Data Transmission Method based on Single Carrier over GSM Voice Channel

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Abstract
Aiming at the problem that the security mechanism of GSM system can not realize end-to-end security communication, a data transmission scheme and optimization method based on the single carrier is proposed. A speech-like signal is designed based on single carrier firstly. Then a two-step of channel estimation and compensation are employed at the receiver. A simulation platform of data transmission over GSM voice channel is built which shows that in AMR with bit rates 12.2 kbps (AMR 12.2), the error rate after compensation can reach about 10^{-4}, and the proposed method can achieve significant performance improvement in data transmission over GSM voice channel with different AMR bit rates.

Keywords: Single carrier; GSM voice channel; data transmission;

1. INTRODUCTION

Global System for Mobile Communications (GSM) is widely used and has more than 2 billion subscribers worldwide. The voice channel of GSM still protects its popularity with recent consumer applications because of its wide service availability in all over the world. Due to the special needs of the real-time voice, at present, Circuit Switched FallBack (CSFB) is also adopted which allows voice and SMS services deliver to LTE devices through the use of GSM. Besides, network coverage of GSM is better than 3G and 4G network. Therefore, GSM technology still plays a key role because of its availability, reliability and robustness.

However, as an early communication standard, there are a lot of leaks in existing security mechanism, and information security issues such as information leakage, mobile phone eavesdropping become serious day by day. Therefore, end-to-end communication security for GSM network becomes more and more urgent. The mechanism in is under the assumption that the core network is safe and reliable, and is not perfect. In addition, the encryption algorithms in GSM system are unreliable and easy to crack. Therefore, the GSM network can not ensure users’ secure end-to-end communication. In some special applications such as finance, military applications, the users need end-to-end secure communications. Recently, additional secure applications are proposed in which GSM is used as a data transmission service, some end-to-end secure communication products for GSM network has been developed. The encryption products nowadays are based on the data channel of GSM network, therefore, the main problem is that the end-to-end delay is large which means impossible for real-time voice communications, another problem is in some remote areas, users can not access the data network.

Except for data channels, GSM voice channels can also carry out data transmission to achieve end-to-end secure communication, with the advantages of being less delayed and more flexible. Data transmission over GSM voice channel (DoGSMV) is suggested for many other applications recently. It is proposed to be used in Point of Sale (POS) payment systems and wireless automatic teller machines (ATM). Unfortunately sending data via GSM channel is a challenging task since it is speech sensitive and suppresses other forms of signals, and the main problem of this kind of encrypted speech technology is the signal distortion effect caused by the voice coder. Regular Pulse Excitation - Long Term Prediction, RPE-LTP is a first technology to be used with coding Rate 13 Kbps, every 20 ms a voice frame, the scheme is also known as Full Rate (FR) speech coder which was standardized in 1998. On the basis of FR, Enhanced Full Rate (EFR) is developed which achieves better voice quality without changing the coding rate, meanwhile, The half rate speech coder (HR) standard was established to cope with the increasing number of subscribers, it is introduced which doubles system capacity with the cost of the voice quality. In 1998, 3GPP also introduced the adaptive multi rate encoding (Adaptive Multi-Rate AMR) as speech enhancement encoding, its core idea is to adjust voice encoding mode based on the signal quality of GSM air interface. AMR has the better voice quality which becomes the main commercial vocoder standard. The vocoders above compress voice according to the phonetic characteristics, and therefore, the vocoders will change the amplitude and phase of the modulation signal which makes the conventional digital transmission schemes can not work properly.

Therefore, we need to design some new data transmission methods. Codebook transfer methods are developed in (Ladue, Sapozhnykov, Fienberg, 2008; Shahbazi, Rezaei, Sayadiyan, et al. 2010). Speech-Like (SL) symbols are designed to be capable of passing through such a compressed voice channel, so these methods utilize a set of predefined symbols, however, the disadvantage of codebook methods are massive storage pressure. Another methods are through the construction of special signal, which has the ability to penetrate the
vocoder, Frequency-shift Keying (FSK), Quadrature Amplitude Modulation (QAM) and Orthogonal Frequency Division Multiplexing (OFDM) technology are explored in (Ali, Baudoin and Venard.2013; Chmayssani, Baudoin and Hendryckx, 2008; Chen 2007) respectively for application in the GSM voice channel.

But such methods do not take into account the non-ideal factors of vocoder channel. This paper presents a data transmission method based on single carrier over GSM voice channel. In this method, we design the frame format and transmission parameters with speech-like characteristics. Consider the non-ideal factors of signal distortion caused by the vocoder, we apply equalization methods to compensate the vocoder channel so as to realize reliable data transmission over the GSM voice channel.

This paper is organized as follows. Section II presents the General description of the GSM voice decoding/encoding procedure. The system architecture based on single carrier over GSM voice channel is presented in section III. Consider the non-ideal factors of signal distortion caused by vocoder, a two-step of channel estimation and compensation are employed in section IV so as to realize reliable data transmission over the GSM voice channel. Section V demonstrates the performance of the proposed method with illustrative simulation result. Conclusions are drawn in section VI.

2. GSM VOICE CODEC

AMR was the first widely adopted variable bit rate codecs in the GSM networks. Because of its higher voice quality on similar bit rates like FR and EFR, and owing to its flexibility to adapt the speed when the network is congested, it is the most widely used codec in both GSM and UMTS. Because there are bit errors over radio interface, AMR adds redundant information to the stream, making strong improvements to the voice quality.

This method is based on code excited linear predictive coding, and uses full pole linear filter to simulate speech characteristics. In the AMR vocoder, 20 ms of speech is considered as a frame, each frame is divided into four 5 ms sub-frame. It calculate autocorrelation using 30ms asymmetric window, the autocorrelation results of windowing speech are converted to the linear prediction (LP) coefficients using Levinson-Durbin algorithm and then to line spectral pairs. An open-loop pitch analysis is then performed to find pitch delay for each frame. These delays are fed into the speech synthesizer, and the optimal algebraic codebook is searched according to the minimum error criterion, and the final synthesized speech is sent to the GSM physical layer for transmission.

The corresponding decoder functions include two parts, one part is the decoding of transmit parameters, such as LP parameters, adaptive codebook vector, adaptive codebook gain, fixed codebook vector and fixed codebook gain. The other part is the speech reconstruction.

To sum up, AMR is a parameter encoding method, because of the vocoder’s nonlinear nature and memory, it is impossible to represent it by an analytic transfer function.

3. SYSTEM ARCHITECTURE

The architecture of GSM voice channel is shown in Fig.1.

![Figure 1. Architecture of GSM voice channel](image)

It can be seen that when a end-to-end speech communication is initiated at one terminal, the original speech signal will be encoded firstly in voice encoder at the terminal, through uplink transmission, the signal is decoded at the base station. For downlink transmission, the signal is encoded in the base station for the second time, finally, the signal received in the other terminal will be decoded for the second time. Therefore, data
transmission based on single carrier needs through voice codec twice which makes the original signal has a certain degree of distortion, and because the memory of the AMR codec, if the transmitted signal is not appropriate, the distortion will be expanded and spread. In addition to that, another cause of the error for data transmission depends on the factor of GSM Channel’s instantaneous wireless link quality.

In this paper, a single-carrier modulation scheme is designed, so as to ensure that the distortion in the transmission is as small as possible. The data transmission system based on single carrier over GSM voice channel is shown in Fig.2.

![Data transmission system based on single carrier over GSM voice channel](image)

Figure 2. Data transmission system based on single carrier over GSM voice channel

Considering that vocoder have a certain memory, the whole GSM speech channel can be modeled as follows:

\[ y = V(s_i, \varphi) \]

where \( s_i \) is modulated symbol, \( i \) is the transmitted decimal word, \( y \) is received signal through vocoder and air interface, \( \varphi = \phi(s_{i-1}, ..., s_{i-k}) \), \( \phi \) is vocoder state which depends on all previous inputs. Thus, the symbol-to-data-detection process can be viewed as the a posteriori probability distribution maximization.

\[ \hat{i} = \arg \max_{i} \{P(s_i \mid y)\} \]

3.1. Speech-like signal design

In order to ensure the reliability of the transmission channel, one of the key points is to determine the best available frequency bands. The amplitude frequency response of the voice channel is shown in fig.3.

![Amplitude-frequency response after twice AMR codecs](image)

Figure 3. Amplitude-frequency response after twice AMR codecs

As shown in the figure, the 200~3000Hz band range is relatively flat, so this frequency band can be used for data transmission.

Another key point is the choice of modulation technique, (Chmayssani and Baudoin, 2009) shows that phase modulation is possible to realize over memoryful codecs in which digital information is transmitted through discrete phase changes of a carrier wave, the number of discrete phases used across the symbol space determines the number of unique symbols. Since the phase of the waveform changes for every single sinusoid that is transmitted, the raised cosine filter need to be employed so as to decrease discontinuities in the time domain signal.
3.2. Implementation for single carrier modulation

Different from traditional communication system, in the transmission of GSM voice channel, the multipath problems have been solved at the bottom layer of the voice channel. As discussed in this paper based on single carrier voice encryption, we can assume the multipath effects are very small, and therefore, in the design of frame format, the prefix in the traditional communications are omitted. The main parameters are shown in Table 1, and the corresponding parameters have a transmission rate of 1333bps.

<table>
<thead>
<tr>
<th>parameters</th>
<th>value</th>
</tr>
</thead>
<tbody>
<tr>
<td>center frequency</td>
<td>2500Hz (QPSK)</td>
</tr>
<tr>
<td>transmission bands</td>
<td>2000Hz (BPSK)</td>
</tr>
<tr>
<td></td>
<td>1800Hz (BPSK)</td>
</tr>
<tr>
<td></td>
<td>900Hz (QPSK)</td>
</tr>
<tr>
<td>sample rate</td>
<td>8000Hz</td>
</tr>
<tr>
<td>roll-off factor</td>
<td>raised-cosine (0.35)</td>
</tr>
<tr>
<td>modulation</td>
<td>BPSK, QPSK</td>
</tr>
</tbody>
</table>

4. GSM VOICE CHANNEL ESTIMATION AND COMPENSATION

As mentioned earlier, the GSM voice channel causes the received signal distortion which is nonlinear. To overcome this problem, in this paper, the initial phase compensation combined with adaptive filter are used for signal equalization.

4.1. Initial phase compensation

When employing QPSK modulation, constellation of continuous received signals are shown in fig.4.

\[ Y = HP + W \]  

\[ \hat{H} = \min\{(Y - HP)^H(Y - HP)\} \]
In the initial compensation stage, only the phase compensation is needed, assume \( \hat{\theta}(\hat{H}) \) is the estimated phase frequency response, the average phase offset is obtained:

\[
\hat{Q} = \hat{\theta}(\hat{H})
\]  
(5)

The data streams can be equalized as:

\[
x = s^{-j\hat{Q}}
\]  
(6)

The constellation after initial phase compensation is shown in Fig.5.

![Scatter plot](image)

**Figure 5.** Constellation after initial phase compensation

### 4.2. Adaptive filter

Initial phase compensation is suitable for time invariant speech channel, however, if the voice communication channel response varies with time, it is necessary to apply adaptive filter for further compensation. In this paper, the constant modulus algorithm (CMA) algorithm is employed for weight adaptation.

Let \( \{x_k\} \) be the received signal after initial phase compensation, the output of adaptive filter is \( \{y_k\} \), \( y_k = W^T X_k \), where \( W^T \) is the weights of adaptive filter, \( X_k = \{x_{k-l+1}, ..., x_k, x_{k+l-1}\} \), \( 2l-1 \) is the order of the filter, the objective function of CMA can be expressed as:

\[
J = E(\|y_k\|^2 - g)
\]  
(7)

where \( g = \frac{E(\|r_k\|^2)}{E(\|r_k\|^4)} \), \( E(\cdot) \) represents expectation and can be replaced by average value. Our goal is to minimize the objective function.

The weight iteration formula is given:

\[
W_{k+1} = W_k + \mu X_k^* y_k (\|y_k\|^2 - g)
\]  
(8)

where \( \mu \) is a step factor. The CMA algorithm above is only for real symbols such as BPSK, when using QPSK modulation, the real and imaginary parts of the symbols can be equalized separately.

### 5. SIMULATIONS

As mentioned above, the end-to-end voice transmission channel includes two voice codecs. In order to verify the effectiveness and reliability of the proposed scheme, we built a simulation platform shown in Fig.6.
Table 2 shows the bit error rate before and after compensation methods proposed in this paper. After compensation, the BER performance is improved obviously. For the commonly used AMR with bit rates 12.2 kbps (AMR 12.2), the error rate after compensation can reach about $10^{-4}$. With the decrease of bit rates, the transmission performance of the system becomes worse gradually. This is because the signal distortion caused by the GSM voice channel is more serious with the increase of speech compression. In addition, as shown in Table 2 and table 3, BPSK outperforms QPSK in AMR 12.2, but is less robust at other bit rates.

Table 2. Performance of BPSK

<table>
<thead>
<tr>
<th>AMR bit rates</th>
<th>BER before compensation</th>
<th>BER after compensation</th>
</tr>
</thead>
<tbody>
<tr>
<td>AMR 12.2</td>
<td>4.6e-4</td>
<td>1.5e-4</td>
</tr>
<tr>
<td>AMR 10.2</td>
<td>4.13e-2</td>
<td>3.3e-2</td>
</tr>
<tr>
<td>AMR 7.4</td>
<td>1.2e-1</td>
<td>6.35e-2</td>
</tr>
</tbody>
</table>

Table 3. Performance of QPSK

<table>
<thead>
<tr>
<th>AMR bit rates</th>
<th>BER before compensation</th>
<th>BER after compensation</th>
</tr>
</thead>
<tbody>
<tr>
<td>AMR 12.2</td>
<td>3.6e-3</td>
<td>5.5e-4</td>
</tr>
<tr>
<td>AMR 10.2</td>
<td>1.2e-2</td>
<td>6.4e-3</td>
</tr>
<tr>
<td>AMR 7.4</td>
<td>1.0e-1</td>
<td>6.8e-2</td>
</tr>
</tbody>
</table>

6. CONCLUSIONS

In view of the demand of GSM speech encryption, a data transmission scheme and optimization method based on single carrier are proposed. Firstly, we design the speech-like signal based on single carrier. Secondly, taking into account the signal distortion caused by the voice codec, a two-step channel estimation and compensation are employed at the receiver. In order to verify the effectiveness and reliability of the proposed optimization method, a data transmission simulation platform over GSM voice channel is built. The BER performances of QPSK and BPSK modulation schemes with different AMR bit rates are tested respectively. Simulation results show that the proposed method can achieve significant performance improvement in data transmission over GSM voice channel with different AMR bit rates.

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