Adaptive Digital Technologies, Inc.

G.728 Audio Coder

PRODUCT DESCRIPTION

The Adaptive Digital Technologies G.728 voice coder is an implementation of the ITU G.728 compression standard toll quality voice coder, based on the Low-Delay Code Excited Linear Prediction (LD-CELP) compression principles. G.728 coders are widely used for applications of telephony over packet networks, especially voice over cable and VoIP, where low delay is required. Additionally, G.728 is specified as part of the H.320 international video conferencing standard.

The G.728 vocoder operates on 2.5 milliseconds frames of input speech, corresponding to 20 16-bit samples at a sampling rate of 8000 samples per second. The G.728 Encoder compresses each frame of speech into 40 bits. These 40 bits are stored in 4 16-bit words, but only the lower 10 bits are significant.
The G.728 decoder takes in the 4 10-bit words of compressed data and expands them into 20 16-bit samples.

The decoder also incorporates an adaptive post-filter to enhance performance for multiple transcodings. The post-filter function can be omitted for single coder-decoder operation to reduce the processing power required on the DSP. The Adaptive Digital G.728 software is DAIS compliant.

Unlike other low bit rate voice compression algorithms, G.728, when running at 16 kbps, is able to pass voiceband modem signals at up to 2400 bits per second. Furthermore, G.728 is able to pass DTMF tones without an external bypass mechanism that is needed by most low bit rate vocoders.

G.728 Annex H: Variable bit rate LD-CELP which operates at 12.8 kbps, 9.6 kbps
This annex is intended to augment the performance of Recommendation G.728 by providing lower bit rate options in some specific applications, such as in those to be used with digital circuit multiplication equipment (DCME). G.728 Annex H enhances DCME by providing a balance between bit rate, voice quality, and delay that the other DCME codecs like G.723, G.729, and G.726 do not afford.

FEATURES

- G.728 delivers toll quality voice comparable to 32 kbit/s ADPCM in only half the bandwidth.
- Functions are C-callable.
- Can be integrated with echo cancellers, VOX and tone detection/regeneration.
- The encoder and decoder meet all ITU G.728 compliance data files interoperability requirements.
- Optional in-band synchronization available.
- eXpressDSP™ Algorithm Interoperability Standard Compliant

AVAILABILITY

ADT G.728 is available on the TMS320™ DSP Family
C54x™DSP Generation
C55x™DSP, & C64x™DSP Generations *(C55x data presented is preliminary)*

SPECIFICATIONS

Coding Rate: 16 kbps, 12.8 kbps, 9.6 kbps selectable at run time on the fly
Sampling Rate: 8 kHz
Delay: 2.5 msec (algorithmic)

C54x

All Memory usage is given in units of 16-bit word.

<table>
<tr><th rowspan="2">Function</th><th colspan="3">MIPS (Peak)</th><th rowspan="2">Program Memory</th><th rowspan="2">Data Memory</th><th rowspan="2">Per-Channel Data Memory</th></tr>
<tr><th>16000</th><th>128000*</th><th>9600*</th></tr>
<tr><td>Encode</td><td>24.24</td><td>19.66</td><td>17.93</td><td rowspan="2">8.0 k</td><td rowspan="2">0.5</td><td>724</td></tr>
<tr><td>Decode</td><td>16.5</td><td>16.5</td><td>16.5</td><td>1057</td></tr>
</table>

* G.728 Annex H

Last update: 08/23/2002

C55x Preliminary Data

All Memory usage is given in units of byte.

Function	MIPS (Peak)	Program Memory	Data Memory	Per-Channel Data Memory
Encode	35.0	18k	1k	3800

Last update: 02/11/2002

C64x

All Memory usage is given in units of byte.

<table>
<tr><th rowspan="2">Function</th><th colspan="6">MIPS (Peak)</th><th rowspan="2">Program Memory</th><th rowspan="2">Data Memory</th><th rowspan="2">Per-Channel Data Memory</th></tr>
<tr><th colspan="2">16000</th><th colspan="2">128000*</th><th colspan="2">9600*</th></tr>
<tr><td>Encode</td><td colspan="2">16.5</td><td colspan="2">14.5**</td><td colspan="2">13.5**</td><td rowspan="3">41 k</td><td rowspan="3">3.1 k</td><td>1444</td></tr>
<tr><td rowspan="2">Decode</td><td>PF</td><td>No PF</td><td>PF</td><td>No PF</td><td>PF</td><td>No PF</td><td rowspan="2">2140</td></tr>
<tr><td>14.9</td><td>11.8</td><td>15.1</td><td>11.8</td><td>15.1</td><td>11.8</td></tr>
</table>

* G.728 Annex H

**Preliminary Data

Last update: 06/27/2005

FUNCTIONS

```
G728_ADT_initEnc(…);
G728_ADT_initDec(…);
G728_ADT_encode(…);
G728_ADT_decode(…);
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TERMINOLOGY

DCME - type of voice compression equipment that is installed at either end of a long-distance link (typically communications satellite or submarine communications cable)

LD-CELP - Low-Delay Code Excited Linear Prediction

DAIS –TI's eXpressDSP™ Algorithm Interoperability Standard

ITU - International Telecommunication Union

Adaptive Digital is a strategic member of the Texas Instruments Developer Network.

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