Lost VOIP Packet Recovery in Active Networks

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Abstract

Current best-effort packet-switched Internet is not a perfect environment for real-time applications such as transmitting voice-over the network (Voice Over Internet Protocol or *VOIP*). Due to the unlimited concurrent access to the Internet by users, the packet loss problem cannot be avoided. Therefore, the VOIP based applications encompass problems such as "voice quality degradation caused by lost packets".

The effects of lost packets are fundamental issues in real-time voice transmission over the current unreliable Internet. The dropped packets have a negative impact on voice quality and concealing their effects at the receiver does not deal with all of the drop consequences. It has been observed that in a very lossy network, the receiver cannot cope with all the effects of lost packets and thereby the voice will have poor quality. At this point the *Active Networks*, a relatively new concept in networking, which allows users to execute a program on the packets in active nodes, can help VOIP regenerate the lost packets, and improve the quality of the received voice. Therefore, VOIP needs special voice-packing methods. Based on the measured packet loss rates, many new methods are introduced that can pack voice packets in such a way that the lost packets can be regenerated both within the network and at the receiver. The proposed voice-packing methods could help regenerate lost packets in the active nodes within the network to improve the perceptual quality of the received sound. The packing methods include schemes for packing samples from low and medium compressed sample-based codecs (PCM, ADPCM) and also include schemes for packing samples from high compressed frame-based codecs (G.729).

Using these packing schemes, the received voice has good quality even under very high loss rates. Simulating a very lossy network using NS-2 and testing the regenerated voice quality by an audience showed that significant voice quality improvement is achievable by employing these packing schemes.