EMPLOYING PRESENCE AND PROFILES IN VOIP SERVICES

- Enriching User Experience

KARYN WEI-JU KHOO, HUI-NA CHUA
Malaysian Research Centre, Kuala Lumpur, Malaysia
weiju.khoo@bt.com, hui.chua@bt.com

AND

CHENG-SUAN LEE, KEE-LEONG TAN
Malaysian Research Centre, Kuala Lumpur, Malaysia
chengsuan.lee@bt.com, keeleong.tan@bt.com

Abstract. This paper explores how presence and profiles can enrich the user experience by providing relevancy and flexibility in setting up and forwarding calls. Our objective is to gain understanding of how technology in Presence and Profiles can be brought together to create compelling human communication VoIP applications that deliver enriched experience to end users. In this paper, we present our work in delivering enriched VoIP call services by taking presence and profiles information into considerations to assist user in handling call setting up and incoming call intelligently.

1. Introduction

While the adoption of VoIP (Voice over IP) continues to accelerate in the markets, the question still exists: "What happens when you try to contact someone and they are not there?" Unfortunately, the answer to this question lies outside of VoIP-specific technology. Traditionally, telephony's answer to this question was provided via two standard PBX (Private Branch eXchange) features: 1. voicemail, 2. hunt groups (call automatically forwarded along a predetermined path of phone numbers). However, neither of these features really addresses the problem of trying to assure a connection for a caller who needs to communicate real-time. One possible solution to this problem: Presence and Profiles.

Presence is dynamic information of a user, which is used to represent a particular user, share information and control services. Presence can be seen as a user’s status as perceived by others and others’ status as perceived by the user. Status may contain information such as personal and device status, location or context, device terminal capabilities, preferred contact method as well as services the user is willing to use to communicate with others [3GPP]. Profile contains all the data associated with a user, that includes the service related information [3GPP] such as incoming call handling information, service selection and area coverage, service subscriptions, and service parameters [ANSI, 1994]. Profiles can also consist of a description of a user access rights and the pattern of a user's activity that can be used to detect changes in the activity [ISO/IEC, 1998].

The concept of Presence and Profiles are well established but still highly and tightly bound to instant messaging (IM). While most will appreciate the evolution of VoIP products' inclusion of presence, for instance Skype [Skype], such offerings are still mostly vendor specific and do not have the scope of design that could fully exploit the value-add of presence with VoIP. As presence and profiles evolves to a system or network service and can be accessed across multiple networks, it becomes a crucial functionality for enabling dynamic and virtually unrestricted communication between individuals.

The aim of this paper is to explore how presence and profiles can enrich the user experience by providing relevancy and flexibility in setting up and forwarding calls. Our objective is to gain understanding of how the current VoIP technology, Presence and Profiles can be brought together to create compelling human communication VoIP applications that deliver enriched experience to end users. Our primary methodology is to learn through prototyping. The prototypes provide the basis for demonstrating the concepts and identifying technology components in realizing the objective.

2. The User Scenarios

The following scenario demonstrates how a conference call can be setup automatically based on conference participants' availability (presence state) and user company hierarchical contact list (profile information) to forward an incoming call.

2.1. SCENARIO DESCRIPTIONS

Anne wants to have a conference call with her colleagues, Ben and Claire as soon as they are available. Claire is away from office carrying her PDA terminal while Ben is having his mobile phone. Anne makes a conference setup request and invites the participants via web browser.

The system then sends notification via IM or SMS to Ben and Claire regarding the conference request. Claire receives the IM and accepts the invitation. Ben receives the SMS notification and chooses to forward the invitation to his subordinate, Den. The system detects Ben's company profiles and prompts a contact list automatically for Ben to select forwarding called party. Den accepts the invitation, but currently having a meeting. Anne receives the notification about Ben's choice and accepts Ben's call forwarding.

The availability statuses of Claire and Den are monitored by the system. After Den's meeting, the system detects that now Claire and Den are all available. The system then automatically establishes a conference call for Anne, Claire and Den.

2.2. SCENARIO PROCESS FLOW

Figure 1 shows the flow of scenario described in section 2.1.

3. System Architecture

The system architecture to realize the employment of presence and profiles in delivering VoIP services is proposed in Figure 2. The main components of this architecture solution are Conference Bridge Server, Presence Server, Enterprise Directory Server, Call Service Manager, Presence Status Monitor, Conference Mediator and Sip clients. The communication protocols used in this architecture are SIP, HTTP and LDAP.

Call Service Manager provides web interface to allow user to initiate conference. It also handles participants' response to the conference invitation, whether to accept, reject or forwarded it to other contacts. The Call Service Manager will then send a request Presence Status Monitor to monitor the presence status of the participants.

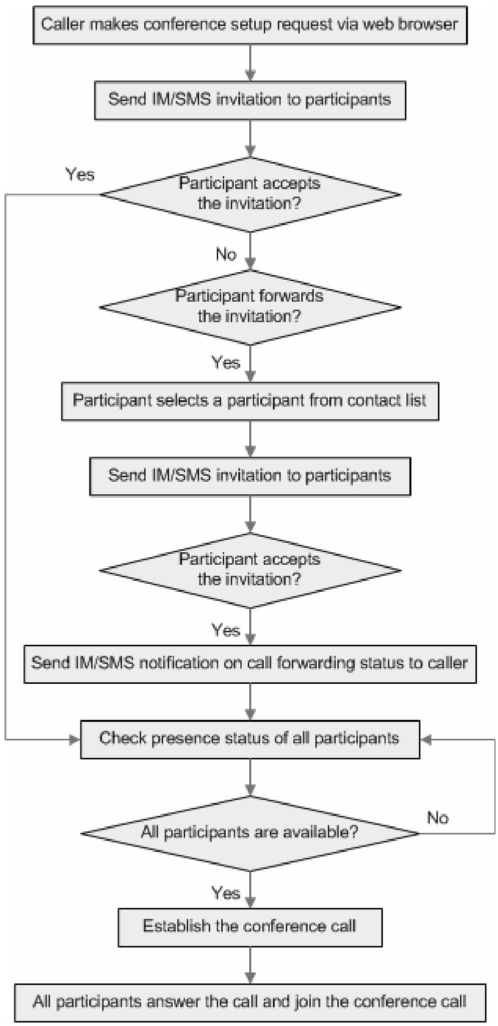

Figure 1. Flow of conditional conference call scenario.

Presence Status Monitor retrieves and checks the presence activities of the participants from the Presence Server. When all participants are available, it sends conference setup request, together with the list of participants, to Conference Mediator.

Conference Mediator provides an interface for Presence Status Monitor to initiate conference when all participants become available. It then sends conference setup request to Conference Bridge Server. The Conference Bridge Server provides and handles the media resource in the network such as playing announcement and audio mixing capability for conference call. It receives request from Conference Mediator to establish conference call for the SIP clients.

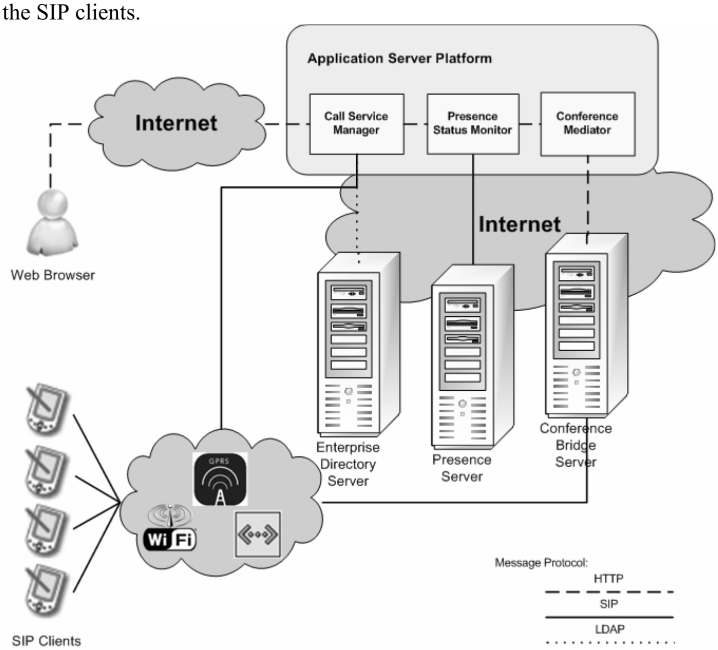

Figure 2. System Architecture.

4. Prototype Implementation

As shown in Figure 3, Call Service Manager allows user to invite conference participants to join a conference and send IM/SMS notification to participants via web browser. If any participant chooses to forward the conference invitation, Call Service Manager is able to detect that the participant is at work and retrieve corresponding contact list from the Enterprise Directory Server. User can select who to forward the conference invitation to from the contact list. When all participants have responded to the invitation, the Call Service Manager will then send the request to Presence Status Monitor server via Web Service to request for monitoring the presence status of the participants. When all participants are available, Presence Status Monitor sends a conference setup request to Conference Mediator via Web Service. Conference Mediator will then send a request to Conference Bridge Server to initiate the conference call.

The Lightweight Directory Access Protocol (LDAP) Server [Wahl, 1997] provided by Fedora Directory Server [Fedora], version 1.0.4, is used as the enterprise directory server. LDAP stores its data in hierarchically manner, similar to top-down representations of DNS trees or UNIX file directories. In this scenario, record(s) stored one level above (manager) and one level below (subordinates) of the called party is retrieved. These records are parsed as contacts list in hierarchical structure to the Call Service Manager.

Asterisk [Asterisk], an open source IP PBX software by Digium Inc. is used as conference bridge server running on Linux CentOS 4.4 Operating System. A customized CGI/Perl scripts is written in Conference Bridge Server which act as a gateway between Conference Mediator and Asterisk server.

Conference Bridge Server receives conference participants' contact information provided by the Conference Mediator when all participants are ready to join the conference, and dynamically creates Asterisk call files for each participant in real-time. The call files created are used to trigger and automate a conference call setup by ringing and connecting (through auto-call dial-plan) all participants' devices to its audio MeetMe conference room.

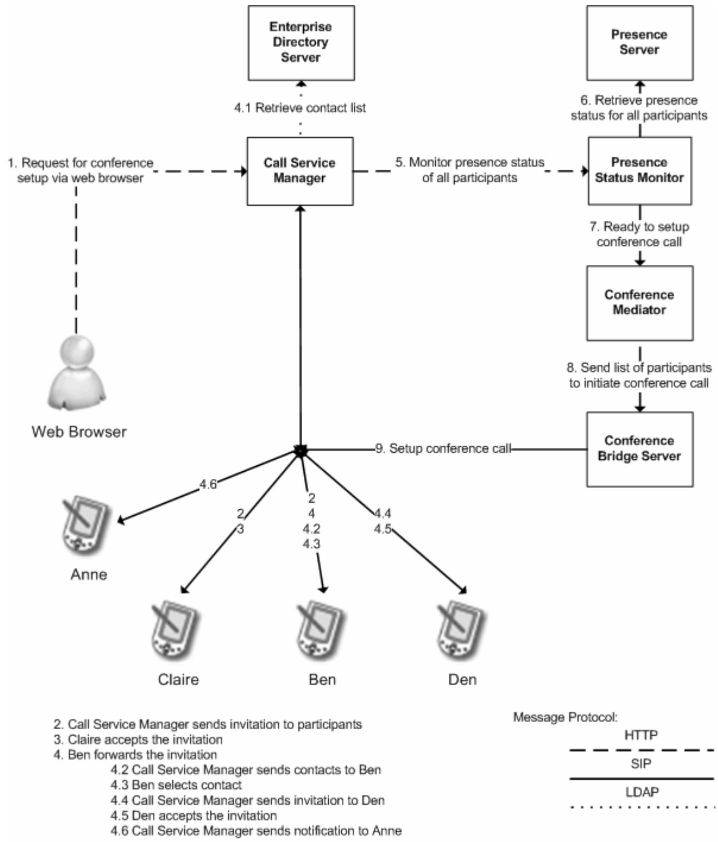

Figure 3. Implementation flow of presence-profiles based VoIP service scenario.

5. Conclusion

In this paper, we present our solution in developing and implementing the presence and profiles based VoIP call services. It describes our exploration on how the merging of technology in VoIP, Presence and Profiles can be brought together to automate conference call setup and provide forwarding options based on presence and user profiles.

Presence and profiles can be used as parameters to adapt VoIP system behaviours to enhance the quality of user experience by dynamically enabling automated and personalized services. However, further investigations and experiments are required in order to assess the usability aspects of our proposed system such as effectiveness of use, user interface design metrics and acceptance analysis in handling the call service process.

As presence and profiles evolves to a system or network service, it becomes a crucial functionality for enabling dynamic and virtually unrestricted communication between individuals – this will require services to be accessed across multiple networks via fixed mobile convergence technology. This is one of the important keys in delivering VoIP services - which technologically enable human-to-human communication that drives decisions making - should be accessed seamlessly, regardless of user device and connection.

References

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