D-4

Speech Coding

Because the telephone is the system for sending sampled speeches via communication channels, the study of speech coding—how to compress speech—and speech synthesis from compressed data were required. Around 1970, F. Itakura proposed the LSP system which became a basic special drop for speech compression and coding. T. Moriya, et al., proposed CS-ACELP targeting speech and PSI-CELP with a low bit rate, thereby contributing to its standardization. MPEG, the international standard, included audio coding in addition to speech coding. To this MPEG audio, A. Sugiyama, et al., proposed variable block length adaptive transform coding and the TwinVQ method, contributing to international standardization. As communication networks became broadband systems, demand for speech coding changed as well. So, Y. Hiwasaki, et al., proposed a coding system aimed at low delay, high quality in addition to compression rate, contributing to international standardization. T. Nomura, et al., proposed audio compression technology for one seg and mobile terminals, contributing to international standardization.

CS-ACELP: Conjugate Structure and Algebraic Code Excited Linear Prediction

PSI-CELP: Pitch Synchronous Innovation CELP

MPEG: Moving Picture Experts Group