2G, 3G Network Planning and Optimization...

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1.3 Data Transmission

Radio channel has totally different characteristics from wired channel. Radio channel has a strong time-varying characteristic. It has a high error rate when the signal is influenced by interferences, multipath fading, or shadow fading. In order to solve these problems, it is necessary to protect the signals through a series of transformation and inverse transformation from original subscriber data or signaling data to the information carried by radio wave and then to subscriber data or signaling data. These transformations include channel coding and decoding, interleaving and de-interleaving, burst formatting, encryption and decryption, modulation and demodulation.

1.3.1 Voice Coding

Modern digital communication system usually uses voice compression technology. GSM takes tone and noise from human throat as well as the mouth and tongue filter effect of acoustics as voice encoder to establish a model. The model parameters transmit through TCH channel.
Voice encoder is based on residual excited linear prediction encoder (REIP) and its compression effect is strengthened through long term predictor (LTP). LTP improves residual data encoding by removing the vowel part of voice.
Voice encoder divides voice into several 20 ms voice blocks and samples each block with 8 kHz, so each block has 160 samples. Each sample is quantified through frequency A 13 bits (frequency μ 14 bits). Since the compression rates of frequency A and frequency μ are different, add three and two "0" bits to the quantification values respectively, and then each sample gets 16 bits quantification value. Therefore, 128 Kbit/s data flow is obtained after digitizing but before encoding. This data flow is too fast to transmit in radio path and has to be compressed in encoder. With full speed encoder, each voice block is encoded into 260 bits to form a 13 Kbit/s source coding rate. Next is channel coding. With 20 ms as a unit, 260 bits are output after compression encoding, so the encoding rate is 13Kbit /s.
Compared with the direct coding transmission of voice in traditional PCM channel, the 13kbps voice rate of GSM is much lower. More advance voice encoder can reduce the rate to 6.5kbps (half rate encoding).

1.3.2 Channel Coding

Channel coding is used to improve transmission quality and remove the influence of interferential factors on signals at the price of increasing bits and information. The basic way of coding is adding some redundant information to the original data. The added data is calculated on the basis of original data with certain rules. The decoding process of receiving end is judging and correcting errors with this redundant bit. If the redundant bit of received data calculated with the same way is different from the received redundant bit, errors must have occurred in transmission. Different code is used in different transmission mode. In practice, several coding schemes are always combined together. Common coding schemes include block convolutional code, error correcting cyclic code and parity code.
In GSM, each logical channel has its own coding and interleaving mode, but the principle is trying to form a unified coding structure.
Encode information bit into a unified block code consisting of information bits and parity check bits.
Encode block code into convolutional code and form coding bits (usually 456 bits).
Reassemble and interleave coding bits and add a stealing flag to form interleaving bits.
All these operations are based on block. The block size depends on channel type. After channel coding, all channels (except RACH and SCH) are made of 464-bit block, that is, 456 coded information bits plus 8-bit header (header is used to distinguish TCH and FACCH). Then these blocks are reinterleaved (concerning channel).
In TCH/F voice service; this block carries one speech frame of information. In control channel, this block usually carries one piece of information. In TCH/H voice service, speech information is transmitted by a block of 228 coded bits block.
For FACCH, each block of 456 coded information bits is divided into eight sub blocks. The first four sub blocks are transmitted by even bits of the four timeslots borrowed from the continuous frames of TCH, and the rest four sub blocks borrows odd bits of the four timeslots from the four continuous frames delayed for two or four frames after the first frame. Each 456 coded bit block has a stealing flag (8 bits), indicating whether the block belongs to TCH or to FACCH. In the case of SACCH, BCCH or CCCH, this stealing flag is dummy.
The synchronous information in Downlink SCH and the random access information in uplink use short coded bit blocks transmitted in the same timeslot.
In TCH/F, a 20ms speech frame is encoded into 456-bit code sequence. The 260 bits of the 13 Kbit/s 20ms speech frame can be divided into three categories: 50 most import bits, 132 important bits and 78 unimportant bits. Add 3 parity check bits to the 50 most important bits, and these 53 bits together with 132 important bits and 4 tail bits are convolutionally encoded (with 1/2 convolutional coding rate) into 378 bits, plus the 78 unimportant bits, and the 456 bits code sequence is obtained.
In BCCH, PCH, AGCH, SDCCH, FACCH and SACCH, data is transmitted by Link Access Procedure on the Dm channel (LAPDm). Each LAPDm frame has 184 bits, together with 40 bits error correcting cyclic code and 4 tail bits, through 1/2 convolutional coding rate, and the 456 bits code sequence is obtained.
Each SCH contains 25-bit message field. Among them, 19 bits are frame number and 6 bits are BSC number. These 25 bits plus 10 parity check bits and 4 tail bits are 39 bits. Through 1/2 rate convolutional coding, 78 bits are obtained, which occupy an entire SCH burst. .
RACH message only has 8 bits, including 3-bit setup cause message and 5-bit discrimination symbol. On the basis of these 8 bits, add 6 bits of color code (obtained through the MOD 2 of the 6-bit BSIC and 6-bit parity check code), plus 4 tail bits to get 18 bits. Through 1/2 rate convolutional coding, 36 bits are obtained, which occupy an entire RACH burst. 。

1.3.3 Interleaving

If speech signal is modulated and transmitted directly after channel coding, due to parametric variation

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of mobile communication channel, the long trough of deep feeding will affect the succeeding bits, leading to error bit strings. That is to say, after coding, speech signal turns into sequential frames, while in transmission, error bits usually occur suddenly, which will affect the accuracy of continuous frames. Channel coding only works for detection and correction of signal error or short error string. Therefore, it is hoped to find a way to separate the continuous bits in a message, that is, to transmit the continuous bits in a discontinuous mode so as to change the error channel into discrete channel. Therefore, even if an error occurs, it is only about a single or very short bit stream and will not interrupt the decoding of the entire burst or even the entire information block. Channel coding will correct the error bit under such circumstances. This method is called interleaving technology. Interleaving technology is the most effective code grouping method to separate error codes.
The essence of interleaving is to disperse the b bits into n bursts in order to change the adjacent relationship between bits. Greater n value leads to better transmission performance but longer transmission delay. Therefore, these two factors must be considered in interleaving. Interleaving is always related to the use of channel. GSM adopts secondary interleaving method.
After channel coding, The 456 bits are divided into eight groups; each group contains 57 bits. This is the first interleaving, also called internal interleaving. After first interleaving, the continuity of information in a group is broken. As one burst contains two groups of 57-bit voice information, if the two-group 57 bits of a 20 ms voice block after first interleaving are inserted to the same burst, the loss of this burst will lead to 25% loss of bits for this 20 ms voice block. Channel coding cannot restore so much loss. Therefore, a secondary interleaving, also called inter-block interleaving, is required between two voice blocks.

After internal interleaving, the 456 bits of a voice block B are divided into eight groups. Interleave the first four groups of voice block B (B0, B1, B2, and B3) with the last four groups of voice block A (A4, A5, A6, and A6), and then (BO, A4), (B1, A5), (B2, A6), and (B3, A7) form four bursts. In order to break the consistency of bits, put block A at even position and block B at odd position of bursts, that is, to put B0 at odd position and A4 at even position. Similarly, interleave the last four groups of block B with the first four groups of block C.
Therefore, a 20 ms speech frame is inserted into eight normal bursts after secondary interleaving. Theses eight bursts are transmitted one by one, so the loss of one burst only affects 12.5% voice bits. In addition, as these bursts have no relations with each other, they can be corrected by channel coding.
The secondary interleaving of control channel (SACCH, FACCH, SDCCH, BCCH, PCH, or AGCH) is different from voice interleaving which requires three voice blocks. The 456-bit voice block is divided into eight groups after internal interleaving (the same as that of voice block), and then the first four groups are interleaved with the last four groups (the same interleaving method as that of voice block) to get four bursts.
Interleaving is an effective way to avoid interference, but it has a long delay. In the transmission of a 20 ms voice block, the delay period is (9*8)-7=65 bursts (SACCH occupying one burst), which is 37.5 ms. Therefore, MS and trunk circuit have echo cancellers added to remove the echo due to delay.

1.3.4 Encryption

Security is a very important feature in digital transmission system. GSM provides high security through transmission encryption. This kind of encryption can be used in voice, user data, and signaling. It is used for normal burst only and has nothing to do with data type.
Encryption is achieved by XOR operation of poison random sequence (generated through A5 algorithm of encryption key Kc and frame number) and the 114 information bits of normal burst.
The same poison random sequence generated at receiving end and the received encryption sequence together produce the required data after XOR operation

1.3.5 Modulation and Demodulation

Modulation and demodulation is the last step of signal processing. GSM modulation adopts GMSK technology with BT being 0.3 at the speed of 270.833 Kbit/s and Viterbi algorithm. The function of modulation is to add a certain feature to electromagnetic wave according to the rules. This feature is the data to transmit. In GSM, the phase of electromagnetic field bears the information.
The function of demodulation is to receive signals and restore the data in a modulated electromagnetic wave. A binary numeral has to be changed into a low-frequency modulated signal first, and then into an electromagnetic wave. Demodulation is the reverse process of modulation.
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