*International Journal of Innovative Computing, Information and Control*
Volume **8**, Number **3(A)**, March **2012**

ICIC International ©2012 ISSN 1349-4198
pp. 1879–1895

# ITTP: A NEW TRANSPORT PROTOCOL FOR VOIP APPLICATIONS

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Received December 2010; revised July 2011

Abstract. *Over the past few years, the telecommunications sector has started moving toward the use of Voice over Internet Protocol (VoIP) technology. VoIP technology employs Internet infrastructure and protocols to transfer VoIP data between call parties. Unfortunately, none of the existing Internet transport layer protocols address VoIP application requirements. Typically, the Real-time Transport Protocol (RTP) application layer protocol and the User Datagram Protocol (UDP) transport layer protocol are bound together to address VoIP application requirements. However, a combination of RTP and UDP reduces the quality of VoIP applications and causes inefficient bandwidth utilization. In the present work, a dedicated transport protocol named Internet Telephony Transport Protocol (ITTP) was designed to carry VoIP application data. ITTP is designed to address key VoIP requirements and handle the problems resulting from RTP/UDP. A simple mathematical model was used to evaluate ITTP bandwidth efficiency and compare it with RTP/UDP. ITTP was found to improve bandwidth usage substantially. In addition, ITTP was simulated using Network Simulation 2. The results showed that ITTP had better performance compared with RTP/UDP in terms of packet loss, delay and bandwidth usage.*

**Keywords:** ITTP, VoIP, RTP, VoIP quality, Signaling protocols

1. **Introduction.** In the last decade, network equipment underwent a major performance "revolution". Network developers took advantage of this change to provide customers with new technologies. Voice over Internet Protocol (VoIP) is one such technology. VoIP utilizes network infrastructure to replace current circuit switching telephone networks called the Public Switched Telephone Network (PSTN) with packet switching telephone networks [1-3]. Furthermore, VoIP technology employs network protocols to transfer calls around the world.

Two main protocol categories are used in VoIP systems [3,4]. The first category comprises the signaling protocols which are used to establish and manage a session between call endpoints [4]. There are two standard signaling protocols for VoIP, namely, H.323 and the Session Initiation Protocol (SIP). H.323 was the first VoIP signaling protocol developed by the International Telecommunication Union (ITU), whereas SIP is a standard developed by the Internet Engineering Task Force (IETF). Gradually, SIP overtook H.323 and dominated the VoIP application world [5,6]. Recently, the InterAsterisk Exchange Protocol (IAX) has been introduced as a new signaling protocol. Unlike SIP and H.323, however, IAX is not yet a standard [4].

The second category comprises the media transfer protocols. Typically, media transfer protocols are used for the exchange of media data once a session is established between