Abstract

This work performs an analysis of several VoIP sources, where the sources are capable of adapting their transmission rates. This analysis is based on the network parameters (packet loss rate, end-to-end delay and jitter), on the voice quality and mainly on the fairness of the system. An algorithm is proposed for the adaptive adjustment of the transmission rate of VoIP sources, based on the voice quality estimated at the receiver. This adjustment is achieved through the appropriate use of differing voice codecs as the conditions of the voice quality change, with the aim of maintaining an efficient utilization of the available resources. In order to validate this proposal, an effort was made to simulate more realistic VoIP calls using sources that follow human conversations. The effect of this model is investigated and the proposed algorithm, adaMOS, is compared with the results of existing related work. Both simple and complex topologies modeling the actual Internet are considered. Simulation results show that the proposed algorithm makes efficient use of the available bandwidth, achieving a fair and superior performance in comparison to similar works from the literature.