How to Restrict Voice Quality Degradation In Digital GSM Network

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ABSTRACT
Mobile communication existed about a half century ago. It was developed in the 1980s. The main objective of mobile communication is to realize communication among the communicating objects at any time and any place. Currently the number of global mobile subscribers exceeded six billions. From here you can take an idea how the wireless communication has greater development potential than the wired communication. From the communication network point of view mobile communication network is an extension of wired network. The wireless communication network consists of wireless and wired part. The wireless part provides the access of user terminal and uses limited frequency resources to transmit voice and data reliably over the air interface. The wired part completes the network functions like switching, roaming, authentication and user management.

In order to communicate with each other the quality of voice is the basic demand of subscriber from the service provider. Another important factor is cost; in this competitive market each cellular mobile service provider is using an optimizer in order to compress the transmission media and also compress the voice (Mean Opinion Score) by using codec G.711 to save the bandwidth and reduce the cost. By doing this the quality of voice degrades. In the result of this subscriber will suffer a bad quality voice. This research will discuss the solution that will not only reduce the cost from the transmission media point of view but also restrict the voice quality degradation.

Indexing terms/Keywords
Mobile communication, GSM voice codec, Mean Opinion Score for G.711 μ-law, PSQM testing, GSM voice quality improvement, G.711 voice codec, GSM digital network architecture and features.

Academic Discipline And Sub-Disciplines
Telecommunication, Wireless communication, 2 G networks

SUBJECT CLASSIFICATION
Mobile Communication

TYPE (METHOD/APPROACH)
Telecom Network Implementation in the telecommunication industry
INTRODUCTION

GSM is the primarily standard formulated for the communication system operating at 900 MHz in Europe and at 900 MHz, 1800 MHz, and 1900 MHz in Pakistan. On the basis of limited expansion capability of analog communication system, the GSM is developed based on the requirement of service capability addition.

FEATURES OF GSM SYSTEM

Below are the major features of GSM system:

1. Spectral Efficiency
2. Voice quality
3. Capacity enhancement
4. Open interface
5. Security
6. Roaming on SIM card based
7. Security
8. Interconnection with PSTN, ISDN & PDN

GSM NETWORK ARCHITECTURE

The Fig. 1 shows the GSM Network Architecture:

The GSM network consists of following components:

- BTS: Base Transceiver Station
- MS: Mobile Subscriber
- BSC: Base Station Controller
- TRAU: Transcoding & Rate Adaptation Unit
- MSC: Main Switching Center
- VLR: Visitor Location Register
- HLR: Home Location Register
- IWF: Interworking Function
- GMSC: Gateway MSC
- SMC: Short Message Center
- AUC: Authentication Center
- EIR: Equipment Identity Register
• PSTN: Public Switched Telephone Network
• ISDN: Integrated Services Digital Network
• PDN: Public Data Network

All the above components connected with each other to form a GSM network as shown in the Fig. 1.

**GSM NETWORK SUBSYSTEMS**

The components of GSM network constitutes following three subsystems:

1. NSS (Network Subsystem)
   - NSS consists of following functional entities:
     - MSC
     - VLR
     - HLR
     - EIR
     - AUC
     - SMC

2. BSS (Base Station Subsystem)
   - BSS consists of following functional components:
     - BSC
     - BTS

3. OMS (Operation & Maintenance Subsystem)
   - OMS consists of:
     - OMC system part (OMC-S)
     - OMC radio part (OMC-R)

**GSM FRAME STRUCTURE**

The frame structure of GSM network consists of five levels:

1. Time slot
2. TDMA frame
3. Multiframe
4. Superframe
5. Hyperframe

Time slot is the basic unit of physical channel. A TDMA frame consists of eight time slots. Multiframe are of two types, one consists of 26 TDMA frames whereas the second type consist of 51 TDAM frames. The superframe composed of consecutive 51 x 26 multiframes. And a hyperframe consists of 2,048 superframes.
PHYSICAL CHANNEL OF GSM SYSTEM

In digital GSM system each TDMA frame consists of eight time slots. Each time slot is called a physical channel. On these physical channels logical channels are mapped.

LOGICAL CHANNELS OF GSM SYSTEM

The content transmitted over the physical channel is the logical channel. These logical channels are related to control, speech and voice and data logical channels assigned on physical channels (time slots) of TDMA frame. The pulse train are the information transferred over physical channels by the digital GSM system. This pulse train consists of 100 modulation bits. This pulse train is also called Burst.

Below are the logical channels in GSM digital network:

- Common channel
- Broadcast channel (BCH)
- Common control channel (CCCH)
- Paging channel (PCH)
- Random access channel (RACH)
- Access grant channel (AGCH)
- Dedicated control channel (DCCH)
  1. Stand-alone dedicated control channel (SDCCH)
  2. Slow associated dedicated control channel (SACCH)
  3. Fast associated dedicated control channel (FACCH)
- Traffic channels (TCH)

SECURITY MEASURES IN GSM SYSTEM

There are security specific objectives in the GSM system. First is to protect network from unauthorized access and second is to protect the user privacy. In order to prevent the unauthorized access of network an authentication is adopted to check whether the inserted SIM card is consistent with the subscriber ID provided by the mobile service provider. The protection of user privacy is implemented through different means, in which the most common is transmission encryption that can prevent the interception of communication over the radio channel.

Encryption and decryption are the exclusive-OR operation (This is also known as A5 algorithm) of 114 radio burst pulse code bits and one 114 bit encryption sequence generated by a special algorithm. Hence, two different sequences are used over the uplink and downlink. One sequence is used for encryption inside the MS and meanwhile used as the decryption sequence in BTS and vice versa. This scenario is shown in the below figure:
VOICE QUALITY DEGRADATION ISSUE IN GSM SYSTEM

The testing of PSQM under the ideal condition yields Mean Opinion Scores of 4.45 for G.711 μ-law and 4.45 for G.711 A-law. Transcoder (TC) is equipment that basically used to complete the voice conversation between 16 kbit/s RPE-LTP code and 64 kbit/s A-law PCM. TC is located between the BSC & MSC (as shown in the Fig.4). When TC is located on the MSC side, the low voice encoding transmission rate can be fully utilized through the SM (sub multiplexer) and BIE (base station interface equipment) of the transmission line between MSC and BSC and also between the BSC and BTS to minimize the cost of transmission line.

But the problem is that when call flow passes through TC G.711 encoder will compress the voice quality up to 15% hence it will degrade the MOS (mean opinion score) from 4.45 to 3.78 (4.45 – 0.66 (15% of 4.45)). And when it will back from MSC to BSC it will again compress 15% more of that already compressed voice so it will degrade the MOS from 3.78 to 3.21 (3.78 – 0.56 (15% of 3.78)). It means here is degradation in the voice quality is of 30% instead of 15%. Due to this issue the subscriber experience a bad quality voice. The detail is shown in the below Fig.5 and Fig.6.
SOLUTION TO FIX VOICE DEGRADATION ISSUE

In order to restrict this degradation two time if we use DSP kit or FPGA hardware in which an algorithm/table has developed to store the first degrade value of MOS and tag an identity with that to remember that value. In this way after the first 15% degradation of MOS when back from MSC to BSC that DSP/FPGA kit will cross check that with its store value either it will be already compressed, if it matches the value then it will not allow this to degrade second time and send that to BSC in order to handle the call. By the use of such hardware we can restrict Voice quality to degrade two time. This phenomenon is clear from the below fig. 7.
CONCLUSION

For smooth voice communication it is concluded that if we introduce such a centralized hardware box (DSP/FPGA) in the network design between BSCs & MSCs as shown in the Fig. 7 of GSM digital system, we can restrict voice quality degradation and achieve a smooth level of voice communication between the subscribers. Such solution not only stable the network in term of voice quality but also increase the revenue in the form of increasing subscriber growth due to quality network.

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REFERENCES


Author’s biography

Kashif Faheem has received BSc. Engineering degree in Computer Information System Engineering (CISE) from KPK University Of Engineering & Technology, Peshawar, Pakistan (KPK UET Peshawar) in 2005. He had served as a O&M engineer for CDMA system from September 2005 to January 2005 in Telecard Pvt Ltd, Pakistan. From January 2006 to January 2008 had served as Planning Engineer in ZhongXing Telecom Pvt Ltd Pakistan (ZTE Pakistan a telecom vendor). Currently he is serving as a Manager Product in Pakistan Telecommunication Company Ltd (PTCL), a leading telecom operator in Pakistan from January 2008.