Abstract

Voice over IP (VoIP), the transmission of real-time voice over an Internet Protocol (IP) network, is gaining much momentum in the industry. Transmitting voice over an IP network has many benefits over transmitting voice over conventional telephone systems. These benefits are what have motivated the industry to develop VoIP systems, but the impairments present on an IP network make implementing such a system quite a challenge.

In this thesis, Voice over IP (VoIP) will be studied with a particular emphasis on the voice compression algorithms used in such a system. The desired characteristics of a voice compression algorithm for VoIP will be analyzed. Theses characteristics will then be used to judge how appropriate are some existing standards of voice compression algorithms for VoIP. An evaluation of voice quality of some existing voice compression standards will be done and the result will be compared with subjective tests.