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Methods for Converting Speech codec between GSM and G.729 through transcoding system

Use Transcoding system for converting speech codec
Between GSM and G.729

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***Abstract*— As we know that GSM coder is used for mobile communication and G.729 is commonly used in internet communication eg.VOIP. For modern communication network it is important that both networks should use interoperability. In this paper we discuss the transcoding system which convert speech codec format between G.729 and GSM. Transcoding system which we required for connecting two different networks, it should required less computation than the conventional decode then-encode (DTE). Through this transcoding system speech codec format can easily exchange information between these two different networks**

***Keywords*—G.729, GSM, Transcoding System**

I. Introduction

The G.729 speech codec is standardize by International Telecommunication Union (ITU) is introduced for Internet applications. G.729 will be used for Voice over IP (VoIP), Videophones, Digital Satellite Systems, Integrated Services Digital Network (ISDN), Land-Digital Mobile Radio, Future Public Land Mobile Telecommunication Systems (FPLMTS), Digital Circuit Multiplication Equipment (DCME), Digital Simultaneous Voice and Data (DSVD), and other applications. This speech codec's relative low complexity makes it an attractive choice for Internet telephony. The ITU-T standardized 8 kbits/s speech codec to operate with a discrete-time speech signal. G.729 provides coding of speech signals used in multimedia applications at 8 kbits/s using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP) [1][2]. For wireless communications, the Global System for Mobile communication (GSM) suggests an efficient speech coding system called the GSM codec and gains a lot of popularity internationally [1]. It is obvious that transcoding between the GSM and the G.729 speech compression standards will become more and more important when we need to integrate these two speech phone standards to establish wireless and Internet telephony connections. A simple way for the transcoding is through a so-called Decode-Then-Encode (DTE) approach.First we decode the GSM speech code to reconstruct the compress speech then perform encoding of G.729 to complete transcoding process. In this process we study that transcoder have capability to direct transfer of coding parameter and it reduces the maximum computation but the quality of speech also maintain like DTE. As transcoding technique used, it reduce the cost of interoperability and promotes the feasibility in links of wireless mobile phone and Internet phone. In this paper we describe as follows. From reduction of computation, the transcoding methods are described in Section II. The proposed transcoding technique preferable for that we discuss some experimental result in section III. And finally conclusion discuss in section IV.

II. METHOD OF TRANSCODING

All the system have individual frame size and this frame create a problem for code transformation Frame synchrony transformation is extremely important since different frame size will cause data loss or overlap. Here we discuss a new transcoder which is very effective and have ability to transfer all the coding parameter from GSM to G.729 speech codec. Figure 1 shows the transformation of speech frame length. To transform each other directly, we adopt two G.729 speech frame and one GSM speech frame with 160 samples each.

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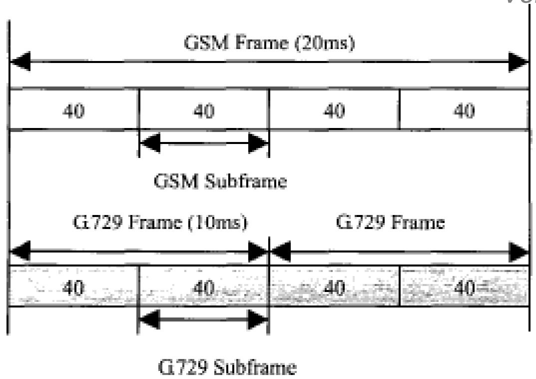

Figure1:- GSM/G.729 frame transformation structure

Table 1 show the function of each frame. The CPU time requirement is estimated from the C program of the G.729 encoder with Intel Pentium I1 CPU. In Table 1, the main three parts are short-term synthesis filter, long-term synthesis filter and the excitation. These three parts are required 63.1% CPU time. So the coder transformation is based on these three parts, will reduce the computation. Each of these transfomations is addressed below.

Function description	CPU time requirement
LPC, quantised and interpolated LSP	20.68%
Open-loop & close-loop analysis	23%
Fixed codebook search	19.42%
The others	36.9%

Table1:- The CPU time requirement of G.729 encoder

II TRANSFORMATION OF LPC PARAMETER

G.729 and GSM both contained LPC parameter. These LPC parameter are calculated from Levinson-Durbin algorithm
The vocal model is shown as

$$H(z) = \frac{1}{1 + \sum_{i=1}^{P} \alpha_i z^{-i}},$$

where α_i for all ***i,*** are the LP filter coefficients with ***P10.***
In G.729 we use LSP i.e. Line Spectrum Pair.LSP is more quantize and this LPC used for represent LPC parameters. These LSP are quantized by two stage VQ.And for GSM we applied three-segment vector quantizer of the reflection coefficients.

Let us consider the frame size of these two system we need to transform one GSM frame and two G.729 frames simultaneously. The LSP of subframe from GSM are obtained named G1, G2 G3 and G4 And the LSP of the G.729 for the second subframe are named *A2*and *B2.*As we obtain LSP from GSM then we used parameter in G.729 coder with different methods. The methods are as follow

Method 1: LSP ofA2= LSP of G*2*
LSP of *B2*= LSP of *G3*

Method 2: LSP ofA2= LSP of G*2*
LSP of *B2*= LSP of *G3*

Method 3: LSP of A2 =
1/4*LSP of G,+3/4* LSP of G2
LSP of B2
1/4* LSP of G,+3/4* LSP of ***G4***

We then use logarithmic spectral distortion (LSD) to measure the similarity between the DTE and methods of LPC coefficients defined above. For a detailed comparison, plot of the spectral distortion in dB with three different methods is depicted in Figure 2. We learn that Method 3 (solid line) gives the most preferred result due to the existence of the correlation between adjacent subframes.

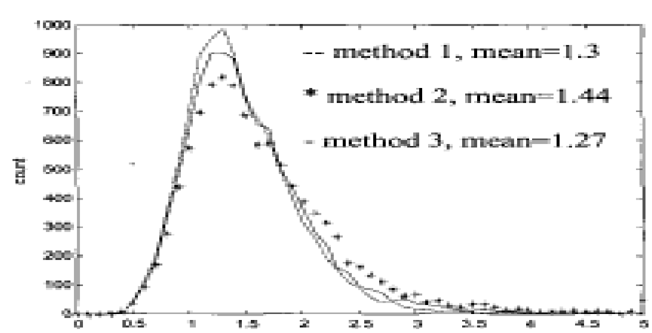

Figure 2:- Spectral distortion Histogram

III TRANSFORMATION OF SPEECH

Here we used a Pitch predictor to predict the quansi periodic signal of Speech. So pitch of longterm prediction is an important parameter for speech encoding. The fractional pitch resolution of G.729 is *1/3* while that of GSM is 1/6, 1/3 or 1/2.

Pitch search is divided in two part one open loop pitch search and closed loop pitch search. Pitch search required maximum calculation. How to utilize the received pitch information from the other coder to reduce the search complexity is the research concern. We are considering 40 samples for per subframe for both GSM and G.729 and for that the transformation process is done by one by one. Speech signal is differentiate in voice and unvoiced signal. Here we used pitch lag by G.729 which is found by GSM for reduce the pitch search in G.729 encoder.

IV TRANSFORMATION OF VCELP AND ACELP

VSELP i.e. Vector Sum Excited Linear Prediction and ACELP i.e. Algebraic Code Excited Linear Prediction both methods are used to determine the random excitation from residual speech after the speech estimation is done. The action and calculation perform by excitation is more time consuming. This all done by transcoding system so it is required that transcoding system should work consistently.

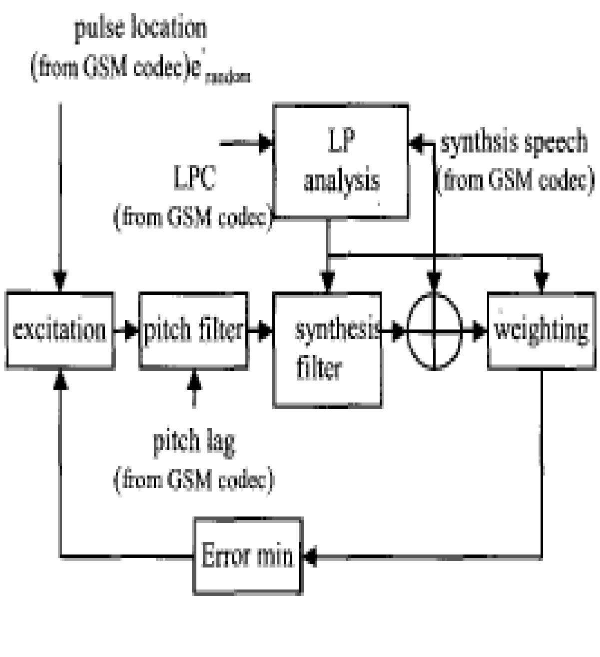

Figure 3:- block diagram of proposed method of G.729 encoder

The G.729 encoder method is shown in figure 3. The Linear Prediction (LP) coefficient and pitch lags are obtain by GSM encoder. The random excited signal of G.729 is expressed as

$$e'_{random}[n] \cong e_{pitch}[n] + e_{random}[n] - e'_{pitch}[n] \,,$$

Where $e_{pitch}[n]$ are stand for adaptive excitation and $e_{random}[n]$ for random excitation and it is obtain by GSM coding parameter and $e^{\wedge}_{pitch}[n]$ is the pitch excitation which is obtained from transformation of speech (2.2). Now, we will find pulses $\bar{x}$ from $e_{random}[n]$ that let the algorithm

$$\left| Y - H\bar{x} \right| = 0 \,,$$

Here Y is the target signal from G.729 encoder and H is the synthesis filter. For proper solution it required a very large computation. We required increasing the speed of search procedure by reducing number of pulse position combination. Thus, we calculate the energy of target signal per track as

$$E_i = \sum_{j=0}^{7} [y(i+5j)]^2 \,, \qquad i = 0,\ldots,4$$

$$\Delta = (E_{max} - E_{min}) / 4$$

$$m_i = (E_i - ave(E_i)) / \Delta, \quad i = 0,\ldots,4$$

$$n_i = m_1 + m_i \,, \qquad i = 0,\ldots,4$$

where ***n,*** is the pulse number per track, m,is the initial value. We use the excitation signal from the GSM to find the search range. We will pick up ***n,*** leading pulses from GSM excited signals and put them into G.729 system to search the suboptimal pulses. The pulse location is more important when the energy is higher. So, we can take more pulses in track.

V CONCLUSION

In this paper we discuss the transcoding between two different speech coding system. We used the parameter from GSM and reduce the search complexity in G.729. We have done computation reduction over DTE transcoding.Thus the proposed algoritm can be used for wireless mobile in world wide

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