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MULTIMEDIA QoS AWARE DATA TRANSFER OVER THE PACKET BASED NETWORK

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Анотація

В роботі запропоновано вдосконалений метод передачі даних реального часу по віртуальним каналам пакетної мережі за допомогою кадрів Ethernet. Даний метод спрямований на вирішення питань конвергенції телекомунікаційних технологій та покращення ефективності використання пропускної здатності каналу зв'язку.

Abstract

An enhanced method introduced for real time data transfer over the virtual channel of packet based serial trunk with the use of Ethernet frames. This method addresses the issues of telecommunication technologies convergence and bandwidth utilization improvement.

Introduction

According to the ITU-T recommendation Y.2001, next generation networks (NGN) must be capable of handling voice, video, and data communications with proper quality of service (QoS) in the IP-based packet switching transport infrastructure. Packet switching technique was originally developed for reliable and not latency sensitive data transmission. On the other hand, significant advantage of the circuit switching technique is high QoS guarantees for real-time applications (incl. latency, jitter, and bandwidth control). The crucial question "How to benefit both circuit and packet switching in respect to the NGN's promised QoS?" has no explicit answer yet. Therefore, more researches about this aspect are required.

The objective of this paper is to design an enhanced method of multimedia QoS aware data transfer over the packet based telecommunication network in respect to the QoS demands and using existing network infrastructure.

TDM and packet data flow integration over the packet-based Ethernet network

At the present time, the Ethernet is one of the most commonly used telecom technology. However, the data link layer in an Ethernet local area network experiences some issues in respect to the TDM traffic QoS-provision. On the other hand, a novel conceptual approach to multimedia data flow control on the Ukraine integrated telecommunication technology platform (UA-ITT, [1]) implies deployment of new type on network interfaces, which are compliant with ITT model on the data link layer.

The concept of data flow control due to the ITT/Ethernet paradigm is reflected in fig.1. Any two adjacent routing switches of packet based transport network (e.g. RS1 and RS2 in fig.1) are mutually connected through single or multiple serial communication trunks. Within a distinct serial trunk, the adjacent switches interoperate via two conventional Ethernet interfaces.

The mutual Ethernet frame streams permanently circulates with predefined frequency F and frame size S, regardless the actual data are really transferred over the trunk or some data padding occurs. These frames perform the function of synchronous transporting modules (STM). The payload of frames sequence creates an infinite plain series of distinct bytes. The average byte sequence rate in fig.1 (denoted f_0) can be calculated as certain function f which depends on F, S, frame overhead OH and inter-frame gap IFG: $f_0 = f(F, S, OH, IFG)$.

According to the ITT Dynamic digital Flow Switching (DFS), the plane byte sequence of layer L2 OSI is further handled as a structured sequence of command segments (CS) and data segments (DS) that corresponds to the layer L3 OSI (see fig.1). Each adjacent couple of CS and DS constitutes a virtual dynamic packet of individual flavor (denoted in fig.1 as F1, F2, F3).

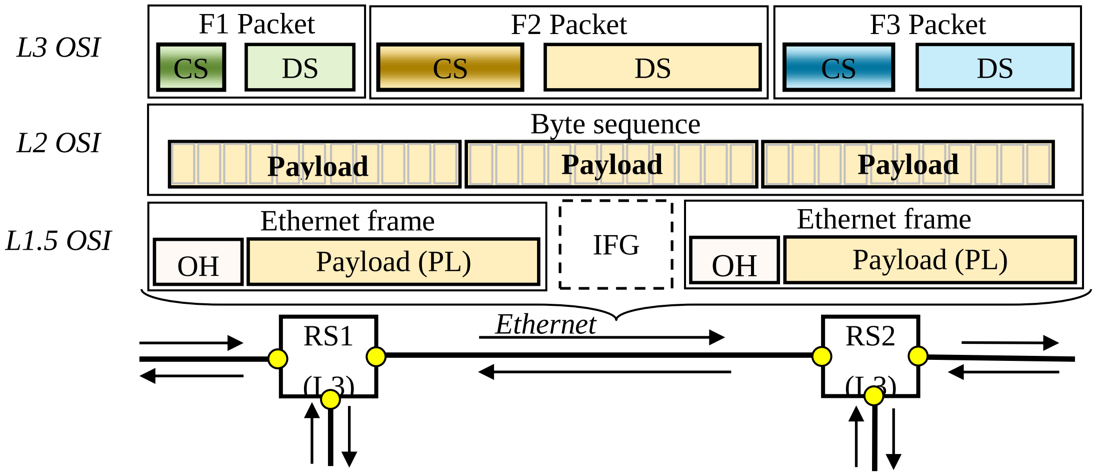

Figure 1 – ITT traffic transmission via Ethernet frames

Decomposition of common byte sequence into segregated command and data segments is performed due to the two one-byte meta-commands: command delimiter "11111111" (FF); data delimiter "00000000" (00). To mis-confuse "FF" or "00" symbols in data segment body, the byte stuffing mechanism is applied: two commands are reserved ("FF01" and "FF02") to replace data symbols "FF" and "00".

To provide the TDM multimedia data transfer within a dynamic packet flow of layer L3 in fig.1, the mechanism of meta-synchronous time-slot allocation is developed (see fig.2). Four real-time data flavors (1 to 4) are allocated at predetermined synchronous time-slots. This enables the guaranteed time delay restriction while forwarding real time data segments. Diverse frequencies of real time data appearance result in unutilized stochastically dispersed time intervals of arbitrary sizes in common byte sequence. To utilize these intervals, an extra packet delivery queue is arranged with a special mechanism of packet fragmentation and defragmentation (flavor 5, fig.2). According to this mechanism, any packet could be divided in fragments of any size due to the four reserved two-byte commands: packet start "FF03"; packet end "FF04"; packet fragment start "FF05"; packet fragment end "FF06". In case the packet queue (flavor 5 in fig.2) is not empty, the serial trunk is apparently 100% utilized.

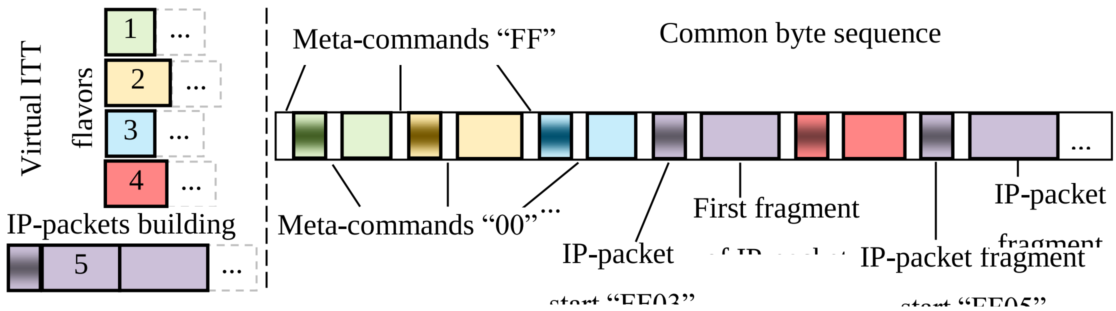

Figure 2 – Formation of virtual ITT flavors and IP-packets queue from common byte sequence

Conclusion

The proposed enhanced method of multimedia QoS aware data transfer over the packet based telecommunication network with generic Ethernet interfaces forms a solid background to integrate QoS aware multimedia services on the common platform.

References

1. Tikhonov V.I. Integrated telecommunication technology for the next generation networks / V.I. Tikhonov, P. P. Vorobiyenko // Proceedings of the ITU Kaleidoscope Academic Conference “Building Sustainable Communities”. – Japan, April 22-24, 2013. – P. 187-193.