ADPCM: US Patents from 2010 to 2016

Muhanned AL-Rawi
University of Ibb, Yemen, E-mail: muhrawi@yahoo.com

***Abstract*— Due to its importance and its wide applications in telecommunication networks, the Adaptive Differential Pulse Code Modulation(ADPCM) was standardized by International Telecommunication Union(ITU). This paper presents and describes briefly the recent US patents on ADPCM starting from 2010 to 2016. Also, the general structure of ADPCM is described. The work in this paper opens the minds of the researchers and encourages them to continue doing research on this field.**

***Index Terms*— US Patents, ADPCM**

I. INTRODUCTION

With the increase in demand for efficient use of digital communication channel, various types of highly effective speech coding methods have been developed. As one of these coding methods is international standard Adaptive Differential Pulse Code Modulation (ADPCM)[1] which was standardized by International Telecommunication Union(ITU). The superior performance, economy and application flexibility of ADPCM relative to other bandwidth reduction techniques were the prime reasons for its selection.

The specification of ADPCM opens the door to a host of applications in telecommunication networks. These applications can be divided into three categories: telephone company use, end customer applications, and new service offerings.

II. STRUCTURE OF ADPCM

Fig.1 shows simplified block diagram of ADPCM codec. Two major components form the algorithm: an adaptive quantizer and an adaptive predictor. The relation between the encoder and the decoder is also depicted. The difference between them is that the encoder has adaptive quantizer(**Q**) and inverse adaptive quantizer($\mathbf{Q^{-1}}$) ,while, the decoder has inverse adaptive quantizer only. The decoder is simply a subset of the encoder and transmits r(n) as its output instead of c(n). The adaptive predictor computes an input signal estimate ŝ(n) which is subtracted from input signal s(n) resulting in a difference signal d(n). The adaptive quantizer codes d(n) into codeword c(n) which is sent over the transmission facility. At the receiving end, an ADPCM decoder uses c(n) to attempt to reconstruct the original signal s(n). Actually, only r(n) can be reconstructed which is related to the original input signal s(n) by

$$r(n) = s(n) + e(n) \tag{1}$$

where

$$e(n) = dq(n) - d(n) = r(n) - s(n) \tag{2}$$

is the error introduced by the quantizer, and dq(n) is the output of inverse adaptive quantizer
A typical measure of the ADPCM performance is given by signal-to-noise ratio (SNR)

$$SNR = E[s^2(n)]/E[e^2(n)] = \sigma_s^2/\sigma_e^2 \tag{3}$$

where E denotes expectation, σ_s^2 is the power (or variance) of input signal, &σ_e^2 is the power (or variance) of the error signal.

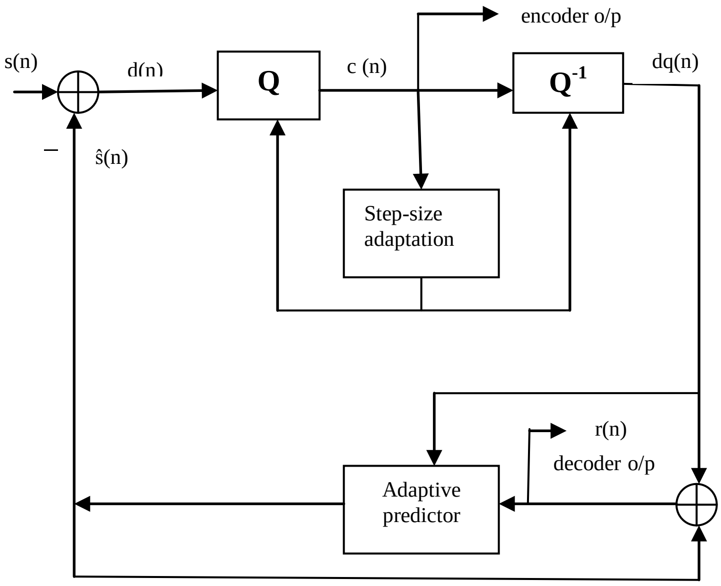

Fig.1 ADPCM Codec

III. US PATENTS ON ADPCM

This section describes briefly the recent US patents on ADPCM published from 2010 to 2016 as follows:

In [2], a communication processor was invented including ADPCM used for audio applications. The invention provides a low-cost, low-power, easily manufactured, small form-actor network access module which has a low memory demand and provides a highly efficient protocol decode. The invention comprises a hardware-integrated system that both decodes multiple network protocols in a streaming manner concurrently and processes packet data in one pass, thereby reducing system memory and form factor requirements, while also eliminating software central processing unit overhead. The presently preferred communications processor incorporates a protocol engine, a set of peripherals for the protocol engine, an Internet tuner core or other network stack, an external controller interface, and a memory interface. The communications processor thus provides network, e.g.

Internet, connectivity to a wide range of consumer network devices and industrial network devices.

In [3], a method and apparatus for performing multiple descriptive source coding in which a plurality of homogeneous encoders are advantageously employed in combination with a corresponding plurality of advantageously substantially identical decoders. In particular, diversity is provided to the multiple encoders by modifying the quantization process in at least one of the encoders such that the modified quantization process is based at least on a quantization error resulting from the quantization process of another one of the encoders. In this manner, diversity among the multiple bit streams is obtained, and in particular, the quality of a reconstructed signal based on a combination of multiple decoded bit streams at the receiver is advantageously superior to that based on any one of the decoded bit streams. In accordance with a first illustrative embodiment of the present invention, two Pulse Code Modulation (PCM) coders are employed. In accordance with a second illustrative embodiment of the present invention, two ADPCM coders are employed. And in accordance with a third illustrative embodiment of the present invention, two Low-Delay Code Excited Linear Prediction (LD-CELP) coders are employed. In each case, diversity is ensured by an appropriate modification to the quantization process of at least one of the encoders, and the total error may be advantageously reduced when decoded bit streams from both coders are combined at the receiver.

The invention in[4] decoded encoded speech using alternative parameters upon detection of a lost packet. Upon detection of a first good packet following packet loss, this invention uses second alternative parameters intermediate between the default parameters and the alternative parameters for a predetermined interval. Thereafter the invention reverts to the default parameters. This minimizes glitches in the decoded speech upon packet loss. This invention is suitable for use in decoding speech data encoded in the ITU Recommendation G.726 ADPCM based speech coding standard. The least mean square (LMS) in the G.726 standard is a sign-sign and leaky algorithm having a two poles and six zeros predictor. This prior art predictor needs persistent excitation to operate stably. In this invention during packet loss, the decoder is excited by the pitch quantized inputs of the previous packet. The leak factor and the step size of the predictor are controlled in two steps to have the better performance and stability during and just after packet loss. In this two steps control: step one changes the leak factor and step size during the packet loss; and step 2 changes the leak factor and step size upon reception of the very first good packet for the duration of one pitch period overlap. Similarly the scale factor of speed control adaptation is controlled in two steps during the packet loss.

A wireless communication device was invented in[5] which sets a digital link on a wireless channel between a master device and a slave device, compresses a sound signal in an ADPCM scheme, and carries the sound signal in a sound packet to perform communication. In a master device, a PCM signal is converted into ADPCM data by an ADPCM encoding unit. Next, the least significant bit of n-bits is set according to the number of "1" of the n-bits of the ADPCM data by a transmission conversion table. In a slave device, it is determined whether error has occurred according to the number of "1" of the n-bit data in received data. If error exists, the n-bit sound data is converted into mute data through a reception conversion table. The sound data converted through the reception conversion table is converted into a PCM signal from an ADPCM format by an ADPCM decoding unit. According to the aspect of the present invention, if the deterioration of the communication environment starts, a transmission side converts a part of the digital sound data into error detection bits to transmit the data with error detection bits, and a reception side converts the digital sound data according to the values of the received error detection bits. Accordingly, error detection can be performed with respect to the respective digital sound data with the data rate of the sound maintained, and thus the error detection accuracy can be improved with the sound quality maintained to some extent even if the communication environment is deteriorated.

In [6], a signal corresponding to a short-period change and a signal corresponding to a long-period change of a sound signal are detected, and optimal quantization is performed based on the combination of the two signals. In an ADPCM encoding apparatus, a differential value between a 16-bit input signal and a decoded signal of one sample ago is calculated by a subtractor. Thereafter, the 16-bit differential value is adaptively quantized by an adaptive quantizing section, so as to be converted to a (1 to 8)-bit length-variable ADPCM value. Thereafter, the ADPCM value is compression-encoded by a compression-encoding section to generate a signal which is framed by a framing section and outputted. Further, in an ADPCM decoding apparatus, a framed input signal is subjected to a reverse of the aforesaid process so as to be decoded.

A method is described in [7] for packing variable-length entropy coded data into a fixed rate data stream along with resolution enhancement data, the method providing tightly constrained propagation of transmission channel errors and graceful degradation of signal resolution as entropy-coded data rate increases. An application to a multiband ADPCM audio codec is also described. According to a first aspect of the present invention, a sequence of signal samples is encoded to furnish a sequence of composite codewords containing bits, by the method of: maintaining a conveyor comprising a sequence of slots, including a start slot and an end slot, wherein each slot is adapted to contain one or more bits and wherein the conveyor is adapted to effectively move the bit content of each slot to an adjacent slot after the passage of a time step, whereby bits are conveyed from earlier slots to later slots with the passage of each subsequent time step; and, for each signal sample performing the steps of: representing the signal sample by a coarse codeword and a touchup codeword, each of the coarse codeword and touchup codeword containing bits; distributing the bits of the coarse codeword into slots in the conveyor; retrieving a delayed codeword from the end slot; and, combining the touchup codeword with the delayed codeword to furnish a composite codeword.

In [8], methods, systems and computer-readable medium reduce and/or eliminate errors in coding/decoding of streamed audio due to resetting of decoder state values based on playback buffer access. An inaudible compensation signal is included with the audio signal. The compensation signal is generated having a characteristic selected so that the encoded streamed audio signal substantially matches the reset state values at block boundaries. In an ADPCM example, the compensation signal is chosen such that the sum of the compensation signal and the original audio signal (=the compensated audio signal) has the characteristic that, at the block boundaries, the compensated audio signal matches the initial predictor value.

In[9], a voice processing apparatus is provided in an ADPCM voice transmission system in which voice data that is differentially quantized through an ADPCM scheme is transmitted. The voice processing apparatus includes an error detector which detects whether or not an error occurs in a transmission frame containing data that indicates a differential value, and an error determiner which determines a level of the error detected by the error detector when the error detector detects the error. The voice processing apparatus also includes a voice processor which corrects the voice data with a correction value depending on the level of the error detected by the error detector and an ADPCM decoder which decodes the voice data corrected by the voice processor. According to the aspect of the invention, since the correction of the voice data is performed by using the correction value according to the level of the error, the voice data in which the level of the error is high and the voice data in which the level of the error is low can be corrected by using a different correction value. Therefore, the error in a low level can be slightly corrected and only the error in a high level can be largely corrected, so that while suppressing the noise of the voice data in which the error is in the high level, the original voice in which the error is in the low level is hardly converted. Accordingly, while suppressing the noise, the recognition of the voice becomes easier.

In lossy data compression of a signal using ADPCM, an adaptive decorrelation or "prediction" filter is used[10] to reduce the amplitude of the signal, the spectral dynamic range of the signal also being reduced. This latter reduction is effected in a nonuniform manner, if known techniques are used, with regions of high spectral density being compressed more than regions of low spectral density. The present invention recognises that using a uniform compression ratio results in a better tradeoff between compression and robustness to transmission channel errors. A method is described for obtaining a uniform compression ratio by adjusting coefficients of the decorrelation filter in dependence on coefficients of an adaptive training filter that is fed from the output of the decorrelation filter. A reverse method is also provided along with encoder, decoder and codec implementing the techniques. The present invention thus employs a filter depending on adjustable time-varying coefficients to furnish a partially whitened signal and also employs an adaptive whitening filter for performing a further filtering operation.

In[11], a device was invented which divides a sampled sound signal into a high frequency signal and a low frequency signal, individually encodes the high frequency signal and the low frequency signal, and generates error detection code pertinent to the high frequency ADPCM data and the low frequency ADPCM data. The device replaces data pertinent to some of a plurality of bits which configure the low frequency ADPCM data with the error detection code and transmits them. A receiver side receives the high frequency ADPCM data and the low frequency ADPCM data, and individually processes the high frequency ADPCM data and the low frequency ADPCM data in accordance with a value of the error detection code. According to the present invention, in a wireless wideband voice communication, high frequency ADPCM data and low frequency ADPCM data are individually processed by means of an error detection code that is transmitted after replacement of some of bits of the low range ADPCM data. Therefore, even when the communication environments become worse, wideband sound quality can be maintained by a fixed-size packet without changing a data rate of sound.

In[12], a method for coordinating an encoder and a decoder in a wireless communication system employing discontinuous transmission was developed, the method comprising performing a syncless reset command on the encoder and the decoder, in order to bring the encoder and decoder into stable and compatible states. The syncless reset command can be implemented by setting the variables of the encoder and decoder into predetermined values, or by performing a normal reset command followed by encoding or decoding a predetermined number of predetermined sample values. An aspect of the disclosure relates to a method for coordinating an ADPCM-based encoder comprised in a transmitting unit of a communication system and an ADPCM-based decoder comprised in a receiving unit of the communication system after a silence period, wherein the communication system employs discontinuous transmission, the method comprising: the encoder performing a first syncless reset operation consequent to receiving audio data; the encoder operating on the audio data and setting one or more encoded values in a buffer of samples; the transmitting unit transmitting the content of the buffer of samples; the receiving unit receiving the content of the buffer of samples; the decoder performing a second syncless reset operation; and the decoder decoding the received content.

In [13], a method and apparatus are provided for controlling the shaping of encoding noise during the ADPCM encoding of a digital audio input signal. The noise-shaping is carried out through the use of feedback that comprises filtering noise. The method includes the following steps: obtaining a parameter for indicating a high spectral dynamic range of the signal, the parameter indicating a risk of instability of the feedback; detecting a risk of instability by comparing the indication parameter to at least one predetermined threshold; limiting the feedback in the event that a risk of instability is detected; and gradually reactivating the feedback over a predetermined number of frames subsequent to the current frame for which the feedback is limited. Also provided is an encoder with feedback, including a control module implementing the control method as described. Thus, the method makes it possible to undertake fast and effective detection of potentially problematic signals running the risk of giving rise to phenomena of instability in the feedback.

This phenomenon is thus avoided since limitation of the feedback is performed upon the appearance of these at-risk signals. This limitation can even consist of complete deactivation of this feedback. The method therefore has a preventive action on the appearance of troublesome phenomena. The reactivation of the feedback loop is thereafter undertaken in a progressive manner so as not to give rise to overly abrupt variations of the coded signals.

In[14], a method of packet loss concealment in ADPCM codec with a packet loss compensation (PLC) circuit is provided. The method provides a predetermined transition period between a correct signal and a substitute signal and a difference between the substitute signal and a computed prediction signal is combined with a dequantized prediction error to receive a dequantized combined prediction error which is added to a predicted signal to provide a combined transition signal as basis for an output signal during the predetermined transition period for adapting all decoder parameters. In an embodiment, the error combiner circuit comprises at one input an analysis filterbank for downsampling of the substitute signal, received from the PLC circuit, into subband signals and at another input, an adaptive dequantization unit for the encoded, quantized, downsampled prediction error received from the input of the ADPCM decoder. An adaptive prediction unit is connected with one of two outputs to a subtractor, receiving the subband substitute signal from the analysis filterbank, and with the other output to an adder. A concealment prediction error shaper, connected to the output of the adaptive dequantization unit, is positioned between the subtractor and the adder and the output of the adder has a feedback loop to the adaptive prediction unit and leads to a synthesis filterbank for recombining the resulting combined subband substitute signals to gain an output signal. The concealment prediction error shaper produces, in a predetermined manner, a weighted sum of the dequantized prediction error and the prediction error of the subband substitute signal.

IV. SUMMARY AND CONCLUSION

Recent US patents on ADPCM were presented in order to show the importance and the applications of ADPCM in telecommunication networks. The general structure of the ADPCM was also presented.

REFRENCES

[1] *40, 32,24,16 kb/s ADPCM,* CCITT Recommendation G.726-1990.

[2] J. Minami , et al," Communications processor", US Patent 7,646,790, January 12, 2010.

[3] L. Cheng-Chieh, "Method and apparatus for diversity control in multiple description voice communication", US Patent 7,756,705, July 13, 2010.

[4] S. Kumer, "Stabilization and glitch minimization for CCITT recommendation G.726 speech CODEC during packet loss scenarios by regressor control and internal state updates of the decoding process", Us Patent 8,204,753, June 19, 2012.

[5] Y, Suwa, et al, "Wireless communication device and wireless communication system", US Patent 8,457,185, June 4, 2013.

[6] Y. Sato, et al, "Adaptive differential pulse code modulation encoding apparatus and decoding apparatus", US Patent 8,482,439, July 9, 2013.

[7] P. Craven, et al, " Lossy coding of signals", US Patent 8,533,571, September 10, 2013.

[8] A. Chau, "Methods and systems using a compensation signal to reduce audio decoding errors at block boundaries", US Patent 8,649,523, February 11, 2014.

[9] Y. Suwa, et al, "Voice processing apparatus and method for detecting and correcting errors in voice data", US Patent 9,021,318, April 28, 2015.

[10] P. Craven, et al, "Prediction of signals", US Patent 9,106,241, August 11, 2015.

[11] A. Shibuta, et al, "Wireless communication device and communication terminal", US Patent 9,270,419, February 23, 2016.

[12] A. Gur, "Non-synchronized ADPCM with discontinuous transmission", US Patent 9,332,585, May 3, 2016.

[13] S. Ragot, et al, "Controlling a noise-shaping feedback loop in a digital audio signal encoder avoiding instability risk of the feedback", US Patent 9,489,961, November 8, 2016.

[14] Z. Markus, et al,"Method of packet loss concealment in ADPCM codec and ADPCM decoder with PLC circuit", US Patent 20160148619A1, May 26, 2016.

Muhanned AL-Rawi was born in Iraq,1969. He received B.Sc. degree in electrical engineering from AL-Mustansiryia University, Iraq, in 1990, M.Sc. degree in communication and electronics engineering from Baghdad University, Iraq, in 1993, and Ph.D. degree in telecommunication engineering from Bandung Institute of Technology, Indonesia, in 1999.
He is currently with the department of electrical engineering, University of Ibb, Yemen. His research interests include digital communications, and digital signal processing.