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CONFERENCE GATEWAY FOR HETEROGENEOUS CLIENTS: REAL TIME SWITCHING CLIENTS AND INTERASTERISK EXCHANGE CLIENTS

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Abstract. *The InterAsterisk eXchange (IAX) protocol has been available for a few years. Meanwhile, the popularity of Real time SWitching (RSW) has increased due to its ability to easily combine voice and video services. Incidentally, these two heterogeneous clients pose considerable problems for users who have to choose between two solutions offering different advantages and disadvantages. While RSW is being used in many areas, IAX is being deployed in many VoIP products. Hence, RSW interoperability and coexistence with IAX is considered very important when maximizing the return on current investments and to support new deployments that could be RSW as an alternative packet telephony signaling protocol. We used IAX as opposed to SIP, which although started as a simple and an attractive method for VoIP, has become a complex and heavy protocol to implement. Similarly, H.323 is a very complex protocol suite that can result in the transmission of many unnecessary messages across the network.*
Keywords: InterAsterisk eXchange protocol, VoIP, SIP, H.323, Real time switching

1. **Introduction.** Voice over IP (VoIP) is a technology to transport voice over IP networks. This technology connects people via packet switched networks instead of traditional circuit switched networks. Because of the low cost of the Internet usage, VoIP can make telephone calls much cheaper than traditional public switched telephone networks (PSTN). Therefore, VoIP has attracted an increasing number of customers who previously opt for conventional communication solutions; as such, it is expected that VoIP will ultimately replace the PSTN [1]. Universities, enterprises, businesses and corporate entities have also invested in the development and utilization of VoIP systems. VoIP, as IP telephony, has technologically matured and currently gives service providers opportunity to offer both traditional services, apart from a wide range of new multimedia conferencing applications, and exploit the power of sophisticated user terminals. While services continuously provided for conventional VoIP, potential interaction with networks could further increase and enhance capabilities of multimedia conferencing systems. Yet, the VoIP system poses substantial challenges. The VoIP architecture must be able to support both traditional services and new services enabled by real-time multimedia conferencing endpoints. Real-time multimedia communications imply interactive communications can use various media (i.e., audio-conferencing and video-conferencing using IP networks). By leveraging the capabilities of multimedia conferencing system, VoIP service providers can offer multiple levels of services based on different levels of communication capabilities.