Chapter XLIV
The Playout Control Management: An Issue for the IP Telephony Service Providers

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ABSTRACT

IP Telephony provides a way for an enterprise to extend consistent communication services to all employees, whether they are in main campus locations, at branch offices, or working remotely, also with a mobile phone. IP Telephony transmits voice communications over a network using open standard-based Internet protocols. This is both the strength and weakness of IP Telephony as the involved basic transport protocols (RTP, UDP, and IP) are not able to natively guarantee the required application quality of service (QoS). From the point of view of an IP Telephony Service Provider this definitely means possible waste of clients and money. Specifically the problem is at two different levels: i) in some countries, where long distance and particularly international call tariffs are high, perhaps due to a lack of competition or due to cross subsidies to other services, the major opportunity for IP Telephony Service Providers is for price arbitrage. This means working on diffusion of an acceptable service, although not at high quality levels; ii) in other countries, where different IP Telephony Service Providers already exist, the problem is competition for offering the best possible quality. The main idea behind this chapter is to analyze specifically the state of the art playout control strategies with the following aims: i) propose
the reader the technical state of the art playout control management and planning strategies (overview of basic KPIs for IP Telephony); ii) compare the strategies IP Telephony Service Provider can choose with the aim of saving money and offering a better quality of service; iii) introduce also the state of the art quality index for IP Telephony, that is a set of algorithms for taking into account as many factors as possible to evaluate the service quality; iv) provide the reader with examples on some economic scenarios of IP Telephony.

INTRODUCTION

The combination of IP and a telephonic service gives IP Telephony. IP Telephony implements services like sending/receiving/management of voice and data-voice, between two or more users in a real time fashion over an already existing IP channel. The basic frameworks for implementing an IP Telephony solution are ITU-T H.323 (Rec. H.323, 2006) and SIP (RFC 3261, 2002). The ITU-T H.323 is a recommendation that defines the protocols to provide audio-visual communication sessions on any packet network, involving both the management of the call and the speech packet transport, including the speech coding, (see Figure 1).

The H.323 standard specifies four kinds of components, providing a point-to-point and point-to-multipoint IP Telephony service (see also Figure 2 from packetizer.com):

- **Terminals:** A personal computer (PC) or a stand-alone device, running an H.323 and the multimedia applications.

Figure 1. The ITU-T H.323

![Figure 1. The ITU-T H.323](image1)

Figure 2. The ITU-T H.323 components

![Figure 2. The ITU-T H.323 components](image2)