Chapter XIV
Real-Time Multimedia Delivery for All-IP Mobile Networks

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ABSTRACT

Recently, the Internet has become the most important vehicle for global information delivery. As consumers have become increasingly mobile in the recent years, introduction of mobile/wireless systems such as 3G and WLAN has driven the Internet into new markets to support mobile users. This chapter is focused not only on QoS support for multimedia streaming but also dynamic session management for VoIP applications: As the types of user devices become diverse, mobile networks are prone to be "heterogeneous." Thus, how to effectively deliver different quality levels of content to a group of users who request different QoS streams is quite challenging. On the other hand, mobile users utilizing VoIP services in radio networks are prone to transient loss of network connectivity. Disconnected VoIP sessions should be effectively detected without introducing heavy signaling traffic. To deal with the above two issues, an efficient multimedia broadcasting/multicasting approach is introduced to provide different levels of QoS, and a dynamic session refreshing approach is proposed for the management of disconnected VoIP sessions.

INTRODUCTION

By providing ubiquitous connectivity for data communications, the Internet has become the most important vehicle for global information delivery. The flat-rate tariff structures and low entry cost characteristics of the Internet environment encourage global usage. Furthermore, introduction of mobile/wireless systems such as 3G and WLAN has driven the Internet into new markets to support mobile users. As consumers become increasingly mobile, wireless access to services available from the Internet are strongly demanded. Specifically, mobility,

privacy, and immediacy offered by wireless access introduce new opportunities for Internet business. Therefore, mobile/wireless networks are becoming a platform that provides leading edge Internet services.

The existing point-to-multipoint (i.e., multicasting and broadcasting) services for the Internet allow data from a single source entity to be transmitted to multiple recipients. With rapid growth of wireless/mobile subscribers, these services are expected to be used extensively over wireless/mobile networks. Furthermore, as multimedia applications (e.g., video streaming and voice conferencing) are ubiquitous around the Internet world, multimedia broadcasting, and multicasting is considered as one of the most important services in future wireless/mobile communication systems.

As the number of mobile devices and the kinds of mobile applications explosively increases in the recent years, the device types become diverse, and mobile networks are prone to be "Heterogeneous." Multicast/broadcast users with different kinds of mobile devices may request different quality levels of multimedia streams due to (1) users' preferences, (2) service charges, (3) network resources, and (4) device capabilities. Thus, how to effectively deliver different quality levels of content to a group of users who request different QoS streams is quite challenging in the existing/future wireless/mobile communications. In this chapter, an efficient QoS-based multimedia broadcasting/multicasting approach to transmit multimedia streams to the users requesting different levels of service quality would be discussed.

Based on satisfactory and reliable streams delivered over radio network, services provided to fulfill user's strong demand for mobile technologies should then be considered. With the explosive growth of Internet subscriber population, supporting Internet telephony services, also known as voice over IP (VoIP), is considered as a promising trend in telecommunication business. Thus, how to efficiently provide VoIP services over mobile/wireless networks becomes an important research issue. Two major standards are currently used for VoIP products. One is proposed by the ITU-T/H.323, and the other is developed by the IETF/SIP (Internet engineering task force/session initiation protocol). SIP brings simplicity, familiarity, and clarity to Internet telephony that H.323 does not have.

Mobile users roaming in radio networks are prone to transient loss of network connectivity. For example, when a wireless VoIP user in conversation fails to connect the network (e.g., due to abnormal radio disconnection), the failure of this session might not be detected. As resources are still reserved for the failed session, new sessions could not be granted due to the lack of resources. To resolve this problem, one of SIP extensions, *SIP session timer* (Rosenberg, et al., 2002), specifies a keep-alive mechanism for SIP sessions. In this mechanism, the duration of a communicating session is extended by using an UPDATE request sent from one SIP user to the proxy server (then to the other SIP user). A session timer (maintained in the proxy server and the user) records the duration of the session that the user requests to extend. When the session timer nearly expires, the user re-sends an UPDATE request to refresh the session interval. Existing approaches to implement the SIP session timer mechanism are based on static (periodic) session refreshing. The selection of the length for the session timer significantly affects the system performance in the static session refreshing approach due to a tradeoff between resource utilizations and housekeeping traffic. In this chapter, a dynamic session refreshing approach to adjust the session interval according to the network state is discussed. The objective