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RESEARCH ARTICLE

Optimum Mobility Performance with Mobile IP & SIP

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Abstract— Mobility will place an efficient role in now days in the wireless communications. Mobile IP and SIP provide the mobility service to the handset users. The problems in Mobile IP and SIP s are triangular, handoff, Intra domain problems. These problems create signal lose and improper signalling to the user. To overcome these we provide add of service to the Mobile IP and SIP with the integration of the two services. This service provides optimum performance of the system.

Keywords: - Mobile IP; SIP; Mobile Agent; Foreign Agent

I. INTRODUCTION

Recent years have seen an explosive growth both in the number of laptop and notebook computers sold, and in the number of nodes connected to the Internet and the World Wide Web. The notebook computers are themselves ever more powerful, equal in processing capability to many systems sold as desktop workstations. In fact, the future growth of the Internet is likely to be fueled in large part by these very notebook computers, since they account for the part of the computer market that is growing fastest.

Our main theme here is to compare the network layer solution (i.e. Mobile IP) with the application layer solution (i.e. SIP) to support terminal mobility in VoIP services and then propose a integrated model, to reduce the inter-domain handoff(also known as macro-mobility) delay in VoIP services in a mobile environments. This is a continuation of our on-going work in wireless/mobile networks.

Unlike Mobile IP [1], however, the proposed approach limits tunneling to UDP connections that are active during a movement. Also, for SIP traffic, it limits tunneling to traffic that are sent to mobile node until the re-INVITE is completed in the rest of this paper, we provide problems in mobile IP and SIP in Section 2. Section 3 is devoted to our solutions to mobility support in SIP. Section 4 shows the optimum service of the system evaluation. Finally we give some conclusion remarks in Section 5.

II. OVERVIEW OF MOBILE IP & SIP

Mobile IP is a standard that allows users to move from one network to another without losing connectivity. Mobile devices have IP addresses that are associated with one network and moving to another network means changing IP address. Using the mobile IP system will allow users to achieve this and at the same time make the underlying process transparent for a user.

Basics of IP addressing

All computers that are connected to the Internet need to have a valid IP address. This address is usually assigned by an Internet Service Provider (ISP) which in turn has bought a block of addresses from the Internet Cooperation for Assigned Names and Numbers (ICANN). Most companies never interact with the ICANN directly. In order for a company to receive valid IP addresses they contact a local ISP.

Even local ISP: s do not interact with ICANN but in turn they contact larger ISP:s and only they contact ICANN.

IP addresses are not assigned randomly, but are carefully chosen. When a host wants to connect to the Internet he is assigned an IP address that represents the network that he is located at. This network is in turn a part of a larger network, which is then again a part of an even larger network, and in this way all networks are arranged, in a hierarchical manner. The IP address is divided into several parts where each corresponds to a given network. The rightmost part in the address always represents the host and the part to the left of it then represents the network that the host is located at and so on. In this way the leftmost part always represents the largest network. This hierarchy can be compared to a resident's home address.

Session Initiation Protocol (SIP)

SIP was originally developed by the IETF Multi-Party Multimedia Session Control Working Group (MMUSIC). Version 1.0 was submitted as an Internet Draft in 1997. Significant changes were made to the protocol and resulted in a second version, version 2.0, which was submitted as an Internet Draft in1998. The protocol achieved proposed standard status in March 1999 and was published as RFC 2543 [33] in April 1999. In September 1999, the now closed SIP Working Group was established by the IETF to meet the growing interest in the protocol. An Internet Draft containing bug fixes and clarifications to SIP was submitted in July 2000, referred to as RFC 2543 "bis." This document was eventually published as RFC 3261 [1], which obsoletes (or replaces) the original RFC 2543 specification. In addition, many SIP extension RFC documents have been published.

The popularity of SIP in the IETF has led to the formation of other SIP related working groups. The now closed Session Initiation Protocol Investigation (SIPPING) working group was formed to investigate applications of SIP, develop requirements for SIP extensions, and publish best current practice (BCP) documents about the use of SIP. Currently, the SIPCORE working group is Responsible for the core SIP standards. The SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) working group was formed to standardize related protocols for presence and instant messaging applications. Other working groups that make use of SIP include the PSTN and Internet Internetworking (PINT) working group and the Service in the PSTN/IN Requesting Internet Services (SPIRITS) working group.

III. ARCHITECTURE

The basic concept of the proposed architecture is that the SIP mobility support approach does not necessarily exclude the Mobile IP approach, rather it may work to complement based on the kind of application. The network architecture of the proposed location management scheme is shown in Fig.2. It uses a SIP network server, and a Mobility Agent with SIP Registrar to facilitate location management. While the SIP network server handles call/session delivery, the Mobility Agent with SIP Registrar is used for handling location registration, location updates, and location queries.

Handoff

For a successful handoff, a mobile node registers with the affiliated home agent to indicate its location, allowed by a dynamic join and departure of the multi-party conference for handoff. A location update may be performed upon the mobile crossing a service area boundary as described in the previous subsection.

Figure 3 shows the exchange of [8] handoff-related message when a mobile node moves into a new domain while call in progress. The process of using proposed approach to handoff is outlined below. The mobile node not only obtains an initial care-of address, but also acquires globally unique Care-of Address (i.e. co-located care of address).

A. Mobility Agent with Registrar

The purpose of the registration process in mobile IP is to inform mobility agent of a mobile node's new IP address and update the binding information between home address of mobile node and the care-of address. This allows TCP corn on-SIP connections to be maintained without a disruption.

For the purposes of our paper, it is useful to consider two types of binding. "user address binding", analogous to binding in an SIP Registrars, is the mapping between user level identifier and a temporary IP address of host name, and "IP address binding", roughly the binding between a permanent IP address identifying a host to a temporary care of address. The Purpose of this table is to map a mobile node's *home address* with its *Current*

Location (or *collocated care-of address*) and forward packets accordingly. Also, it is used to support the binding between a *mobile user's permanent identifier* and *his/her computer's actual IP address*. The former is designed for IP address binding, and the latter is defined for user address binding.

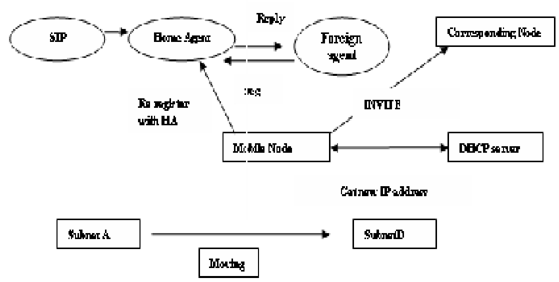

Fig 1: Architecture

In addition it may store other user profiles such as QoS requirements. This information is stored on a home agent using SIP REGISTER message when a host is first registered on the network. Also, if a user changes the device being used, it updates binding information, such as *User URI*, *Contract URI*, *Home Address*, and *Current Location* by using SIP Re-REGISTER message *[5]*.

IV. OPTIMUM SERVICE OF THE SYSTEM EVALUATION

In this section we present our simulation results.

Simulations are conducted to investigate two critical performance issues, i.e. packet loss during handoffs and handoff latency. The simulations are run using NS2 (v2.1b9a) [6] from Lawrence Berkeley National Laboratory (LBNL), which is widely used in the networking community to study IP networks. Because the current version of NS2 does not support SIP for VoIP [9] service, we add a suite of new features (user agent, redirect server, proxy server, registrars) and procedures that are specific to this paper.

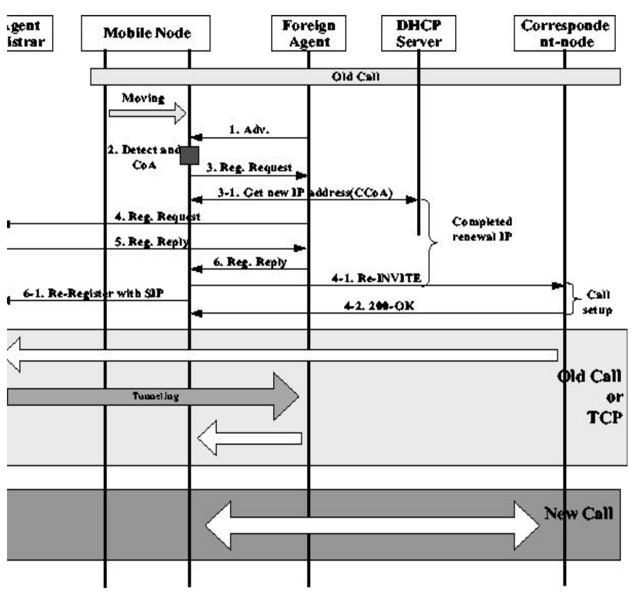

Fig.2 Message flow during a Handoff

Comparison SIP vs. IP mobility

	IP	SIP	RTP
constant	IP address	SIP url user@home.com	SSRC/SDES
update	binding upd.	re-INVITE	audio packet
security	via IIA	call-ID, crypto	crypto
impl.	OS	app.	app.
reg.	one (*)	proxy	none

Comparison of IP and SIP

V. CONCLUSION

In this paper, we proposed an efficient approach to deal with handoff during ongoing call over VoIP service. Unlike the previous work, the proposed approach uses integrated procedures of mobile IP and SIP mobility for real-time communication over UDP [7]. Compared with other proposals, our proposal has three primary advantages. First, the proposed mechanism can reduce packet loss and handoff latency by compensating mobile IP and SIP-mobility shortcomings mutually. Second, the only modification to the existing infrastructure is the extension in the home agent and the addition of a database to find registrars. Third, the complexity of the proposal occurs only in registration, call setup shares the single-lookup efficiency of SIP and is therefore relatively fast.

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