fig. 5 A sample of queuing network model with three super-nodes

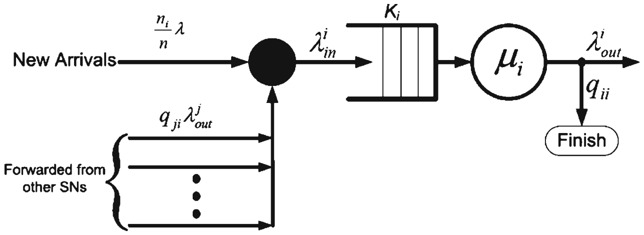


Fig. 6 M/M/1/K model for a super-node

and

$$\Lambda_{\text{new}} = \begin{pmatrix} \frac{n_1\lambda}{n} \\ \vdots \\ \frac{n_i\lambda}{n} \\ \vdots \\ \frac{n_N\lambda}{n} \end{pmatrix}_{N\times 1}$$

the request arrival rate matrix (Λ_{in}) for all super-nodes in the overlay can be calculated by solving the following equations.

$$\Lambda_{\text{in}} = (Q - D_Q)\Lambda_{\text{out}} + \Lambda_{\text{new}} \tag{6}$$

$$\Lambda_{\text{out}} = (I - D_Q)\Lambda_{\text{in}} \tag{7}$$

$$\Rightarrow \Lambda_{\text{in}} = (Q - D_Q)(I - D_Q)\Lambda_{\text{in}} + \Lambda_{\text{new}} \tag{8}$$

which will give $\Lambda_{\text{in}} = [\lambda^i_{\text{in}}]_{N\times 1}$ where λ^i_{in} is the total arrival rate from other super-nodes to SN_i and is expressed as follows:

$$\lambda^i_{\text{in}} = \frac{n_i\lambda}{n} + \sum_{j\neq i} q_{ji}(1 - q_{jj})\lambda^j_{\text{in}} \tag{9}$$

Letting $a_i = \frac{\lambda^i_{\text{in}}}{\mu_i}$ and using *Little*'s formula [7], we can calculate the waiting time in a given SN_i (denoted δ^i_W) as follows:

$$\delta^i_W = \begin{cases} \frac{1}{\mu_i}\left[\frac{a_i}{1 - a_i} - \frac{(K_i + 1)}{1 - a_i^{K_i+1}} \cdot a_i^{K_i+1}\right], & a_i \neq 1 \\ \frac{K_i}{2\mu_i}, & a_i = 1 \end{cases} \tag{10}$$

where $K_i - 1$ and μ_i are respectively the queue size and service rate of SN_i. Then, by defining q_i as the probability that a request passes through SN_i, the mean waiting time in a super-node, called δ_W, can be calculated as:

$$\delta_W = \sum_{i=1}^{N} q_i \delta^i_W \tag{11}$$

where q_i can be estimated as follows:

$$q_i = \frac{\lambda^i_{\text{in}}}{\sum_{j=1}^{N} \lambda^j_{\text{in}}} \tag{12}$$

We also refer to q_i as the *hit rate factor*. Finally, the mean waiting time in all intermediate super-nodes in between source and destination (Δ_W) can be calculated as follows.

$$\Delta_W = M \times \delta_W \tag{13}$$

According to Eq. 10, the three parameters K_i, μ_i and λ^i_{in} affect the waiting time in a given SN_i. K_i and μ_i are dependent to the node capabilities such as processing and storage capacity; they are not related to the P2P lookup design and architecture. Considering that we cannot change a super-node's hardware characteristics, the request arrival rate (λ^i_{in}) is the main parameter affecting the waiting time in a super-node. Figure 7 shows how waiting time in super-node changes with arrival rate.

It is true that λ^i_{in} depends on the request arrival rate (λ) in the whole system while λ is independent of the overlay design. The distribution of overall request load on each super-node affects λ^i_{in} and depends tightly on the DHT overlay design.

Equation 9 shows that λ^i_{in} is mainly influenced by the new and forwarded requests rates. The new requests are from the local clients. Hence, the larger SN_i's cluster population is, the higher the request arrival rate will be. On the other hand, the rate of the requests which are forwarded by other super-nodes depends on the position of SN_i in DHT identifier space. Indeed, the position of a super-node in identifier space determines, firstly, the number of other super-nodes that point to it and, secondly, the number of clients who register their contacts with this super-node. For instance in Fig. 1c, super-node T is located in a position where there is a large number of clients within its covering range in ID space. On the other side, super-node S is overloaded because it is in the finger tables of several other super-nodes and they point to it. The hit rate factor (q_i) which is defined in Eq 12, is a factor defining the proportion of request arrival rate in SN_i to the sum of arriving rates to all the super-nodes. It can be proved that in a system with $\sum_{j=1}^{N} \lambda^j_{\text{in}} = Cte.$, δ_W is minimum when $\forall i,\ \lambda^i_{\text{in}} = \lambda_{\text{in}}$. This implies a load balanced overlay with uniform super-node hit rate.

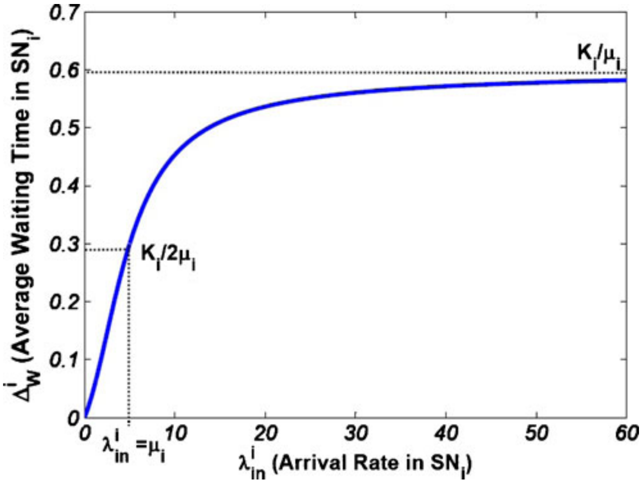


Fig. 7 Waiting time in a super-node vs. arrival rate

In order to achieve a balanced P2P overlay with uniform super-node hit rate, each super-node should be admitted to the system with careful consideration of its ID position in the DHT identifier space. We will address thisz z in Section 5.

5 Super-node admission strategies

The key performance metric of the proposed system, namely, average session setup delay Δ_{SE} must be kept below the predefined threshold Δ in order for the proposed P2P IP Telephony system to meet its performance criteria. We admit new super-nodes to help the overlay recover when $\Delta_{SE} > \Delta$. The main challenge that we have recognized is that there is no central authority in the P2P overlay which is aware of the Δ_{SE} in our system. To this end, we seek a solution to give to each super-node a self-awareness of its performance. Our goal is to be as efficient as possible and recover the system by admitting the minimum number of new super-nodes. Therefore, if each super-node can measure its own performance, then when it detects its performance degrading, it is able to admit the best choice super-node to alleviate the performance degradation.

Considering that $\Delta_{SE} = \Delta_W + 2\Delta_{Tr}$, to give each super-node a self-awareness of its performance, we assume that $\Delta_W = \alpha\Delta_{SE}$ where $\alpha < 1$. A restricted set of conditions to avoid degradation of the system performance would then be as follows:

$$\begin{cases} \Delta_W \leq \alpha\Delta \\ \Delta_{Tr} \leq \dfrac{(1-\alpha)}{2}\Delta \end{cases} \tag{14}$$

To transform the system level conditions in Eq. 14 to the super-node level, we take into account that $\delta_W = \Delta_W / M$ (Eq. 13). Then, the inequalities in Eq. 14 lead us to:

$$\begin{cases} \delta_W \leq \dfrac{\alpha\Delta}{M} \\ L \times \tau_{tr} \leq \dfrac{(1-\alpha)\Delta}{2M} \end{cases} \tag{15}$$

The conditions in Eq. 15 explain how to keep the session setup delay below its threshold:

- The mean waiting time in a given super-node should be less than $\frac{\alpha\Delta}{M}$; and
- The mean transmission time between a given super-node and any of its fingers should be less than $\frac{(1-\alpha)\Delta}{2M}$.

In order to satisfy these two conditions, we enforce more restrictive conditions on the super-nodes, as in the following:

$$\begin{cases} \delta_W^i \leq \dfrac{\alpha\Delta}{M} \\ L_i \times \tau_{tr}^i \leq \dfrac{(1-\alpha)\Delta}{2M} \end{cases} \tag{16}$$

δ_W and $L \times \tau_{tr}$ are respectively the mean values of δ_W^i and $L_i \times \tau_{tr}^i$. Therefore, Eq. 16 automatically satisfies Eq. 15 and consequently keep $\Delta_{SE} \leq \Delta$.

For δ_W^i, there is also another constraint. When the waiting time in SN_i is greater than $\frac{K_i}{2\mu_i}$, it means that $\lambda_{\text{in}}^i \geq \mu_i$ (refer to the Fig. 7). This means that the arrival rate in the super-node is greater or equal to its processing rate; in other words, the super-node is overloaded. This issue defines the new constraint $\delta_W^i \leq \frac{K_i}{2\mu_i}$ and we extend the inequality in Eq. 16 as follows:

$$\delta_W^i \leq Min\left\{\frac{\alpha\Delta}{M}, \frac{K_i}{2\mu_i}\right\} \tag{17}$$

The conditions in Eqs. 16 and 17 define our criteria for detecting the under-performance of the system. Therefore, whenever one of these conditions is not satisfied, a new super-node should be admitted to the system to improve the performance. However, blindly increasing the number of super-nodes may not necessarily lead to better performance of the system. Hence, a dedicated super-node admission strategy should be defined for each factor that affects the performance. In the following subsections, we discuss the conditions and our porposed strategies for super-node admission.

5.1 Increase in number of cluster members

When n_i (the number of local clients) increases, the rate of new request arrivals from the local clients of SN_i and consequently the waiting time (δ_W^i) increase. According to the condition of inequality in Eq. 17, if $\delta_W^i > Min\{\frac{\alpha\Delta}{M}, \frac{K_i}{2\mu_i}\}$, SN_i is overloaded and a new super-node should be admitted. If the new arrivals account for more than half of the total arrival rate of SN_i (i.e., $n_i > \frac{1}{2}\frac{n\lambda_{\text{in}}^i}{\lambda}$), we consider this super-node's cluster to be overcrowded. In this case, a new super-node among the super-node candidates of CH's list will be selected to create a new cluster.

As discussed in Section 3.2.4, each CH creates a list of super-node candidates. These candidates are selected according to their processing and storage capacity as well as their connection reliability. In the case of an overcrowded cluster, in order to choose the best candidate, our strategy is based on the following two considerations.

5.1.1 New super-node ID

As shown in Eq. 5, clustering will be the most effective when the users of a cluster have not already registered with their CH based on DHT rules (i.e., $n_{\{C_i \cap D_i\}} = 0$). To achieve this, the super-node identifier in DHT space should be far from the majority of the local clients' keys. This ensures that most of the new super-node's local and global (DHT) clients do not overlap.

Hence, in the case of an overcrowded cluster, the CH should consider the spectrum of local client keys and choose, from among this list, a super-node candidate which has the farthest ID from the majority of this key spectrum. In case there are several choices among the candidates from the CH list, location information is used to pick the new super-node.

5.1.2 New super-node location

As discussed in Section 4, the physical hop count (distance) en route from source to destination is an important factor influencing the session setup delay. The distance between local clients and their cluster head is part of the end-to-end distance that can be limited with some location considerations at the time of new super-node selection. The shorter the distance between a CH and its members, the faster the query routing will be. Consequently, in the admission process, the location information can be used to make the best choice from the super-node candidate list. To employ such information, we assume that all the CHs maintain information of the farthest node (i.e., largest physical hop count). During cluster splitting, if we let the coordinates of this farthest node be (x_f, y_f) and the coordinates of the CH be (x_{ch}, y_{ch}), we consider the point with coordinates $(\frac{x_f+x_{ch}}{3}, \frac{y_f+y_{ch}}{3})$ as the ideal place to locate the new CH (Fig. 8). Then, we consider an area with radius of 'd' around the ideal point to find a proper candidate to be admitted as new super-node and CH. This area is called *ideal area*. The candidate super-node which is located in this area is given priority to be admitted as the new super-node. With this strategy, the distance of the farthest node to the new cluster head will be

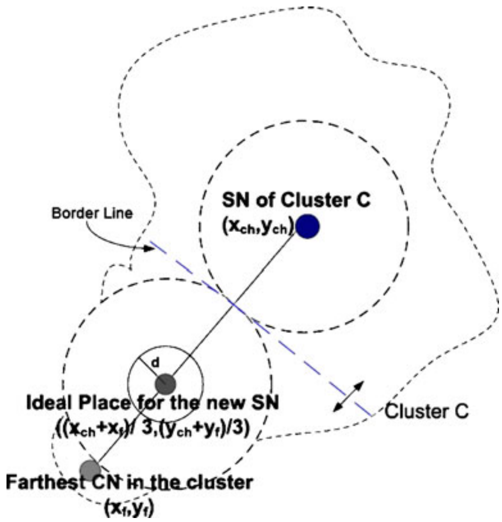


Fig. 8 Best place for new cluster head

approximately one third of its distance to the previous cluster head. Indeed, the border line between the new and existing clusters falls in the middle of their CHs. Therefore, a considerable load of the existing cluster will be transferred to the new CH.

5.2 High hit rate

An ideal P2P system for IP Telephony services is a system with balanced load, i.e., all the super-nodes experience a uniformly shared load. This avoids the situation where a certain number of super-nodes become overloaded and adversely affect the system performance. The overloaded super-nodes are the nodes with high hit rates which increase the session establishment time and aggravate the query failure ratio. A super-node may be overloaded due to the high rate of incoming requests. In such case, a new super-node should be admitted into the overlay to alleviate the load on the overloaded super-node. Similar to the previous section, if the waiting time (δ^i_W) of a request to be processed by SN_i exceeds the limit, this super-node should trigger the admission process for a new super-node. If the rate of the forwarded requests from other super-nodes produce more than half of the whole arrival rate in SN_i, the overloading is considered to be from the high hit rate of this super-node and not the local clients.

As explained in Section 4, the position of a super-node in the DHT ID space determines its hit rate for this super-node. Therefore, the strategy for super-node admission in such a case should be defined based on the location of the super-node candidates in the DHT ID space. The location of a super-node in DHT ID space defines (i) the number of global clients that register with it based on DHT rules and (ii) the number of other super-nodes that point to it (Fig. 1c). Suppose SN_i is the overloaded super-node, then an acceptable choice for a new super-node, is a candidate SN_C such that:

$$SN^i_p < SN_C < SN_i \tag{18}$$

where SN^i_p is the first predecessor of SN_i. With such choice, the contact IDs in the range (SN^i_p, SN_C) which are already assigned to the overloaded SN_i will be transferred to the new super-node. This decreases the number of global clients of SN_i, and the best candidate is the one that takes half of the global clients from the overloaded super-node.

Next, SN_C replaces SN_i as the *j*th finger of the super-nodes, e.g., S such that:

$$\forall S, j: \quad S + 2^j \in (SN^i_p, SN_C) \tag{19}$$

In order to find a super-node candidate that meets the condition specified in Eq. 18, our order of search is as follows: first within the cluster, then in the fingers, and lastly, in all the super-nodes. In each phase, once a super-node candidate meeting the necessary conditions is found, the search process will stop.

5.3 Long distance between super-node and its fingers

From our discussion in Section 4, the mean physical hop number (L) between a super-node and its fingers affects the Δ_{Tr} part of session setup delay. Therefore, a key strategy in the creation of the P2P overlay systems for IP Telephony services is to select super-nodes such that each SN_i's mean distance to its fingers remains in the order defined by Eq. 16.

In our system, each SN_i determines its mean distance to its fingers. Let L_{ij} denote the number of physical hops between SN_i and its *j*-th finger. Then, the mean hop number between SN_i and its fingers, which we denote by L_i, can be calculated based on the distance and the hit rate of each finger as follows:

$$L_i = \sum_j q_{ij} L_{ij} \tag{20}$$

To calculate L_i, each SN_i estimates q_{ij} for each finger j, as follows:

$$q_{ij} = \frac{NQ_{ij}}{NQ_i} \tag{21}$$

where

NQ_i	Total number of queries received in SN_i;
NQ_{ij}	Number of queries with j as the next overlay hop.

With L_i, if SN_i detects that the conditions in Eq. 16 are not satisfied and its $L_i \times \tau_{tr}^i$ value exceeds the limit, it should trigger the admission process for a new super-node. In this case, since the current super-node is far from its fingers, the admission strategy should seek to replace SN_i with a better positioned super-node. SN_i, therefore chooses from among the candidates of its list, the one with smallest mean distance to the fingers; for distance calculation, the location information is utilized. In addition, for the new super-node to have exactly the same finger list, SN_i devolves its ID to the new super-node.

6 Simulation model and results

We model our P2P overlay using the queueing network model shown in the Fig. 5. We consider a 50 × 50 grid (2,500 physical node) shown in Fig. 9 as our test-bed to create the underlying wireless meshed network. At the beginning, none of these physical nodes are active in the overlay except three bootstrap super-nodes. As time progresses, more nodes become active and join the overlay as clients. The hash function calculates the hash ID for each client and each client registers with the super-node having the closest ID. Call requests arrive in the system at a rate proportional to the number of client nodes. Each call request is identified by a (source, destination) pair with the source and destination chosen randomly from among the client nodes. For each call request, the physical hop count and the overlay hop distance are determined. The hit rates of super-nodes in each lookup process are also recalculated and updated. Then, by employing a moving window averaging method over these gathered information, the Q matrix (cf: Section 4.2) is obtained and the mean session setup delay for the system is calculated. In other words, we use a combination of simulation and analytical approach to evaluate the performance of our system.

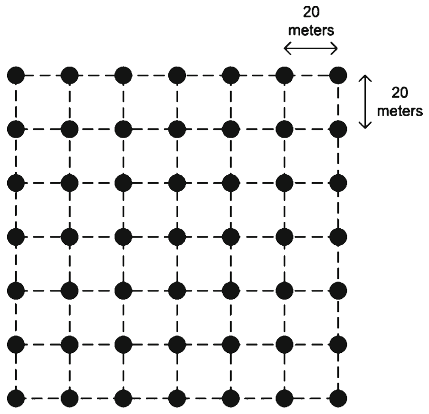


Fig. 9 Simulation test-bed: wireless meshed network

6.1 Overlay construction strategies

We considered three overlay construction strategies and compared their performance according to the metrics defined in Section 6.2.

6.1.1 DHT

In this strategy, a single layer DHT overlay is implemented without clustering. Whenever $\Delta_{SE} > \Delta$, new super-nodes will be admitted into the system incrementally until the session setup delay falls below the threshold. None of the defined admission strategies in Section 5 is applied and super-node admission is said to be blind.

6.1.2 DHT and clustering

In this strategy, we add clustering to the previous strategy; however, we still do not apply any of our proposed admission strategies.

6.1.3 Smart admission

In this strategy, we apply our admission strategies together with *DHT* and *Clustering*.

6.2 Performance metrics

Our main performance metric, based on which we optimize our system and decide when to admit new super-nodes, is the *session setup delay*. However, to better reflect different aspects of our system performance and compare it to the other strategies, we also use other metrics, which are listed below:

6.2.1 Session Setup Delay (SSD)

This is the main performance metric of the proposed P2P IP Telephony overlay. It shows how fast our overlay system can find the destination and establish a session.

6.2.2 Mean hop-count

This is the mean number of intermediate physical hops $(M \times L)$ a query passes through en route to its destination. This metric has direct impact on the session setup delay (Δ_{SE}) and reflects the efficiency of query routing more accurately.

6.2.3 Query failure

In our model, a query may fail because of three reasons: poor link quality, saturated super-node, and TTL (Time to Live) expiration. In this work, we calculate the query failure ratio with and without TTL limitation.

6.2.4 Load distribution factor

To have an estimation of the load distribution over the super-nodes, we obtain the standard deviation of the request load on the super-nodes. This metric shows how uniform is the load on the super-nodes of the system. The lower is this factor, the more uniform is the load distribution.

6.3 Simulation parameters

In our grid network (as shown in Fig. 9), the horizontal and vertical separation between adjacent nodes is set to 20 m. We assume that the transmission range of each node is also 20 m.

Among the physical nodes, we assume the existence of three bootstrap super-nodes that form the overlay network during network initialization; we have fixed their locations in the network. Then, among the other nodes, one will be selected randomly to become active in the overlay according to a *Poisson* random process with rate, denoted by λ_{active}, of two per minute (2/min). The duration that a node remains active in the overlay is an exponential random variable with mean, μ_{active}, equals to 30 min. Each active client node is assigned a parameter pair (K, μ) representing its queue capacity and processing rate. K and μ are uniform random variables, in the ranges $[1, 5]$ and [5/min, 20/min], respectively. These are important parameters for selecting the super-node candidate.

The SHA-1 hashing algorithm with 12 binary digits is used for the hash function in our DHT system. This creates an ID space with the range $[0, 2^{12} - 1]$. Call setup requests are initiated based on a *Poisson* random process with rate, λ, where $\lambda = (0.8\ n)$/min with n being the number of clients in the system. Each request specifies a (source, destination) pair, randomly chosen from among the registered peers in the P2P DHT overlay. In our system, we consider the session setup delay limit, Δ, to be 1.5 s. Lastly, we should mention that all the simulation results fall in the confidence interval of 96%.

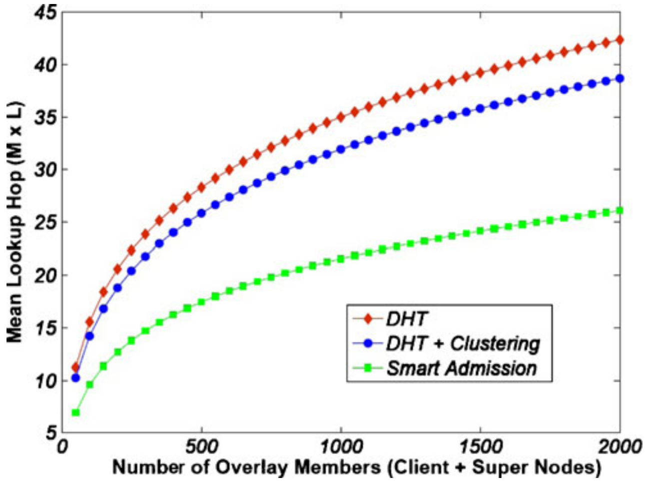


Fig. 10 Mean physical hop count ($M \times L$) in a DHT lookup process

6.4 Simulation results

Figure 10 shows the mean lookup hop versus the number of overlay members (including super and client nodes). Figure 11 illustrates the influence of our admission strategy in limiting the session setup delay (Δ_{SE}). The knee points in other two strategies represent the saturation of super-nodes. The improvement we have obtained is due to (i) reducing the number of physical hops between super-nodes and their clients, (ii) controlling the number of admitted super-nodes, and (iii) uniform load distribution among super-nodes.

The simulation results for *query failure ratio* when there is no TTL restriction is presented in Fig. 12. The reasons for query failure, in this case, are link failure and super-node saturation. To obtain these results, we

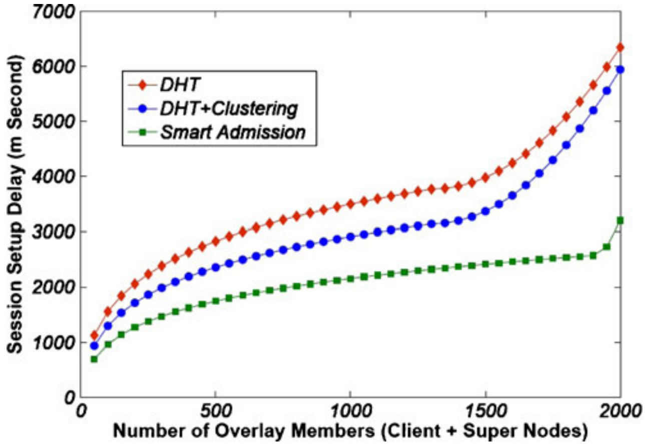


Fig. 11 Session setup delay vs. the number of overlay population

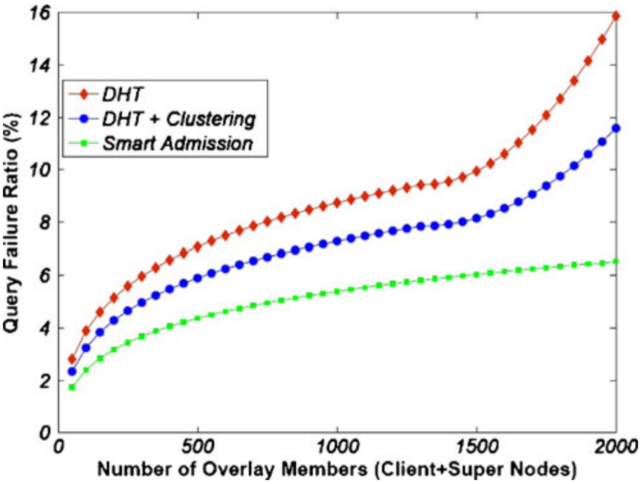


Fig. 12 Query failure ratio—no TTL limit

have applied a link failure probability equal to .01 to every link between two adjacent node.

When there is no TTL restriction, the smart admission strategies that we have defined restrict the query failure ratio to less than 7%. Figure 13 shows this metric when TTL is set to 20. In IP Telephony applications, setting a TTL limit is important for security reasons in order to avoid packets circulating in the network for a long time.

To show the load distribution factor, we have captured a snap-shot of the standard deviation of request load on the super-nodes in Fig. 14. This figure shows a sample of load distribution condition in the system, at steady state, from time t_0 for T seconds.

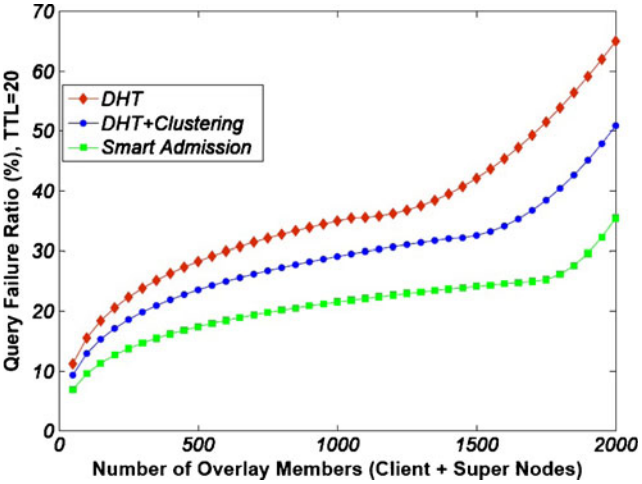


Fig. 13 Query failure ratio with TTL=20

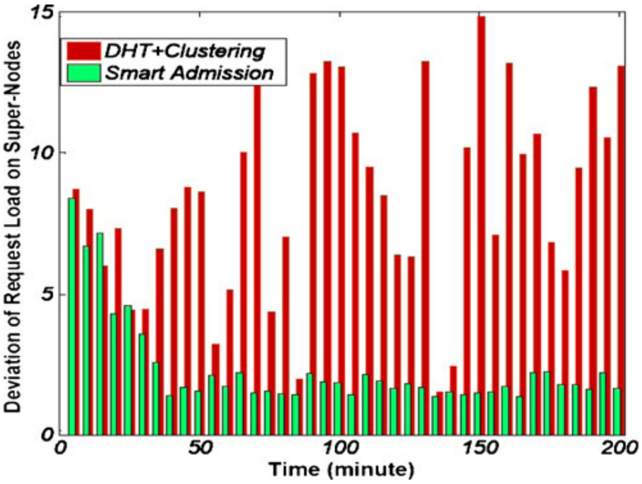


Fig. 14 Load balance factor (%)

If we consider that N_t is the number of super-nodes in the system at the time t, this factor is calculated as

$$100 \times \frac{1}{T}\sum_{t=t_0}^{t=t_0+T}\left(\frac{1}{N_t}\sqrt{\sum\left(\lambda_{\text{in}}^{i}-\bar{\lambda}_{\text{in}}\right)^{2}}\right). \tag{22}$$

For an observation from $t_0 = 0$ and for $T = 200$ min, with our admission strategies, we discover that the standard deviation of the load on the super-nodes is limited to around 2%. This means that all super-nodes have almost the same hit rates (i.e., uniform load distribution). Finally, we compare the number of admitted super-nodes for the two approaches of "DHT+Clustering" and "Smart Admission" in Fig. 15.

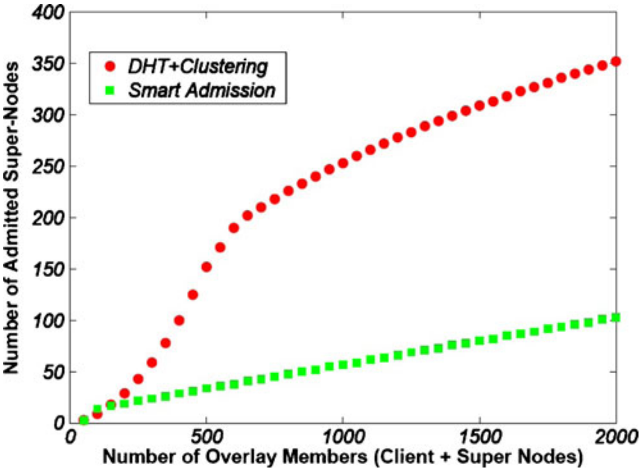


Fig. 15 Number of admitted super-nodes

7 Implementation and experimental results

We have developed a P2P platform running on wireless ad-hoc networks called SCOPE [22]. SCOPE is developed to provide a distributed infrastructure with unified API that enables the development of various P2P services on wireless ad-hoc networks. SCOPE follows a hierarchical P2P system architecture. Hence, we have developed the software for two kinds of nodes: super-nodes and client-nodes. *Super-nodes* organize the DHT overlay and provide the SIP proxy services.

For DHT, our service framework is based on Bamboo open-source DHT [3] which provides Pastry [33] DHT services. Bamboo with its public name OpenDHT [30] has been deployed on PlanetLab [26]. Bamboo has been developed to handle high churn which means the continuous process of node arrival and departure [31]. It also uses proximity neighbor selection [33]. Super-nodes are also responsible for call routing. In our platform, they provide distributed SIP Proxy/Registrar and form a communication service overlay. Finally, we have developed our admission strategies and the required module that enables each super-node to assess its load in terms of request rate and processing delay. Figure 16 shows the simplified software architecture of our super-nodes.

For *client-nodes*, we have developed a client agent called "*What's UP*" [23]. Besides P2P IP Telephony, What's UP provides all the services of social networking, including instant messaging, photo sharing, recommendation system and community management. All the requests that a user submits to the DHT overlay via the What's Up client agent will be transformed into Put/Get messages. This includes all activities, such as, creating user profile, sending messages, starting a call and searching for content. It is important to note that a super-node can also be a client-node.

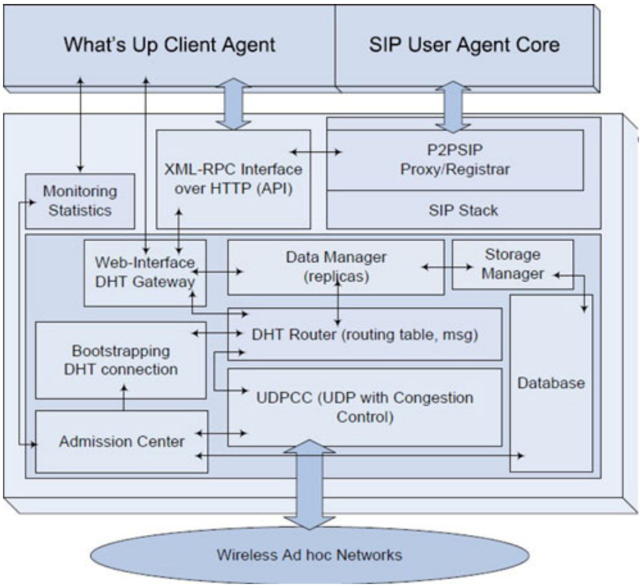


Fig. 16 Super node software architecture: DHT API in XML-RPC provides put/get and remove *(key,value)* pair. Upon receiving a request, modules DataManager and DHT router are called to route (*key,value*) pair to the correct super-node. Monitoring module gathers the call statistics and transfer them to the admission center for triggering new super-node admission process

7.1 Test-bed

We set up our platform with nine laptops connected to one another using IEEE 802.11g wireless technology. The laptops form a 3-by-3 grid network with 6 m of distance between two adjacent nodes. To provide the condition for multi-hop routing in our grid network, we have configured each laptop to see only its adjacent laptops in order to overcome the physical space limitation that results in all the nodes being able to see all the other nodes. We use the OLSR open source as our wireless ad-hoc routing protocol [9].

Each laptop is able to support up to eight virtual machines, on which we have installed our super-node software package. With this configuration, we are able to increase the number of super-nodes from 9 to 72. On each laptop, we are able to run up to 200 active clients. The clients register their identifiers with the corresponding super-node (virtual machine) through DHT rules. Moreover, they are local clients of the virtual machine on which they become active. Each active client sends a call setup request every second. In our experiment, we have limited the processing capacity of each virtual machine to 20 concurrent calls.

Due to the limitations in real experiments, we are unable to test all the features of our super-node admission strategies. We created the environment to test the impact of our admission strategies for *over-populated clusters* and *high hit rate*, and carried out the measurements to verify the impact of these strategies on session setup delay.

7.2 Test scenario and experimental results

In the initial phase, only one virtual machine on each laptop is active and serves as super-node. The overlay is therefore formed with nine super-nodes. In addition, five clients on each laptop are active.

(i) *Over-Populated Cluster Scenario*

In order to test the efficiency of our admission strategy for over-populated clusters, we set up the following scenario. We randomly select a laptop (i.e., a virtual

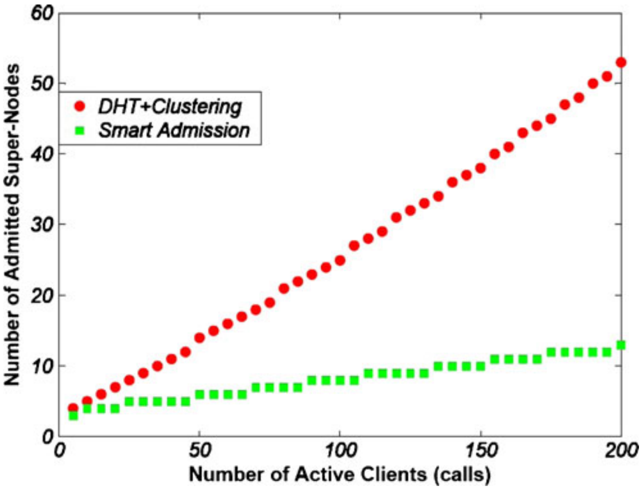


Fig. 17 Over-populated cluster: admitted SNs

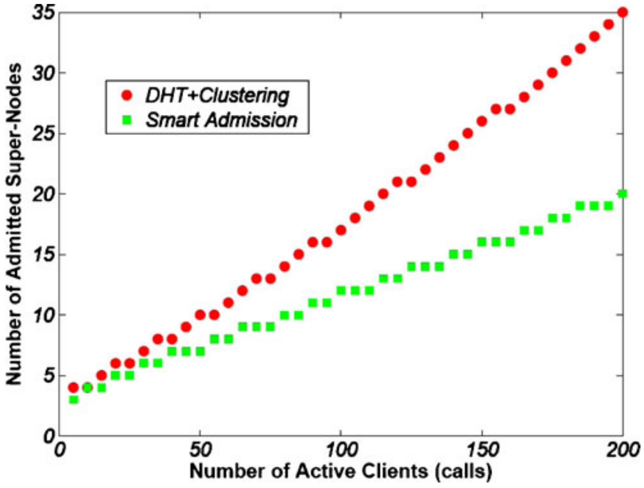


Fig. 19 Average load: admitted SNs

machine) and increment the number of its active clients to the level that the serving active super-node (virtual machine) become saturated (i.e., 20 clients). In the system with no admission strategy, when a super-node becomes saturated, new super-nodes (among the non-active virtual machines) will be randomly selected and added one by one to the overlay system until the problem disappears.

On the other hand, with our admission strategy, we expect that only one super-node (which is in fact a virtual machine on the same laptop) will be added to the overlay system to alleviate the load on the saturated super-node. This is the logical expectation from our admission strategy for the case of over-populated clusters since the virtual machines of the same laptop are physically in the same location (i.e., there is no closer node in the system). We repeated this scenario 100 times and averaged the results.

Figures 17 and 20 show the experimental results for the number of admitted super-nodes and session setup delay, respectively. Based on our assumption that each super-node can handle a maximum of 20 concurrent call requests, the ideal lower bound for the required number of super-nodes, for 200 active clients, is $200/20 = 10$. For this scenario, the smart admission strategy is very close to the lower bound.

(ii) *High Hit Rate Scenario*

To examine the capability of our admission strategy to alleviate the load on the super-nodes with high hit-rates, we use the following scenario. We increase the number of clients whose IDs fall within the cover range of a specified super-node. This increases the number of

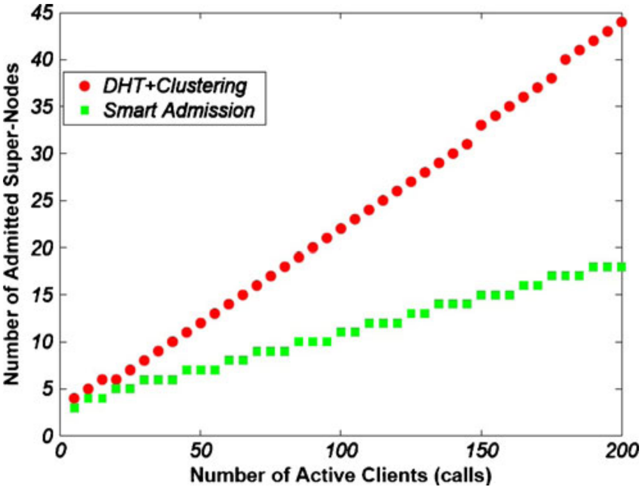


Fig. 18 High hit rate: admitted SNs

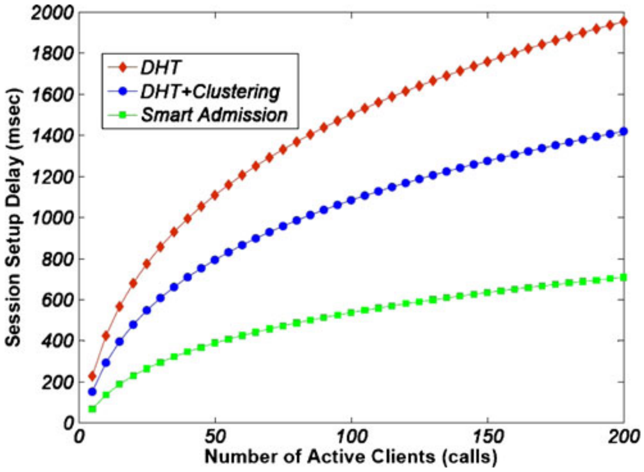


Fig. 20 Over-populated cluster: session setup delay

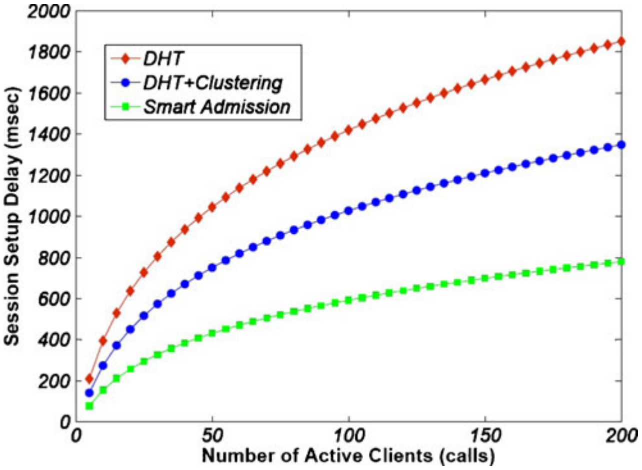


Fig. 21 High hit rate: session setup delay

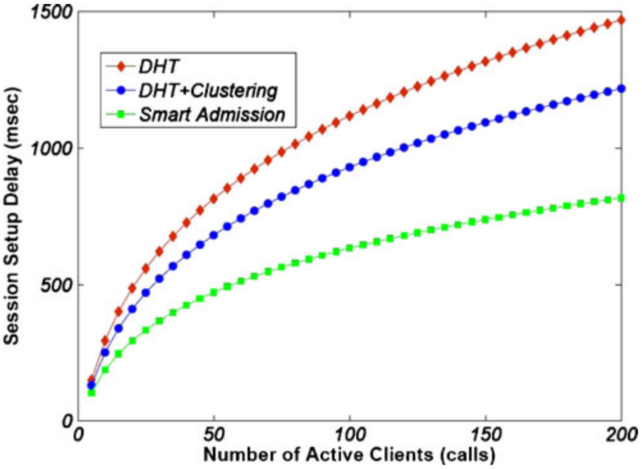


Fig. 22 Average load: session setup delay

global clients of that specified super-node and increases its hit rate. Figures 18 and 21 show the results for this test scenario.

(iii) *Average Load Scenario*

Lastly, we also tested the scenario where we add new clients randomly to the overlay system. This scenario examines the performance of the overlay system under average loads and Figs. 19, 20, 21 and 22 show the results of this test.

8 Related work

In [25], overlays are classified into three categories in terms of topology formation approaches: (i) network-oblivious, (ii) proximity-aware and (iii) network-aware. Structured P2P overlays using DHT algorithms such as Chord [36] and CAN [28] fall in the first category where each peer selects its neighbors just based on the information in the application-layer's logical (and not physical) space. On the other hand, the DHTs such as Pastry [33] and Tapestry [40] are classified as proximity-aware Overlays. They exploit the knowledge of the distances between the nodes to include the physically closed nodes in the routing table of a peer in the overlay.

The locality principle is the main intuition behind the strategies of the network-aware overlay construction category. This locality principle ensures that the traffic with only local relevance stays local. This category includes work such as [37] which focuses on creating an overlay topology which is based purely on physical node distance. A topological aware overlay construction approach, for large scale internet applications, introduces a binning strategy for overlay construction [29]. Nodes partition themselves into bins with each bin containing the nodes that are relatively close to one another. This binning strategy is applied to CAN [28] to form a network-topology-aware overlay. This paper introduces scalable and simple schemes to provide coarse-grained proximity information (instead of precise and costly information) to the overlay nodes. The mainstream research on network-aware overlays aims to show that adding the network topology awareness improves the performance. This improvement could be in load distribution, traffic management and lookup success rate. However, the challenges like super-node selection and the proportion of overlay resources (number of super-nodes) to the load have not been studied.

The importance of super-node selection in P2P overlay formation has been considered in the literature for different applications [2, 19, 39]. A gossip-based protocol [19] optimizes the delay between clients and their corresponding super-nodes for VoIP and multi-player gaming applications by building a proximity-aware super-node overlay. In [39], the performance of hierarchical super-node overlays under different configurations is investigated. The configuration parameters include super-node neighbor number (out-degree), super-node redundancy and request TTL (time to live). This work, which targets a file-sharing application, derives the optimum configuration for the super-node overlay based on the individual and global loads. In [2], another aspect of super-node selection for file-sharing and streaming application is discussed. Each client selects a set of super-nodes that are supposed to satisfy its requirements and criteria (e.g., delay, cost, etc.). The authors have developed theoretical methodologies for these types of super-node selection schemes.

All the optimizations performed in these research efforts are in the terms of the requirements of large-scale overlays in the Internet. Therefore, the network challenges are different from what exist in wireless ad-hoc networks. Moreover, most of the works in the field of super-node selection consider the constraints for file-sharing applications. The main constraints in these applications are the traffic load and bandwidth limitation. P2P VoIP application motivates the work in [19] which tries to form the overlay in a way to reduce the delay between clients and their host super-node. The delay a request experiences between the client and its host super-node is just one part of the whole session-setup-delay. Authors in [24] discuss some strategies for positioning Super Nodes. Lastly, there are also proposals to use DHT as the network layer routing in wireless ad-hoc networks [8]. In [8], the authors propose the Virtual Ring Routing (VRR) algorithm for network layer routing and they apply it on an IEEE 802.11a wireless ad-hoc network. This work emphasizes on the fact that a node avoids the need to flood a route request by using DHT. Our work that is presented in this paper is distinctly different from the work discussed previously since we are focusing on creating the service overlay on top of wireless ad-hoc networks and we are not dealing with network layer routing algorithms.

9 Conclusion and future work

In this paper, a new strategy for selecting and admitting new super-nodes is presented to form structured P2P IP Telephony overlays for wireless ad-hoc networks. We discussed the important criteria for selection of super-nodes in wireless ad-hoc network, which are connectivity, geographical position and position in the overlay. We considered *session setup delay* as the main performance criteria which active super-nodes use to assess the performance of the overlay. Our system admits new super-nodes into the overlay whenever the session setup delay exceeds the standard threshold and causes the system performance to degrade.

We discussed the three main reasons that cause performance degradation of the system: (1) high number of local client, (2) high hit rate (DHT request load) and (3) long distance between clients and super-nodes. We defined the strategies that enable the super-nodes in the system to detect what is the cause of the performance degradation and this is achieved without the need for a central monitoring system. In the proposed system, the affected super-nodes will trigger the appropriate strategy to admit the new super-nodes. We defined a specific admission strategy for each of the three causes of performance degradation. The goal of the new scheme is to maintain the quality of system performance while admitting the minimum number of new super-nodes. Based on the provided analytical model, we proved the efficiency of the new admission strategies via simulations. Moreover, real experimental results obtained from the developed platform are presented as a proof of the concept.

In our study, we have considered the case of a campus network where wireless ad hoc networks of nomadic users utilize IP Telephony to communicate. We have assumed that nodes engaging in IP Telephony while on the move are rare in an environment where the dominant wireless technology is WiFi or IEEE802.11 because of handover latency issues. However, mobile IP Telephony is becoming more prevalent and we are going to study the effects of node mobility on the performance of the system. Concurrently, we will also be enhancing the cluster splitting algorithm by considering other factors like node density and traffic patterns to achieve better performance.

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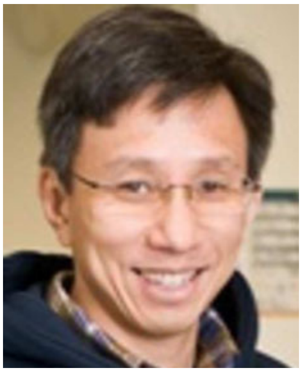

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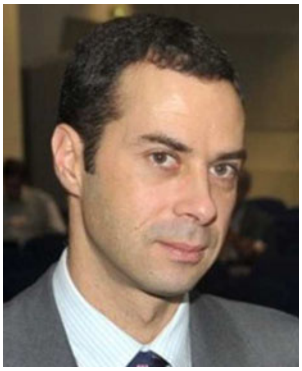

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