Simulation Study of DITMC Technique for Enhancing Channel Utilization in Speech Communication of Mobile Network

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***Abstract* — The scope of multiple uses of various signals in digital form in channel utilization enhancement became possible as a result of inclusion of data communication with speech in the integrated communication networks. Almost unsatisfying demand of a large number of applications for using the internet through mobile at a considerable higher speed has catalyzed the research on optimizing the channel utilization in mobile communication. A variety of methods for enhancing channel utilization and thereby economizing the use of costly channel bandwidth have already been proposed and attempted during the last four decades. As none of these methods could attempt to improve channel utilization upto 100% more efforts were being to achieve a higher channel utilization. This research paper proposes a simulation study of efficient Data Interleaving Technique in Mobile Communication (DITMC) which is expected to enhance the channel utilization as well as the talk time of the mobile handset (both CDMA and GSM) by 47.32%.**

Keywords — Channel Utilisation; Data Interleaving; DITMC; Speech Interpolation

I. INTRODUCTION

In 1960s, long distance communication media was copper only and the speech communication was increasing to an uncontrolled level. As the multiplexing levels were very low, many scientists and engineers starting thinking to various methods to enhance channel utilization. One of the research methods was to detect the silence periods in speech and use it for other purposes. As the technology available in that era was not meeting the practical requirements, the research was limited to theoretical levels only.

However, in mid 1980s, the optical communication using optical fiber became popular with very high bandwidth; the research in the field of enhancing channel utilization was almost stopped. In the middle of 1990s, the mobile communication in the form of cellular structure became popular, so that millions of subscribers could be given service simultaneously. But the wireless band width is limited and so again the research in the field of channel enhancement began [5].

With the inclusion of data communication with speech in the integrated communication networks, the scope of multiple uses of various signals in digital form in channel utilization enhancement became possible. Wireless networks inherently have far lower capacity than wire-line networks. The fiber optical cable can have greater data capacity than the entire RF spectrum. A shared, inherently unreliable medium like radio simply cannot match what wire can bring. And therein lies the problem [3]. Just a smaller number of mobile users with bandwidth-intensive applications can consume the available wireless network capacity. We are not quite at the stage of capacity exhaustion but we are seeing early instances of it. The analysis shows that the available capacity can be consumed by a relatively small percentage of high-bandwidth subscribers.

Based on current trends in mobile broadband usage, a spectrum-demand model developed by Rysavy Research [1] shows that many operators, spectrum could be consumed within three to five years. Except for the optical fiber, the available bandwidth of all contemporary communication media is restricted and therefore every endeavour to make optimum use of the communication channel sustains its undebatable importance. With the proliferation of telecommunications, it was realized that the major traffic was that of telephonic voice communication and the extent of possible channel utilization with voice signal was studied. It was established that the order of utilization is only of 36%, the remaining 64% being wasted [6,7] in a duplex channel. A variety of methods for enhancing channel utilization and thereby economizing the use of costly channel bandwidth have already been proposed and attempted during the last four decades. These included varieties of bit rate reduction techniques, signal compression techniques and speech interpolation techniques. As none of these methods singly could improve the channel utilization up-to 100%, efforts were made to use them in conjunction, which improved the utilization significantly although not reaching 100%.

II. PHILOSOPHY OF DATA INTERLEAVING TECHNIQUE IN MOBILE COMMUNICATION (DITMC)

The most important limitations of Mobile Communication are the limited energy storage of mobile handset with available technology and limited allowable bandwidth for both speech and data communication. Research is continuously going on to increase energy integration of batteries, decreased demand of energy during idle of handset and discovering / developing techniques of unlimited energy storage devices or continuous energy generation without any external source attachment as well as on various techniques employed for channel utilization

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enhancement. It lead to the development of Data Interleaving Technique [2]. The Data Interleaving Technique in mobile communication is actually based on removing the redundancy in speech bytes and the silence period bytes. Whenever the repetition of bytes is more than three, a Battery OFF Code is sent so as to indicate the receiver side about the repetition and when the repetition is over a Battery ON Code is sent to indicate the end of interleaving process and the normal bytes transmission continues. This method of data interleaving [2] possesses the essential attributes like speech delay of 2 samples i.e. only 250 μ seconds time, better channel utilization enhancement, no handover, suppression of the transmission of redundant (repetitive) message codes, making channel space available for additional data transmission, detection of much smaller duration speech pauses which are generally more frequent, negligible inbuilt delay in this system and is insignificant for real time applications. Unlike speech interpolation processes it does not make use of any speech detectors and the signal delay takes place at the originating node only and it is also independent of the network distance.

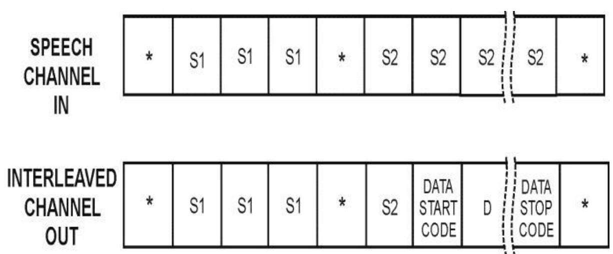

Fig. 1. Philosophy of Data Interleaving Symbols: S1, S2= Repetitive Speech Bytes*= Speech Byte (S) D= Data Byte (S) [8]

However, the two bytes delay in transmission is neither noticeable by the listener nor effects the operation of the system in any adverse way. Moreover the speech quality maintained by interleaved system is far superior to that of the systems using speech interpolation [4].

III. STATE TRANSITION DIAGRAMS FOR SIGNALTRANSMISSION AT VARIOUS NODES

The DITMC technique can easily be implemented in Mobile Communication/Networks with negligible delay incurred along the transmission path. Since the transmission of the signals may take a long path including a number of BTS's (Base Transceiver Stations) , BSC's (Base Station Controllers) and MSC's (Mobile Switching centers) the state transition diagrams for each case is considered. The philosophy of data interleaving can be easily analyzed by considering various states along the transmission of the interleaved signal.

The state transition diagram for generating and transmitting the signal at the MS is shown in Fig. 2. At this MS (Mobile Station) side the redundancy in voice samples or silence period is detected and decision is made about sending the BATTERY OFF, BATTERY ON code and sending the voice sample itself.

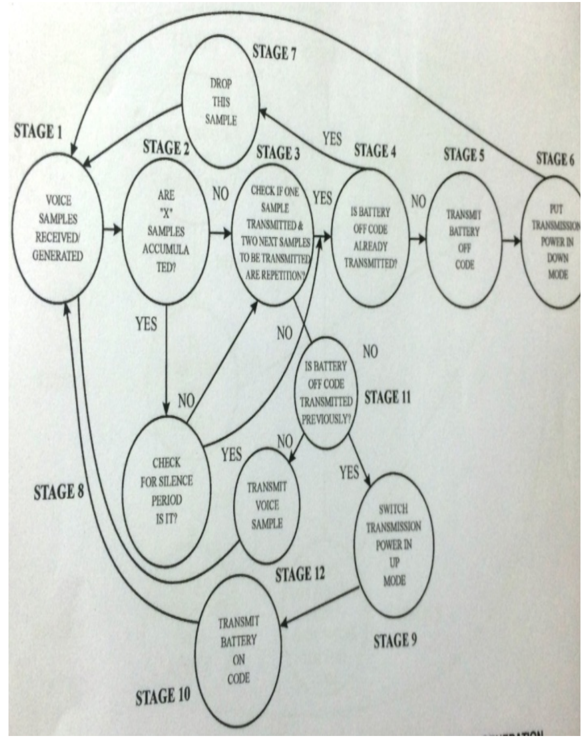

Fig. 2. State Transition Diagram for Signal Generation and Transmission at Mobile Station.[8]

There are twelve stages (Stage 1 to Stage 12) to perform this operation as shown in the state transition diagram of Fig.2. The interleaved signal generated or received from the Mobile Station has to be sent to the next node (BTS). It can be performed with the help of eight stages (Stage 1 to Stage 8) shown in the state transition diagram of Fig.3.

Initially the bytes are received from the mobile handset or in the time slot so achieved and it is checked whether the BATTERY OFF code is previously received or not. If it is received then it is checked that whether it is BATTERY ON code or not. If it is not then BATTERY OFF code status is cleared and it is sent to next node (BTS) and the process continues to receive byte from Mobile handset or time slot achieved. The byte(s) received from the previous node (BTS) has to be transmitted through another node(s) effectively indicating the nodes about the interleaved or non-interleaved information so that process of replication of bytes (in the case of continuous repetition of bytes) can be done easily. This process, of transmitting bytes from one node to another, can be performed with the help of nine stages (Stage 1 to Stage 9) shown in the state transition diagram of Fig.4. As the signal transmission continues in cascaded nodes (BTSs) with continuously indicating the intermediates nodes about the interleaving or non-interleaving information, it becomes of utmost importance that the same information is also conveyed to the Mobile station at the distant side i.e. if there is interleaving incurred then proper de-interleaving should also be done. It can be performed with the help of nine stages (Stage 1 to Stage 9) shown in the state transition diagram of Fig.5.

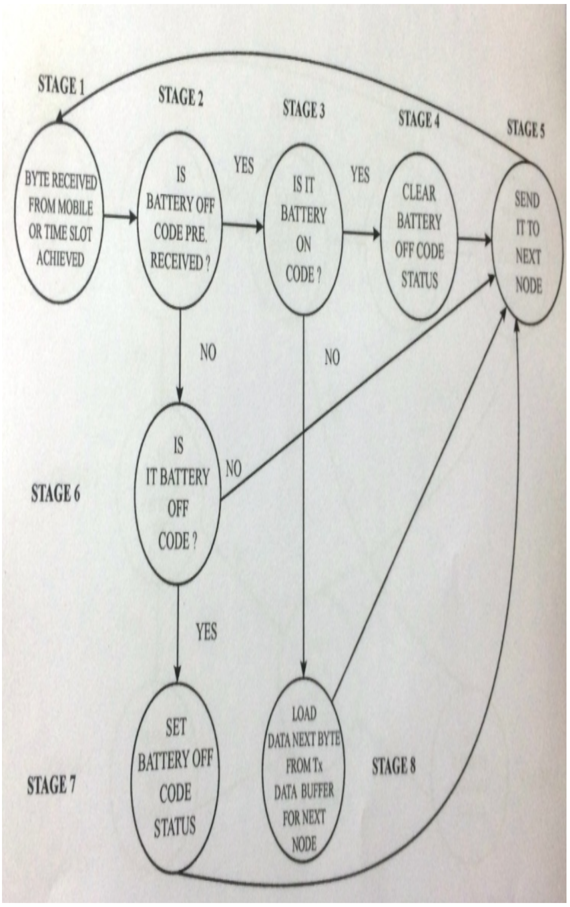

Fig. 3 State Transition Diagram for Interleaved Signal Transmission to Next Node (BTS) Received from Mobile Station at a Local Node (BTS).[8]

First of all the bytes are received from the last node (BTS) and it is checked whether the BATTERY OFF code is received previously. If the BATTERY OFF code is received previously then it is checked for BATTERY ON code and if it is BATTERY ON code then BATTERY OFF code status is cleared. After this, the byte is copied as, say "Z", and sent as voice byte to the local codec. Otherwise (in case it is not BATTERY ON code) the received byte is copied as, say "Z", and sent as voice byte to the local codec.

If the byte from last BTS is received and the BATTERY OFF code is not received previously then the received byte is checked for BATTERY OFF code. If it is BATTERY OFF code then BATTERY OFF code status is SET then the received byte is copied as, say "Z", and sent as voice byte to the local codec. Otherwise (in case the received byte is not BATTERY OFF code) the received byte is stored on previous byte as "Z" and sent as voice byte to the local codec. And the process continues to receive byte again from last node (BTS).

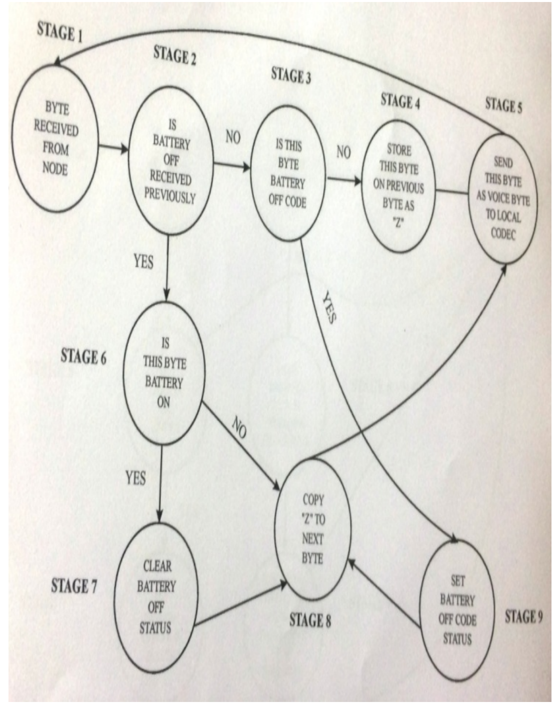

Fig. 4 State Transition Diagram for Signal Transmission Received from Previous Node to Next Node at Intermediate Node (s) [8]

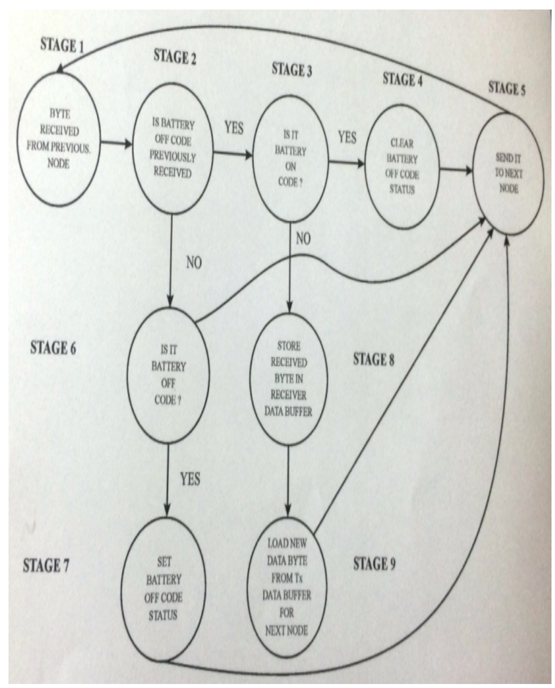

Fig. 5 State Transition Diagram for Signal Reception from Last Node (BTS) to Mobile Station and Its De-Interleaving [8]

IV. EXPERIMENTAL RESULTS

The program for simulation was designed in JAVA (Eclipse) software in which the WAVE sound files were taken as input so as to find out the total bytes in the file (file size in bytes), percentage of bytes that can be interleaved and percentage of overhead bytes. The program, for the simulation of voice/speech samples, aims at reading the input .wav files and then its configured contents are read. After its configuration has been analyzed the total number of bytes in the file and number of bytes per frame (in “t” time period) are calculated.

Then mean square value of all the bytes in frame of this time was calculated. For this, initially the threshold is decided to be a non integer value say 100 and is made adaptive by varying it with a step size of 0.5%. Then the difference of the calculated mean square value and threshold is compared so as to detect whether the bytes indicate the speech signal or silence period. Then the percentage of bytes, that can be interleaved, is calculated followed by the calculation of overhead bytes associated with this process of interleaving.

The simulation was performed on 120 speech samples of different durations, different speakers and of different languages so as to find out the percentage of interleaving and overhead incurred with the interleaving process. The speech samples were recorded (one way) through sound recorder on laptop. The parameters taken in the simulation were total length of the sound file (in bytes), total bytes that can be interleaved, percentage of interleaving, overhead bytes. The results show that a considerable amount of time (channel or bandwidth indeed) is wasted during the conversation in silence period as well as in the repetition of speech/voice bytes. The software demonstrates the channel utilization by considering both of the above-said factors.

Total number of bytes in the speech sample for 120 samples total bytes=2723319872 (2.536GB)

Total number of bytes, which can be interleaved =1288768358 (1.2 GB)

Data Interleaved in Voice = (1288768358/2723319872) x 100 % = 47. 32 %

Overhead involved in Interleaving Process = 0.034394%

Channel utilization without interleaving = 36%

Therefore, channel utilization with interleaving = 36+47.32=83.32%

V. CONCLUSIONS

The overall channel utilization, with Data Interleaving Technique in Mobile Communication, becomes 83.32% from the original 36% (without using any other channel utilization enhancement technique). Consequently it will affect the enhancement in energy optimization of the battery used in mobile handset by the same order i.e. 47.32 %. So the talk time of the mobile handset can be increased by 47.32% while standby time remaining the same. It means that if a mobile handset would have a talk time of 10 hours then by employing DITMC technique it is expected to be 14.73 hours. Consequently it will also be successful in resolving the problems of unwanted call termination due to discharging of the battery.

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