Standard Voice Algorithms

Voice

With the knowledge of algorithms and many years of experience in firmware development, SEDA Solutions® has unique insight into many digital voice processing applications and their transformations onto floating and fixed-point processor platforms. Below are some of the examples of codecs previously implemented at SEDA Solutions®:

- **G.711**
  The most widely used digital representation of voice signals is that of the G.711 or PCM (Pulse Code Modulation). This codec represents a 4 kHz band limited voice signal sampled at 8 kHz using 8 bits per sample A-law or μ-law coding.

- **G.726**
  The protocol for the G.726 codec requires a 64 kbps A-Law or μ-law PCM signal to be encoded into four different bit rate options ranging from 2 bits per sample to 5 bits per sample. The algorithm is based on Adaptive Differential Pulse Code Modulation (ADPCM) and is based on 1 sample backward prediction scheme.

- **G.728**
  The G.728 algorithm compresses PCM codec voice signals to a bit rate of 16 kbps. This algorithm is based on a strong backward prediction scheme and is by far considered as one of the most complex voice algorithms to be produced by the ITU standard organization.

- **G.729**
  For compression of voice signals at 8 kbps the G.729 algorithm offers toll quality with built in algorithmic delays of less than 15 msec. Additional features described in the G.729 Annex ensure VAD[1] and Comfort Noise Generation functionalities to enhance the quality and reduce the overall bit rate.

- **G.723.1**
  The most widely used algorithm for band limited channels, such as VoIP and video conferencing, is that of G.723.1. The algorithm has two operating bit rates of 6.3 kbps and 5.3 kbps. Although the delay is not as low as that of the other ITU standards its quality is near toll quality for the given low bit rates, making it very efficient in bit usage.

- **GSM[2]—AMR**
  The latest GSM standard is the multi rate Adaptive Code Excited Linear Prediction that provides compression in the range of 4.75 to 12.2 kbps. In total the codec provides 12 bit rates that cover the half rate to full rate channel capacity.

- **GSM—FR**
  The first digital codec used in a mobile environment is the GSM Full Rate vocoder. The codec compresses 13 bit PCM sample signals to a rate of 13 kbps. The algorithm is based on a very simple Regular Pulse Excited – Linear Prediction Coding technique.

- **GSM—HR**
  To increase capacity, the GSM committee decided on a lower bit rate of 5.6 kbps for the voice channel. The algorithm is based on the Vector Sum Excited Linear Predictive (VSELP) and is computationally as complex as other low bit rate algorithms.

For more information on Voice related products and services please visit our web site at **www.sedasolutions.com** or contact us by email at **info@sedasolutions.com**.

SEDA SOLUTIONS®

[1] Voice Activity Detection
[2] Global System for Mobile Communications