Accelerating Call Route Query of Multi-domain SIP System via P2P

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Abstract. Large IP Telephone system may contain multiple domains, and the delay of DNS query across domains may be large. In this paper, the efficiency of call routing in multi-domain SIP system is improved by employing P2P technology. The P2P-SIP system architecture is designed and a no-worse-than-two-hop query algorithm is presented. The formulas for evaluating the performance of the proposed solution are derived. The efficiency of call routing process can be effectively enhanced, and the instability in the system operation can be avoided. The proposed solution can be applied to national wide enterprises or organizations.

Introduction

The call control protocol SIP (Session Initial Protocol)[1] has only two kinds of entities: SIP user agents and SIP servers. The SIP servers can be a register server, a proxy server, and a redirection server. The location server is used to store and retrieve the user's potential location information, and it is not a formal functional entity of the SIP system. UA (Agent User) represents a terminal system, which includes the client (UAC) and server (UAS).

The register server (Registrar) is used to process the registration request for the SIP UA. It records the URI (Uniform Resource Identifier) and IP address of the SIP UA. The register server is usually located in the same physical entity with the proxy server or the redirection server. The proxy server is a functional entity that is both a client sending request and a server receiving a request. SIP request can be forwarded to the next hop server through a number of proxy servers, and the proxy servers can handle request or forward the request to the next hop server. If necessary, the proxy servers will interpret the request message and even override the request message. The next hop server may be another proxy server, or the end user agent. The redirection server is used to determine the address of the next hop server that will process the call in the SIP domain, and respond the client with the address of the next hop server.

Simple SIP system has only one management domain, while complex SIP system can have multiple management domains. They are called single-domain system and multi-domain system respectively. Each domain in multi-domain system has independent SIP domain name and servers. The servers can process the call request and connect the caller and collee intra the domain. The processing of inter domain call request is completed by the cooperation of servers in various domains.

Single domain SIP system is not suitable for large IP phone system. This is because the number of users is huge, performance bottleneck is likely to happen in centralized management system, and there is single point of failure. Multi-domain SIP system can be a good solution to the above problems. E.g., in multi-domain SIP system， each sip domain can be isolated securely. When a certain domain is infected with viruses or attacked, that domain can be temporarily isolated, thus quickly solve the security problem. For the partition of SIP domains, they can be divided according to the user distribution in geographical scope, so as to comply with the current communication network division principle, and making the interconnection and integration of different networks easy.

When IP call is being processed in SIP system, ENUM (Number Mapping tElephone) technology is used to translate the phone number into the user's domain name, and then the domain name service system (DNS) is used to obtain the corresponding user domain server address [2]. In the multi domain SIP system, the DNS server in each domain stores the SIP server address in its domain. When the cross domain call happens, information search among multiple DNS servers is needed, and the delay may be large. So the P2P technology can be used to improve the query efficiency.

At present, the SIP phone systems (soft switching or IMS) implemented by some traditional telecom manufacturers and operators are capable to support the hierarchical multi-domain settings. However, as to the structure of the multi-domain system, query efficiency and other issues, there are little quantitative analysis and systematic researches.

The academia has studied application of P2P technology in SIP systems, and made a number of technical solutions in improving call queries and call handling efficiency, speed and reliability [3]. IETF also gives the relevant standards [4]. But multi-domain problem has not yet directly involved in these studies and standards.

Ref [5] designed a locality-aware P2P-SIP system with query hops complexity being O(logN), but its research on multi-domain is only with two-tier structure, and more levels of SIP system is not considered. Ref [6] designed a media communication mechanisms based P2P and SIP combining. It used a hierarchical structure for the P2P-SIP overlay network, and proposed related algorithm with O(logN) query hops complexity. But it stayed on an abstract fully distributed model, and did not analyze the physical distribution of actual SIP multi-domain systems, the hierarchy and the domain name resolution mechanism. Its applicability in practical SIP multi-domain system is not high.

In this paper, according to the application requirements and organization distribution characteristics of the large IP telephone system, the hierarchical structure of multi-domain SIP system is presented. The DNS based call routing mechanism and its shortages are analyzed. The multi-domain SIP system is improved by the use of P2P technology. The call route query algorithm no worse than two hops is given. The algorithm is analyzed quantitatively by the typical user distribution and the traffic statistical model.

Architecture of Multi-Domain SIP System

Concept of Abstract Service Node in SIP System

The multi-domain SIP system consists of multiple single domain SIP systems. Each SIP single domain system has a variety of SIP servers (proxy server, location server, etc.). We use a abstract service node to represent these servers in the single domain SIP system, and the network of service nodes constitutes the core of the SIP system. Service nodes can be composed of multiple functional entities, for example, one service node in a certain domain can be composed of SIP proxy server and location server.

Hierarchical Structure of Multi-Domain SIP System

According to the demand of large IP telephone applications, the SIP multi-domain system can be divided into several levels, and each level can have multiple domains. Each domain has service nodes such as SIP servers. Users within the domain are managed by domain SIP server management, with username expressed in form of URI (Uniform Resource Identifier) telephone numbers: user-name-prefix@multi-level-domain-name. And the query and localization of one user is refined level by level by the collaboration of SIP redirect server, location server and proxy server based on domain name suffix.

As to the national-scale multi-domain SIP system in China, it can be designed as the three level structure, as shown in Fig. 1.

At the top plain, the 32 provincial-level regions are mapped in the first level, the top level is composed of 32 domains. At the middle plain, the cities of one province (e.g. Jiangsu province) is mapped in the second level, they are the sub domains of a first level domain in accordance with administration relationship. Third level (i.e. the bottom plain) can be set up where appropriate. For a second level domain with large number of users, the third level domain can be set up to serve the counties, district, and institutions, etc. Otherwise, if the scope of a second level domain is small, no third level domain would be needed.

According to the hierarchy of these levels, the structure of the domain name can be obtained. For example, the domain names of Jiangsu, Nanjing, Jiangning are js, nj.js and jn.nj.js.

Call routing for multi domain SIP system based on DNS query

Via summary of IETF SIP system and IMS call routing process [7], it can be known that there are two kinds of routing: the direct route (SIP server servicing the caller directly forwarding the call request to the called party) and relay routing (forwarding the call request through SIP relay server to the called party). Direct routing performance is better than that of relay routing, but the SIP server needs to configure the SIP server information in every domain and the management is complex. Relay routing method is simple in management, but the call delay is large. In case of large SIP system, DNS based on relay routing can be used and relay number need be limited to reduce call delay.

The DNS server in each domain stores the address of the SIP server in its domain. For the cross domain call, the caller domain DNS may send queries to the multilevel DNS servers and the time delay may be larger. E.g., when a user in Beijing call another user in Jiangning District, Nanjing, Jiangsu, the Beijing SIP server send query to Beijing domain DNS (DNS.bj), and DNS.bj query about IP address of SIP server responsible for the called user in Jiangsu domain DNS server (DNS.js). DNS.js again send query to Nanjing domain DNS server (DNS.nj.js), DNS.nj.js to Jiang Ning DNS server (DNS.jn.nj.js) in the query forwarding chain. Finally the called domain of SIP server IP address is obtained. Then Beijing SIP server forwards the call request to the Nanjing Jiangning SIP server.

In order to reduce the delay of searching information among multilevel DNS servers, the domain name / address information cache can be used. A first-level DNS caches the addresses of SIP servers in all other first-level domains and the second-level domains directly managed by it, in order to achieve high speed forwarding of the call requests. Each second-level domain DNS can cache the DNS address of its superior domain SIP server and adjacent second-level domain SIP server. In addition, the address information cache can also be used in the area of a relatively large call flow.

The traditional DNS system has the shortcoming of non-structured network, poor scalability and query routing only in accordance with the domains level by level. When the DNS servers in the same level is excessive, the routing hop number and maintenance costs can not be balanced, and the query efficiency for large-scale multi-domain SIP system is low. Although the use of DNS cache technology can reduce the searching time, but there are still shortcomings in cache data consistency and cache efficiency. Especially, the cache data consistency maintenance requires a certain bandwidth overhead, and when the DNS network is in the presence of link instability, the cache data update in real time is influenced. Therefore, with the characteristics of P2P technology in distributed data synchronization and search, the telephone number resolution and query performance can be effectively improved.

Distributed storage and query in SIP system based on P2P

In multi-domain SIP system, the delay to locate user is an important metric to measure the performance of the system. Theoretically, one hop lookup can ensure the fastest user locating, but the continue update of routing tables require an overhead of certain bandwidth occupation. Especially when the system is in a relatively unstable state, routing table updates in one hop lookup system may make the system produce jitter. And a large number of statistics show that the traffic between the nodes decreases with the distance (geographical distance, administrative distance, etc.) increase [7]. Therefore, we design a no worse than two hops lookup mechanism. In the proposed system, maintenance traffic bandwidth is low, and the instability in system operation is avoided. Meanwhile, the query for locating a user can be finished either in one hop with high probability, or in two hops with low probability.

Description of no worse than two hops query

Definition 1: Node, is an entity with all its software and hardware resources for the purpose to provide services in the network. Network head, is the node that has the right to maintain a network, denoted as *s*. Member node in a network is the node that does not have the network maintenance right.

Definition 2: the level-i subnet, is the subnetwork maintained by an level-i head. The level-i subnet head is a node in the level-(i-1) subnet, with i>1. A top-level subnet, i.e. the level-1 subnet, is the subnet composed of the top nodes in the network; it is in the top level of the network topology, and there is no network head.

Definition 3: In the level-i subnet, for nodes q and p with the IDs in P2P system being I_q and I_P if $I_q > I_P$, and q is the first valid node after p, then q is called the successor node of p, and is denoted by $successor(p)$.

The hierarchical topology of the multi-domain SIP system based on P2P is shown in Fig.1.

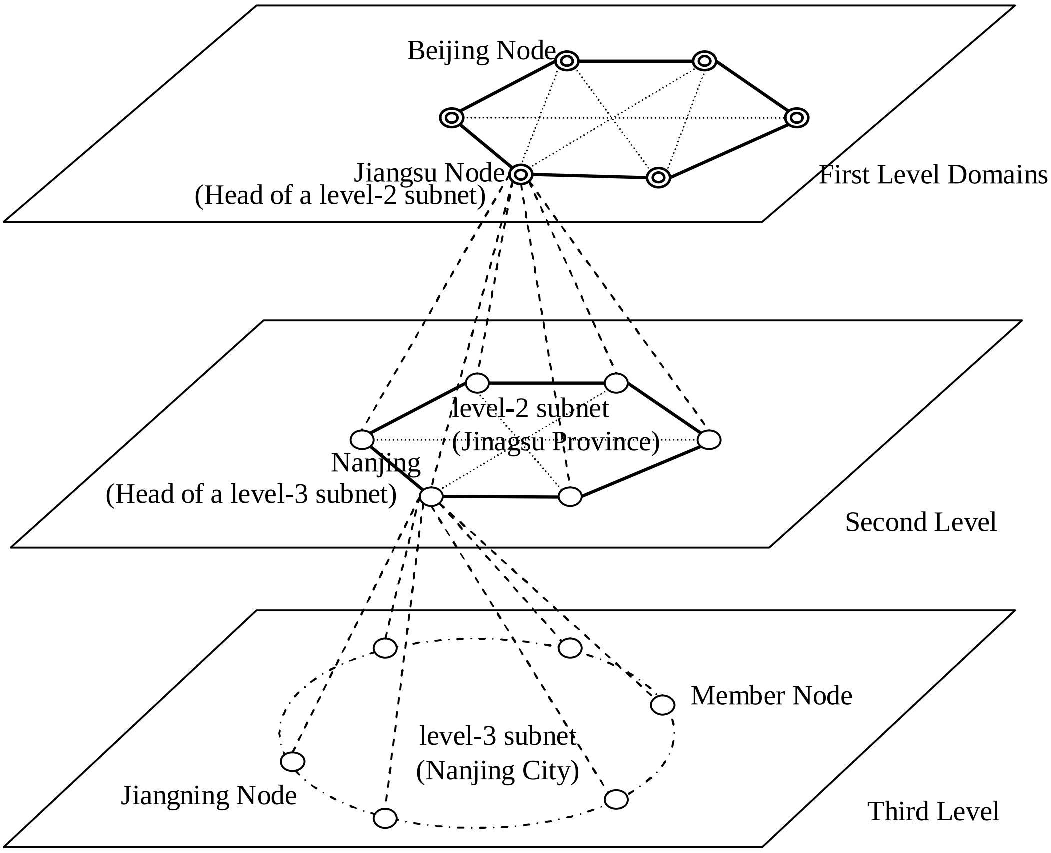

Fig. 1 Hierarchical Topology of Multi-domain SIP System Based on P2P

The hierarchical topology design can meet the requirements of the error isolation and security, provide efficient cache and bandwidth utilization and realize the hierarchical storage and access control. Furthermore, it can effectively alleviate the maintenance overhead brought about by the increase of system capacity. Therefore, in the no worse than two hop lookup system, we adopt a hierarchical topology approach. The whole network is composed of several sub networks, which form a hierarchical relationship among them. The upper subnet node being the lower subnet head is responsible for maintaining lower subnet size and membership changes. The topological organization within a subnet can be more flexible. Full-mesh and chord ring [8] etc. can be used.

In order to ensure any node can find the destination node in no more than two hops, we only need to ensure that the routing table of any node in the subnet of a certain level includes information of all the nodes in that subnet, and all the nodes in the lower level subnets directly or indirectly managed by the subnet.

When a node in the subnet of a certain level joins or leaves the network (later we will denote node joining or leaving the network as an **event**), the successor node of the event generating node detect the event firstly. Then the successor node informs the event to the subnet head, and triggers the event distribution in the subnet. The subnet head sends the event to the managing nodes in the upper subnet.

In General, when a node p in the i_k-th subnet of the level-k (denoted by $S_{i_1 i_2 \ldots i_k}$) produces an event, its subsequent node $successor(p)$ first detects this event, then sends the event notification message to the network head $s_{i_1 i_2 \ldots i_k}$, and triggers the event propagation in the subnet $S_{i_1 i_2 \ldots i_k}$. The Network head $s_{i_1 i_2 \ldots i_k}$ will be periodically send the event notification message to the upper level subnet head $s_s = \{s_{i_1 i_2 \ldots i_j} \mid \mathrm{j}=1,2,...,\mathrm{k}-1\}$ it belonged.

Sturture of Multi-Domain SIP System Based on P2P

The service nodes in the multi-domain SIP system can be connected as a P2P network, which can be called multi-domain P2P-SIP network. The multi-domain P2P-SIP network are designed with a structure of three levels: the top level subnet (primary domain), middle subnet (second level domain) and the bottom subnet (third level domain). The service nodes, as the head of certain subnets, stores information about its subnet nodes, and maintains node number/size and member change of the lower-level subnet, and so on. In the top level, nodes are connected by the way of Full-Mesh and backup the information with each other. So do nodes in the middle level. In the bottom level, the number of nodes may be great, so the nodes can be connected as a Chord ring. More details about the mechanism of hierarchical-structured P2P model can be found in [11].

Mobility Management

The user mobility can be supported by SIP system. When an end user moves into a new location in the IP network, the user's UA will register its new IP address in its home Registrar Server. Then the new IP address can be queried by other UA via P2P service.

The SIP service node can also be supported by the SIP system based on P2P. When a SIP service node change its location and IP address, its ID based on its SIP domain does not change in P2P-SIP system. So the logical structure of the P2P-SIP system remains the same. Its new IP address is propagated in the P2P system quickly, and can be queried by other SIP service node.

Performance of Multi-domain SIP System based on P2P

Time Bound of Event Propagation

It takes some time for an event to be delivered to all nodes. This leads to information inconsistence and node query error. We use *f* to represent the user query error probability. In order to meet the constrain of system error ratio, the propagation time of the event (the event from generation to transmission to the last node in the desired available network) should have an upper limit, denoted by t_{tot}. According to the above analysis, we get the formula (1),

$$t_{tot} \le \frac{f*n}{r}. \tag{1}$$

where *n* is the system node number, *r* represents the event generating rate.

In accordance with the foregoing analysis of events update propagation mechanism, we have $t_{tot} = t_{dectect} + \max(t_{l3},\ t_{l2},\ t_{up} + t_{l1})$, where $t_{\mathrm{det}ect}$ is the time it takes from event generation to event detection, t_{l3} is event propagation time throughout the Chord ring of the bottom level subnet.

The bottom subnet head, which is a node in middle level subnet, delivers event to other nodes in middle level subnet in time interval t_{l2}, and reports event to the head of the middle level subnet in time interval t_{up}. Similarly, the middle subnet head, which is a node in top level subnet, delivers event to other nodes in top level subnet in time interval t_{l1}, where $t_{up} < t_{l1}$.

Maintenance Traffic Analysis

Now we analyze the required system maintenance traffic to ensure the worst two-hop query. The system event message consists of the message payload (m-byte) for event descriptions and header (v bytes). There are several main message types in the system:

(1) heartbeat message. Due to the dynamic characteristics of the nodes, when a node joins the network, it will periodically send heartbeat information to its successor nodes. Taking into account that the nodes in the middle and top level subnets are relatively stable, heart beating cycle can be set relatively long, the heartbeat message traffics in the middle and top level subnets can be neglected.

(2) event message propagating in the bottom subnets. We assume that event generations on the bottom subnet are evenly distributed in the subnet. At the bottom subnet, when the successor node detects an event in the incident node, it immediately sends a message to the subnet head. [9] proposed an incident detection and report algorithm for EDRA based on chord. The algorithm can complete event spread throughout a network with time complexity O(log(n)) (n being the node number), and has good load-balancing performance and low maintenance costs. In this paper, we use the EDRA algorithm for the spread of events on the bottom subnet.

(3) event message propagating in the middle subnets. When an event is detected in the bottom subnet, it is immediately reported to the subnet head in the middle level. The subnet head will packet it in a message and send the message to all other nodes within interval t_{l2} in the middle subnet. In addition, the subnet head needs collect the events and report to the middle subnet head in the top level every t_{up} seconds periodically.

(4) event message propagating in the top subnets. When an event is by the middle subnet head in the top subnet, it will packet it in a message and send the message to all other nodes within interval t_{l1} in the top subnet.

Bandwidth Requirement for a Typical Case

We assume the subnets on the same level have the same size. We use k_1, k_2 and k_3 to represent the sizes of subnets on the top, middle and bottom levels. Then we have $n = k_3 * k_2 * k_1 + k_2 * k_1 + k_1$. The formulas for evaluating the traffics of the head of subnet on the top, middle and bottom levels are shown in table 1, where S_{avg} is the average online time of the service node and θ is the event relay delay in a bottom service node.

Talbe 1 Evaluating the Traffics from different Sources in the proposed System

Source	Uplink Traffic	Downlink Traffic
member node	$\frac{2*(m+v)}{S_{avg}} + v + (2*N_{msgs}*v + r*m*q)/q$	$\frac{2*v}{S_{avg}} + v + (2*N_{msgs}*v + r*m*q)/q$
bottom subnet head	$\frac{2*k_3*(k_2*m+v)}{S_{avg}} + \frac{2v}{t_{l2}}*(k_2-1) + \frac{v}{t_{up}}$	$\frac{2*k_3*(k_2*m+v)}{S_{avg}} + \frac{2v}{t_{l2}}*(k_2-1) + \frac{v}{t_{up}}$
middle subnet head	$\frac{v}{t_{up}}*k_2 + (\frac{2*k_2*k_3*m}{S_{avg}} + \frac{2v}{t_{l1}})*(k_1-1)$	$\frac{2*k_1*k_2*k_3*m}{S_{avg}} + \frac{v}{t_{up}}*k_2 + \frac{2v}{t_{l1}}*(k_1-1)$

As an typical case, the configuration parameters can be set as follows: f=1%, n=10^4~10^5. The heartbeat information is 1 second, and taking into account the message loss we assume $t_{detect} = 2s$. In the experiment, we take $k_1 = 31, k_2 = 10$, m=20 bytes, and v=40 bytes. Form Equation (1), it has $t_{tot} \le 50s$. Because of $t_{tot} = t_{dectect} + \max(t_{l3},\ t_{l2},\ t_{up} + t_{l1})$, so $t_{l3},\ t_{l2} \le 48s$. Constrained by the requirements of rational message overhead and $t_{up} < t_{l1}$, we set $t_{up} = 5s$. According to ref [6], we have $q = \frac{2*t_{l3}}{\lceil \log_2 k_3 \rceil} \le 9.6s$.

For a extreme case with n=10^5, the uplink bandwidth requirements for SIP service nodes in different level with their average online time being 24 hours are less than 1kbps. With the increase of the average online time, event generating ratio r in the system decreases, resulting in bandwidth requirements decreased.

Conclusion

In this paper, the hierarchal structure and call routing issues of multi-domain SIP system are studied, the shortcoming of call routing mechanism based on DNS is analyzed, and the efficiency of call routing is improved via using P2P technology to realize the no worse than two hops routing queries. The proposed multi-domain SIP system solution is suitable for large IP telephone systems, can be applied to national departments and enterprises.

Because of its distributed, self-organizing and scalable features, the service capabilities of P2P network increase with the network size. P2P can effectively spread the load to ensure that the voice quality is not lower than the traditional telephone network. This article mainly uses P2P to improve SIP call routing optimization problem. As to using P2P features in distributed processing and centerlessness to solve problems such as the single point of failure and performance bottlenecks of the existing SIP systems client/server architecture, is the direction for further research.

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