IOSR Journal of Electronics and Communication Engineering (IOSR-JECE)
e-ISSN: 2278-2834,p- ISSN: 2278-8735.Volume 12, Issue 5, Ver. II (Sep.- Oct. 2017), PP 36-40
www.iosrjournals.org

Bandwidth Saving on ABIS Interface When GSM's Feature DTX is Enabled

Ankit Singhal[1], Neeraj Jain[2]
[1](*Electronics & Communication Department, Modern Institute of Technology and Research Centre, India,*
[2](*Electronics & Communication Department, Modern Institute of Technology and Research Centre, India,*
Corresponding Author: Ankit Singhal

Abstract: *This paper comprehends the analysis of Bandwidth saving on ABIS interface when the DTX feature is enable. This paper comprehends the analysis of RTP packets on Wireshark, when DTX is enable. As Global System of Mobile (GSM) with its feature DTX (Discontinuous Transmission) have recently commanded the attention of many researchers. Discontinuous transmission (DTX) is a method of momentarily powering-down, or muting, a mobile or portable wireless telephone set when there is no voice input to the set. This optimizes the overall efficiency of a wireless voice communications system and save Bandwidth on Abis Interface. In a typical two-way conversation, each individual speaks slightly less than half of the time. If the transmitter signal is switched on only during periods of voice input, the duty cycle of the telephone set can be cut to less than 50 percent. This conserves battery power, eases the workload of the components in the transmitter amplifier's, and frees the channel so that time-division multiplexing (TDM) can take advantage of the available bandwidth by sharing the channel with other signals.*

Keywords: *Discontinuous Transmission (DTX), Global System of Mobile (GSM), Wireshark.*

Date of Submission: 25-10-2017 Date of acceptance: 07-11-2017

I. Introduction

The Global System for Mobile communications is a digital cellular communications system. It was developed in order to create a common European mobile telephone standard but it has been rapidly accepted worldwide. GSM was designed to be compatible with ISDN services.

The idea of cell-based mobile radio systems appeared at Bell Laboratories (in USA) in the early 1970s. However, mobile cellular systems were not introduced for commercial use until the 1980s. During the early 1980s, analog cellular telephone systems experienced a very rapid growth in Europe, particularly in Scandinavia and the United Kingdom. Today cellular systems still represent one of the fastest growing telecommunications systems.

GSM system is based on digital technology, which has many advantages over analog technology. It also provides the technology as ROAMING, SMS TELETEX, Fax mail. In the 1980s most mobile cellular systems were based on analog systems as AMPS. Second generation cellular digital systems like GSM, DAMPS, were implemented then. From 1984-86, GSM focused on to compare different technical possibilities for transmission. It was decided to compare several technical proposals based on prototypes allowing actual radio transmission.

The decision of adopting a digital technology for GSM was made in the course of developing the standard. During the development of GSM, the telecommunications industry converted to digital methods. The ISDN network is an example of this evolution. In order to make GSM compatible with the services offered by ISDN, it was decide that the digital technology was the best option. Additionally, a digital system allows, easily than an analog one, the implementation of future improvements and the change of its own characteristics.

II. Discontinuous Transmission in GSM

Discontinuous Transmission (DTX) is a mechanism, which allows the radio transmitter to be switched off most of the time during speech pauses [1-2] for the following two purposes:

- to save power in the Mobile Station (MS);
- to reduce the overall interference level over the air interface.

DTX in uplink shall be in operation within the GSM MS, if commanded so by the network, see GSM 04.08. The MS shall handle DTX in downlink at any time, regardless, whether DTX in uplink is commanded or not.

Whenever there is no speech activity at the transcoder unit instead of sending coded noise frame on to the air interface the TX modulator can be made to relax and send nothing on its carrier for a period of no speech activity and send comfort noise parameters at regular intervals within this period to avoid the muting sound which can cause annoying to the user. This period of no speech activity is called discontinuous transmission.

The SID-UPDATE frame contains the comfort noise parameters, which are sent periodically for the speech decoder to insert the noise instead of muting which may cause annoying during this speech inactivity.

General Organization: The overall DTX mechanism described in the present document requires the following functions [1-2]:

A Voice Activity Detector (VAD) on the transmit (TX) side;

Evaluation of the background acoustic noise on the transmit (TX) side, in order to transmit characteristic parameters to the receive (RX) side;

Generation on receive (RX) side of a similar noise, called comfort noise, during periods where the radio transmission is switched off.

The Voice Activity Detector (VAD) is defined in GSM 06.94 and the comfort noise functions in GSM 06.92. Both are based partly on the speech transcoder and its internal variables, defined in GSM 06.90. In addition to these functions, if the parameters arriving at the RX side are detected to be seriously corrupted by errors, the speech or comfort noise must be generated from substituted data in order to avoid seriously annoying effects for the listener. This function is defined in GSM 06.91. An overall description of the speech processing parts can be found in GSM 06.71.The description for Tandem Free Operation is given in GSM 08.62.

III. Real Transport Protocol in GSM

The Real-time Transport Protocol (RTP) is a network protocol for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications including WebRTC, television services and web-based push-to-talk features.

RTP typically runs over User Datagram Protocol (UDP). RTP is used in conjunction with the RTP Control Protocol (RTCP). While RTP carries the media streams (e.g., audio and video), RTCP is used to monitor transmission statistics and quality of service (QoS) and aids synchronization of multiple streams. RTP is one of the technical foundations of Voice over IP and in this context is often used in conjunction with a signaling protocol such as the Session Initiation Protocol (SIP) which establishes connections across the network.

The sequence numbers included in RTP allow the receiver to reconstruct the sender's packet sequence, but sequence numbers might also be used to determine the proper location of a packet. The audio conferencing application used by each conference participant sends audio data in small chunks of, say, 20 ms duration [6]. Each chunk of audio data is preceded by an RTP header; RTP header and data are in turn contained in a UDP packet. The RTP header indicates what type of audio encoding (such as PCM, ADPCM or LPC) is contained in each packet so that senders can change the encoding during a conference, for example, to accommodate a new participant that is connected through a low-bandwidth link or react to indications of network congestion.

The Internet, like other packet networks [6], occasionally loses and reorders packets and delays them by variable amounts of time. To cope with these impairments, the RTP header contains timing information and a sequence number that allow the receivers to reconstruct the timing produced by the source, so that in this example, chunks of audio are contiguously played out the speaker every 20 ms. This timing reconstruction is performed separately for each source of RTP packets in the conference. The sequence number can also be used by the receiver to estimate how many packets are being lost.

IV. Test Setup, Results and Analysis

Test setup to take the observation is prepared as shown in figure Fig. 1.

To carry out testing Nethawk's RBS Tester (BSC Simulator) is used. Developed code is merged with existing software and BTS is updated with Full rate Discontinuous Transmission [7].

Nethawk's RBS tester is uploaded with Fullrate –Discontinuous transmission enabled configuration file and relevant scripts. Link b/w all nodes are configured and made UP. Mobile handset with having adequate SIM is connected using RF cable.

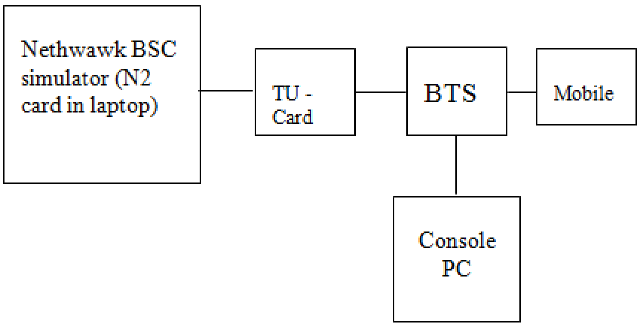

Figure1: Full Rate Discontinuous Transmission Test Setup

The RTP periodicity in normal condition (DTX is disable) is 20 msec [6] as shown in Fig. 2. and in order to reduce the Abis Bandwith, we had changed the SID Update periodicity to 480 msec in DTX for FR Call in BTS as shown in Fig. 3 and this will leads to save the bandwidth of Abis interface as shown in Fig. 4 and the bandwidth when DTX in disable is shown in Fig. 5.

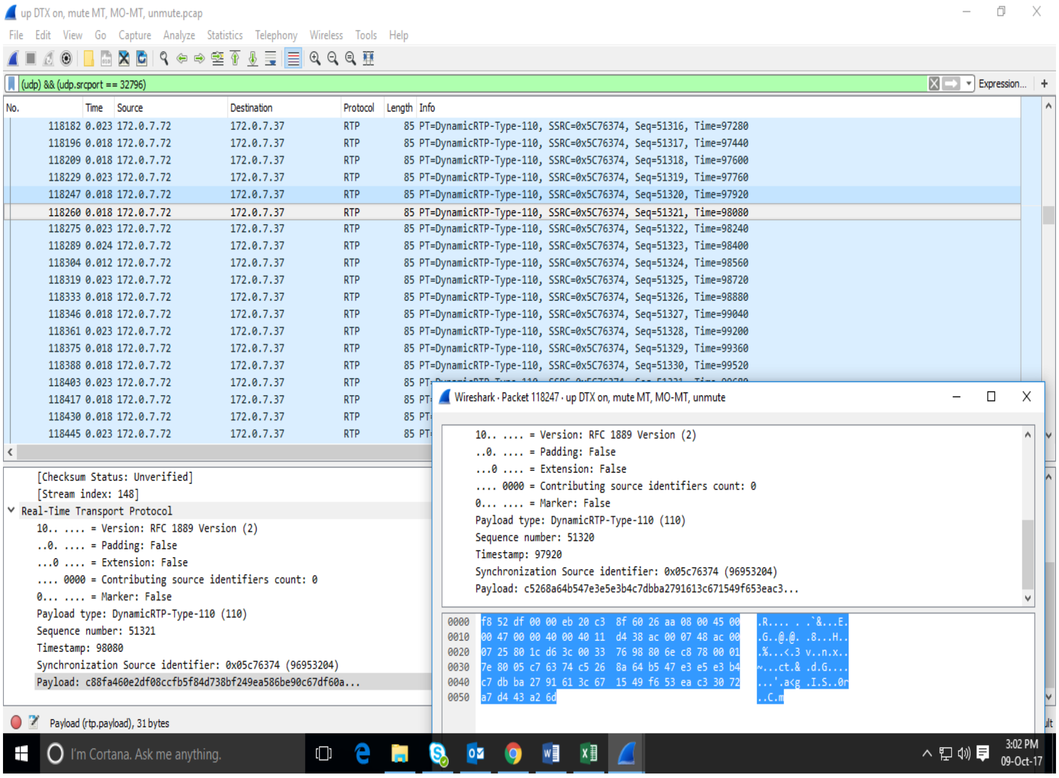

Figure 2: RTP Packets sent at 20 msec

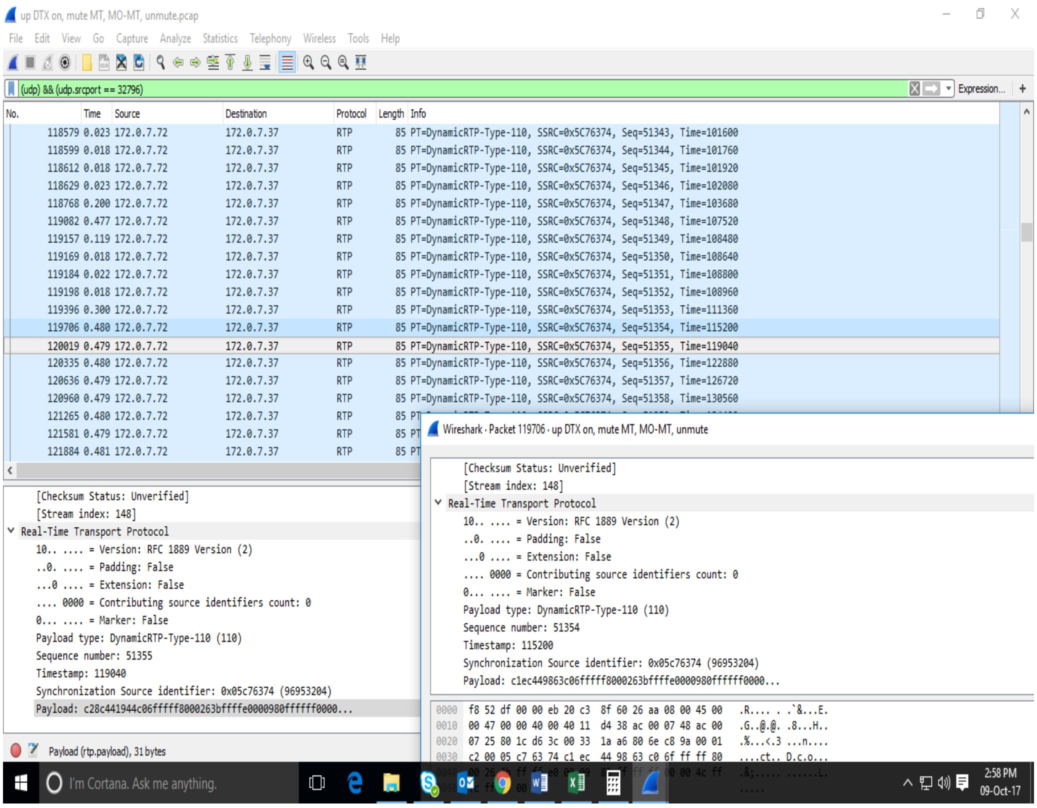

Figure 3: RTP Packets sent at 480 msec

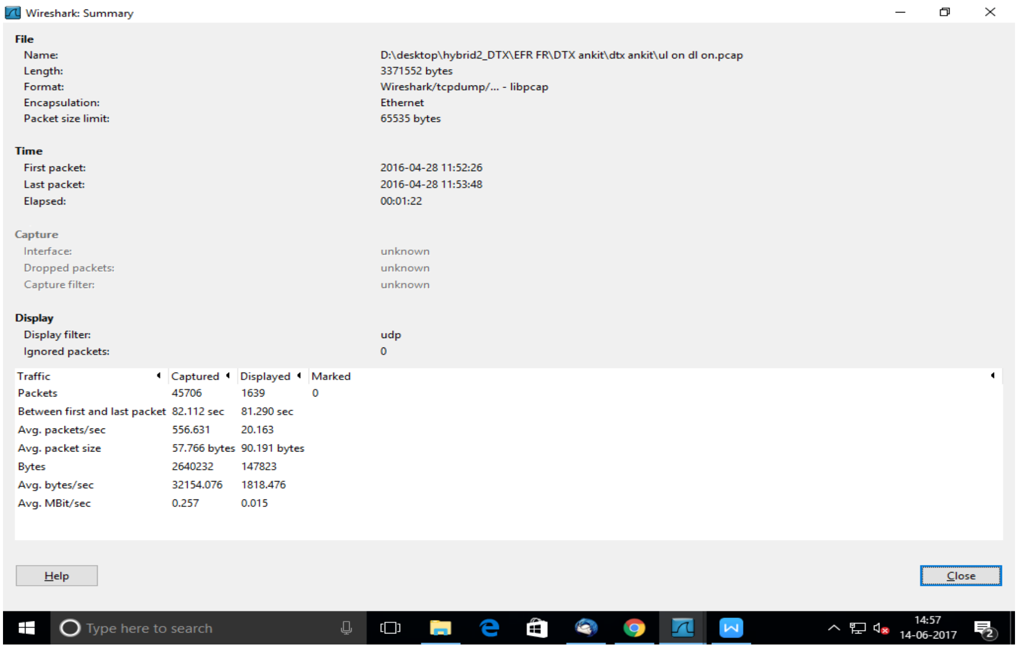

Figure 4: Bandwidth when DTX is Enable

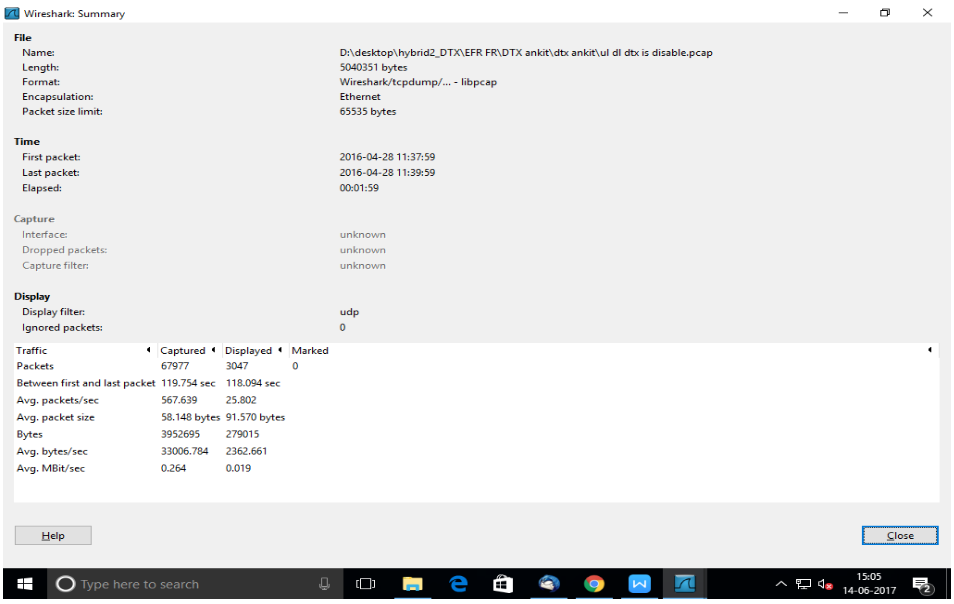

Figure 5: Bandwidth when DTX is Disable

V. Conclusion And Future Work

Successful integration of this code with existing BTS software will serve the purpose To save Bandwidth on Abis Interface, to reduce the overall interference in the networks, to save Mobile and BTS Power and to reduce the overall interference in the networks. Bandwidth saving on Abis interface for full rate speech traffic channels with DTX feature developed as part of the project work. Some possible extension related to the work done are bandwidth saving on Abis interface with AMR codec, Bandwidth saving on Abis interface with HR, EFR codec.

References

[1]. 3GPP TS 06.93 v 7.5.0; "Discontinuous Transmission (DTX), for Adaptive Multi-Rate (AMR) speech traffic channels".

[2]. 3GPP TS 06.101; "AMR speech codec frame structure"

[3]. GSM TS 45.03; "Digital Cellular telecommunication system (phase 2+); channel coding"

[4]. 3GPP TS Divya Choudhary, Abhinav Kumar, "Study and Performance of AMR Codecs for GSM", International Journal of Advanced Research in Computer and Communication Enginnering, Volume 3, Issue 10, Oct 2014, ISBN: 2278-1021.

[5]. Amit Dixit, and Aanchal Garg, "A Study of Cellular GSM & CDMA –Based for New Generation Mobile Radio System", IJCAT - International Journal of Computing and Technology, Volume 1, Issue 8, September 2014, ISSN: 2348 - 6090.

[6]. H. Schulzrinne, S. Casner, R. Frederick, V. Jacobson, "RTP: A Transport Protocol for Real Time Applications", RFC 3550, July 2003.

[7]. H. Schulzrinne, S. Casner, "RTP Profile for Audio and Video Conference with Minimal Control", RFC 3551, July 2003.

[8]. Ankit Singhal, Vikram Singh, "Study of GSM Network with its feature DTX", International Conference on VLSI Communication and Networks, VCAN, April 2015, ISBN 978-93-84869-55-7.

[9]. Guifen GU, and Guili Peng, "The Survey of GSM Wireless Communication System", International Conference on Computer and Information Application (ICCIA), 2010, IEEE Publisher, ISBN 978-1-4244-8598-7.

[10]. GSM Networks: Protocol, Terminology, and Implementation, Gunnar Heine, Artech House mobile communication library, ISBN 0-89006-471-7.

[11]. Communication Systems, Simon Haykin, John Wiley & Sons Inc., 4th ed., ISBN 0-471-17869-1.

IOSR Journal of Electronics and Communication Engineering (IOSR-JECE) is UGC approved Journal with Sl. No. 5016, Journal no. 49082.

Ankit Singhal Bandwidth Saving on ABIS Interface When GSM's Feature DTX is Enabled." IOSR Journal of Electronics and Communication Engineering (IOSR-JECE), vol. 12, no. 5, 2017, pp. 36-40.