How well does JXTA fit Peer-to-Peer SIP?

Holger Schmidt, Burcin Aksoy, Franz J. Hauck
Institute of Distributed Systems
Ulm University
Germany
Email: {holger.schmidt,franz.hauck}@uni-ulm.de

Abstract—More and more people use voice over IP instead of traditional phone networks. The Session Initiation Protocol (SIP), a protocol for session management in general, becomes the standard protocol in use. However, SIP relies on central servers. Thus, maintaining the infrastructure is costly. This results in the trend to transfer peer-to-peer (P2P) concepts to SIP to overcome administration efforts due to self-organisation of the participants. In this paper, we propose a novel JXTA-based P2P architecture for SIP signalling. JXTA is an open peer-to-peer platform, which provides basic infrastructure services for building P2P applications. Our concept does not need central entities for maintaining the SIP traffic, while maintaining compatibility with standard SIP. Additionally, we seamlessly added service location capabilities into our infrastructure to support service discovery in dynamic environments. We evaluated our concept in comparison to standard SIP within three scenarios: local area network, point-to-point network and wide area network. The results show that our approach compares well in terms of response time but results in more traffic for maintaining the peer-to-peer overlay.

I. INTRODUCTION

Voice over IP (VoIP) becomes more and more accepted and the Session Initiation Protocol (SIP [1]) has been adopted as the standard for signalling, e.g., for session management in the 3rd Generation Partnership Project (3GPP [2]). SIP is an IETF protocol for session management in general, which is based on a client-server infrastructure. End-terminals (user agents) are acting as clients, while registrar servers store the clients’ contact information and proxy servers are responsible for forwarding messages between end-terminals.

Using traditional SIP infrastructure results in a set of administrative efforts and potential single points of failure within the network (proxy and registrar servers). Skype started the trend to use peer-to-peer techniques (P2P) for VoIP [3] to overcome these issues. Instead of SIP, Skype uses proprietary protocols for session management. However, providing interoperability requires standard protocols. This leads to efforts to integrate P2P mechanisms into the SIP protocol.

There are different approaches to integrate P2P concepts into SIP, which have the key challenge of making centralised SIP proxies unnecessary. Either the SIP protocol is extended [4] or P2P mechanisms are integrated into the discovery process of local proxies or directly into the user agents [7]. In contrast to the protocol extension, proxy and user agent integration maintain standard-SIP-compliance, as only the discovery process is P2P-based and signalling is still performed according to standard SIP. An extension of the user agent requires implementation changes, whereas using a local extended proxy server keeps the user agent untouched.

The key contribution of this paper is the presentation of a novel generic P2P-SIP architecture on the basis of JXTA [5], an open and language-independent P2P framework. Unlike recent efforts, such as [4] and [6], our architecture is only based on standards and open architectures. JXTA allows the integration of arbitrary application layer P2P protocols, which is necessary for supporting different network topologies. For instance, nodes in structured (distributed-hash-table-based) P2P systems are characterised by processing higher maintenance traffic [7]. Thus, a configurability of the P2P routing mechanism can save resources, especially on mobile devices. Our system is standard-SIP-compliant and integrates service location. Instead of two stacks for service location and session management, only one integrated SIP stack is needed, which saves resources, e.g., on mobile devices. The performance results of our prototype show that it is an alternative to standard SIP in terms of response time. However, maintenance traffic is quite high, which may currently restrict the usage within mobile networks for cost reasons. We published early ideas on our JXTA-based concept [8], however, this paper focuses on the performance evaluation.

The paper is structured as follows. First, we present related work and then give basic information on SIP, P2P and JXTA in Section III. In Section IV, we sketch our novel JXTA-based architecture for SIP. Section V evaluates performance for different topologies: local area network, point-to-point network and wide area network. Last, in Section VI, we conclude this paper and sketch possible future work.

II. RELATED WORK

Using P2P techniques with SIP is a popular research area [9]. There is a lot of work on concepts, terminology, infrastructure, requirements and remaining burdens (e.g., allocation and protection of SIP names) [10]. In general, there are two possible approaches to integrate P2P mechanisms into SIP: either P2P is integrated into the SIP protocol [1] or user agents/proxies rely on a P2P-based location service [11]. In this work, we target the latter approach.

Bryan et al. [4] sketched a Chord-based P2P-SIP approach. They remove central SIP entities, but in contrast to our work, they demand an extended SIP protocol and specify hard-wired support for Chord. Our infrastructure builds on standard SIP...
and the standardised and open platform JXTA that allows the seamless integration of arbitrary routing mechanisms.

Singh and Schulzrinne [?] implemented a P2P proxy that provides an OpenDHT-based P2P infrastructure for registration and discovery and builds on a modular design. We provide a more generic architecture for the integration of P2P technology into SIP by building on standard SIP and JXTA, in which arbitrary routing mechanism can be integrated. This is important for mobile devices and for integrating additional services.

In contrast to all these approaches, we integrated P2P-based service location. Especially in dynamic networks, where service location is essential [12], decentralised techniques in general result in better scalability and fault tolerance among the participating entities.

III. BACKGROUND

A. Session Initiation Protocol

SIP is a protocol for general session management with focus on multimedia applications, e.g., VoIP. As specified [1], SIP builds on the client-server paradigm and defines several entities: user agents, registrars, location services and proxies. User agents (UA) are the end-terminals in the SIP network and register their contact address at a registrar server. The registrar stores contact data into a central location service (LOC), which acts as a kind of database for a particular domain. Proxy servers are responsible for forwarding messages from one UA to another (see Figure 1(a), messages 1, 2, 5). For user agents within the same domain, contact data for forwarding the message can be retrieved from the domain-specific location service (3, 4). For user agents outside the domain, the SRV record from the domain name system (DNS) is used to forward the message to a domain-specific proxy server (2).

B. JXTA

JXTA is a generic and open platform for P2P applications [5]. It tries to establish a common basis for P2P applications by specifying six protocols, because many P2P applications share fundamental concepts.

A JXTA network consists of a set of peers, which are organised in peer groups that restrict the propagation of messages to group members. JXTA specifies different types of peers, e.g., standard peers called edge peers and super peers called rendezvous peers. Each JXTA resource (e.g., peer) is uniquely identifiable (ID) and represented by an XML metadata structure called advertisement. Advertisements are published in the associated peer group for a specific lifetime.

To support the discovery process, rendezvous peers index edge peers’ advertisements. Searching for a specific resource is realised by searching for the associated advertisement. These advertisements are searched by broadcasting within the local network and by using the rendezvous peers’ index data (rendezvous peers build a ring topology). However, JXTA allows replacing its routing mechanism with a customised one.

IV. JXTA-BASED PEER-TO-PEER SIP

A. Design

As already mentioned, standard SIP is based on the client-server paradigm. However, such solutions lead to administrative efforts and single points of failure within the network. In general, P2P techniques provide possible solutions.

For supporting standard SIP devices, we developed a JXTA-based SIP location service (JXTA-LOC), which is used by standard SIP entities. This leads to SIP entities that externally behave according to standard SIP but internally use our JXTA-LOC (see Figure 1(b), messages 2, 3). SIP proxies relying on our JXTA-LOC are able to discover the target user agent in the JXTA network and to directly forward messages without further proxies (1, 4), which results in a reduced session establishment time. Therefore, registrars use the JXTA-LOC to store information about UAs in the JXTA network while maintaining standard SIP compliance. For an improved network usage, the proxy and the registrar should be located locally related to the UA. However, a direct integration into the UA is possible as well, which results in direct communication without proxies, but in the need for code changes at the UA.

Additionally, we integrated service management into our concept, i.e., service registration and discovery. This allows UAs to register services in the JXTA network and to search for them, which is especially useful within dynamic environments. For instance, UAs may search for services when connecting to a new access network, e.g., short message services.

Our approach allows coexistence with the standard SIP infrastructure. There, proxies which are responsible for a specific JXTA domain are able to act as gateways between the different network topologies, which is comparable to traditional telephone network gateways.
B. Implementation with Advertisements

For the implementation of the JXTA-LOC we transferred the concept of SIP domains to JXTA peer groups: UAs register their contact information in specific peer groups, which represents the corresponding SIP domain (peer group name is SIP domain with prefix ‘sip:’). Contact information as well as information about services offered are stored and discovered in the JXTA network as an extended advertisement [8]. This advertisement contains all needed information about UAs and their services: a Key represents the publicly addressable SIP-URI, Contacts contains the current actual contact addresses of the UA and Contents includes the description of services offered (here: printer service described using the service location protocol [13]). SIP registrations are published with provided service information as advertisements with a given time after which they expire. This time directly maps to a SIP registration expires header. Whenever SIP proxies built on our JXTA-LOC receive a SIP message, they extract the target’s SIP-URI from the SIP request, search for the advertisement with the corresponding Key-field and if successful, forward the message on basis of the Contacts-field. Discovered advertisements are stored in a local cache for future use. For an integration of service location into standard SIP, we use the SIP OPTIONS message [14]. Service requests are embedded as SIP attachment and searched by the proxy on the basis of the Contents-field if supported (there has to be a type-specific handler, which is able to interpret the contents).

C. Extensions

Our architecture allows several extensions for future applications. The most important extension point is provided by JXTA enabling a replacement of the standard routing mechanism. This enables an application-specific customisation, e.g., using unstructured P2P flooding mechanisms in mobile networks as structured P2P causes high maintenance traffic [7], which may result in high costs in such networks.

It is also possible to register location information with our infrastructure. This information can either be abstract (e.g., in office or at home) or concrete (e.g., GPS data) and is stored in the advertisements Content-field, which enables location-based services. This allows call-setup with an area, e.g., in order to call everybody at a specific location for emergency reasons. In this case, proxies would forward messages on the basis of the registered location attributes. Moreover, it is even possible to extend the advertisements to include further information about user agents without losing compatibility.

V. PERFORMANCE EVALUATION

In this section, we present a performance evaluation of our JXTA-based P2P-SIP prototype implementation. We implemented the needed entities, i.e. user agents and proxy servers using JAIN SIP 1.2 [15] and JXTA 2.4. The standard SIP proxy uses a simple Java Hashtable as data store.

A. Testbed Description

We evaluated our concept with ns-2 (http://www.isi.edu/nsnam/ns/), an open-source discrete event simulator supporting networks with various topologies. For gaining realistic results we used the emulation feature of ns-2, which imitates a given network configuration to the application. This allows considering timing behaviour, which is important for evaluating P2P applications. We run the emulated nodes with user mode Linux (UML), a virtual machine running Linux. Such an approach enables running the application to evaluate in a standard Linux environment (UML), in which network traffic is intercepted and injected in the ns-2 network simulator without any modifications to the application. For each simulated network node, an UML process has to be started. The emulation was performed on an AMD Athlon XP 2500 with 512 MB RAM.

B. Scenario Description

We designed three scenarios to evaluate our approach.

- **Local Area Network (LAN):** a small switched network with 15 machines, which are connected with 100Mbit/s data rate with 2 ms latency.
- **Point-to-Point Network (PTP):** 30 randomly meshed machines with an average of 0.5Mbit/s data rate with 10 ms latency.
- **Wide Area Network (WAN):** four LANs, which are connected by a router; 100Mbit/s connection within the LANs with 10 ms latency; 5Mbit/s connection between LANs with 50 ms latency.

In all scenarios we deployed one sender, which rotationally sends 319 byte SIP MESSAGE requests to three same receivers on specified nodes. In the P2P-SIP scenarios, a JXTA-based proxy/registrar server was deployed on the receiver and sender node, whereas in standard SIP scenarios proxy servers were located on other nodes within the networks. In the LAN and the PTP scenario the sender and receiver applications were deployed on random but distinct nodes, in the WAN scenario on random nodes but within different LANs.

C. Results

In this section, we present the results of evaluating our concept within the three scenarios of the previous section.

First, we evaluated the round trip time of sequential SIP MESSAGE requests and the number of SIP and JXTA packets within the network according to the simulation time in the LAN scenario (see Figure 2). It is obvious that the first requests in the P2P-SIP scenario take much longer than in the standard SIP case. This can be explained by the JXTA discovery process: when the SIP proxy receives the MESSAGE request, it has to query the JXTA network for the actual contact address of the target user agent. In the standard SIP scenario, this is done using DNS, which is apparently faster because of its central nature. After the third request (three receivers), the round-trip time of P2P SIP is lower than in standard SIP because the target address is already cached from earlier discovery within the proxy server. This saves one hop
because in P2P SIP there is no more need for the target’s proxy server (see Figure 1). The number of packets in P2P SIP is much higher because of discovery, which is based on local broadcast, and the JXTA maintenance traffic.

Secondly, we evaluated the round trip time of sequential SIP MESSAGE requests and the number of packets within the network according to the simulation time in the PTP scenario (see Figure 3). As in the previous scenario, the round-trip time of the first P2P-SIP requests is higher than of standard SIP, and lower from the fourth request on. This is again because of the JXTA discovery process and caching mechanism. The variance in round-trip time, especially for P2P SIP, is higher compared to the LAN scenario because of the randomly meshed topology and therefore not predefined routing of SIP messages within the PTP network. The number of packets in P2P SIP is again much higher compared to standard SIP. In comparison with the LAN scenario, the number of packets in P2P SIP is higher because of the randomly meshed nature of the PTP scenario.

Thirdly, we evaluated the round trip time of sequential SIP MESSAGE requests and the number of packets within the network according to the simulation time in the WAN scenario (see Figure 4). Again, the round-trip time of the first P2P-SIP messages takes longer than in standard SIP, but is lower from the fourth request on because of JXTA discovery and caching mechanisms. The variance is higher compared to the LAN scenario but lower compared to the PTP scenario because of the predefined routing paths with one particular router connecting the four LANs. The number of packets in P2P SIP is again much higher in comparison with standard SIP because of JXTA discovery and maintenance traffic. However, compared to PTP, the number of packets is lower because of the JXTA rendezvous peer concept, which reduces inter-LAN communication by introducing a super-peer structure.

In all scenarios the first requests take considerably longer. This can be improved by periodically searching for advertisements in the domain, which results in more addresses being cached. However, such an approach reduces the response time but increases the overall traffic. Another alternative would be the use of improved routing mechanisms.

As P2P SIP reduces administrative tasks, it is predestinated for already self-organising PTP networks, e.g., in (wireless) mesh networks. Thus, we measured the scalability in comparison to standard SIP in the PTP scenario. Figure 5 shows that the number of overall packets increases with the number of actively participating entities because of the JXTA discovery and maintenance traffic. This results in a relatively high data volume, which leads to high costs in networks that are accounted by data volume (e.g., mobile phone networks). However, as we build on standard JXTA it is possible to implement a different protocol for mobile networks.
P2P-SIP approach with JXTA provides good response time but results in more traffic for maintaining the peer-to-peer overlay in comparison to standard SIP. Additionally, we seamlessly added service management to our solution.

Requests are not forwarded via the target’s home-domain proxy anymore in our approach, which results in a reduction of session establishment time (see Figure 1(b)). This leads to a potential problem as home-domain proxies may be used for resource reservation purposes (e.g., in 3GPP). However, this issue can be solved by introducing a mechanism into our approach to always forward requests via the home-domain proxy of the target. Thus, at least proxies in the proxy-chain between the sender’s home-domain proxy and the target’s home-domain proxy can be skipped, which also leads to an improved time for session establishment.

In future work, we would like to broaden our evaluation with replacing the standard JXTA routing protocol with different P2P protocols, e.g., protocols, which are suitable for mobile networks. Additionally, we intend to do an evaluation with different SIP message sizes, as this may have an impact to the results [16]. Moreover, to gain more realistic results, an evaluation with more senders and receivers is still subject to future work. Due to the fact that we have only results for a limited real world scenario (see [8]), we would like to compare our results with an extended real world evaluation using PlanetLab [17]. Last but not least, an analytical model may complete our work.

VI. CONCLUSION AND FUTURE WORK

In this paper, we presented a novel P2P architecture for SIP signalling on the basis of the generic and open peer-to-peer platform JXTA. Thus, we supersede the need for central proxy servers, which results in a self-organising P2P network without the need of central servers. However, we still guarantee compatibility to standard SIP. We evaluated our concept within a broad range of scenarios: local area network, point-to-point network and wide area network. The results show that our

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