END-TO-END TIMING TRANSPARENCY IN PACKET NETWORKS: GNSS-BASED SYNCHRONIZATION OF THE END-SYSTEMS

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ABSTRACT

In the past telecommunications were dominated by telephony. But now data communications are getting more and more important. Future networks, often called Next Generation Networks (NGN), will have to cope both with bursty computer data traffic and with real-time data streams such as telephony, video, and multimedia. Packet network technology is currently regarded as the most promising NGN approach. However, the real-time capability of today's packet network technology is still insufficient. This paper proposes a new solution for improving jitter performance of real-time communication services over packet networks in presence of high levels of packet delay variation. The idea is to synchronize both the transmitting and the receiving end systems to a stable common reference clock provided by a Global Navigation Satellite System (GNSS). GNSS-receivers installed in the access networks distribute the common time signal to the end systems. Simulations done for a scenario with G.729 coded voice communication over two medium-sized Ethernet LANs and a long-distance path in the global Internet show clearly, that a substantial reduction of the residual jitter at the receiving end can be achieved. In the case studied, standard deviation of the residual jitter was approximately 200 μs, compared to 3.0 ms with state-of-the-art packet network technology based on the Real-Time Protocol RTP.

1. INTRODUCTION

In the past telecommunications were dominated by telephony. But now data communications are getting more and more important. Future networks, often called Next Generation Networks (NGN), will have to cope both with bursty computer data traffic and with real-time data streams such as telephony, video, and multimedia. Packet network technology is currently regarded as the most promising NGN approach. However, the real-time capability of today's packet network technology is still insufficient. This paper proposes a new solution for improving jitter performance of real-time communication services over packet networks in presence of high levels of packet delay variation. The technique is based on the idea of synchronizing both the transmitting and the receiving end systems to a stable common reference clock. The proposed solution makes use of a Global Navigation Satellite System (GNSS) for the distribution of a common time signal. GNSS-receivers installed in the LANs (see Figure 1) act as Stratum 1 NTP servers which distribute the common time signal via the Network Time Protocol NTP (see [16]) to the end systems. The transmitting end systems uses the common time signal to timestamp the packets. The receiving end system uses the common time signal to decide at what time instant a timestamped data unit is to be delivered to the real-time application. The whole idea is to exploit the fact that the packet delay variation introduced by the LANs on the NTP association is much lower than the packet delay variation introduced by the internet on the user association. The new system is called Synchronous Real-Time Protocol or SyRTP (by analogy to the existing Real-Time Protocol RTP, see [16]). Using a simulation program written with Mathcad 7.0 (a commercial mathematical calculation program), a scenario with G.729 coded voice communication (see [6]) over two medium-sized Ethernet LANs and a long-distance path in the global Internet was simulated.

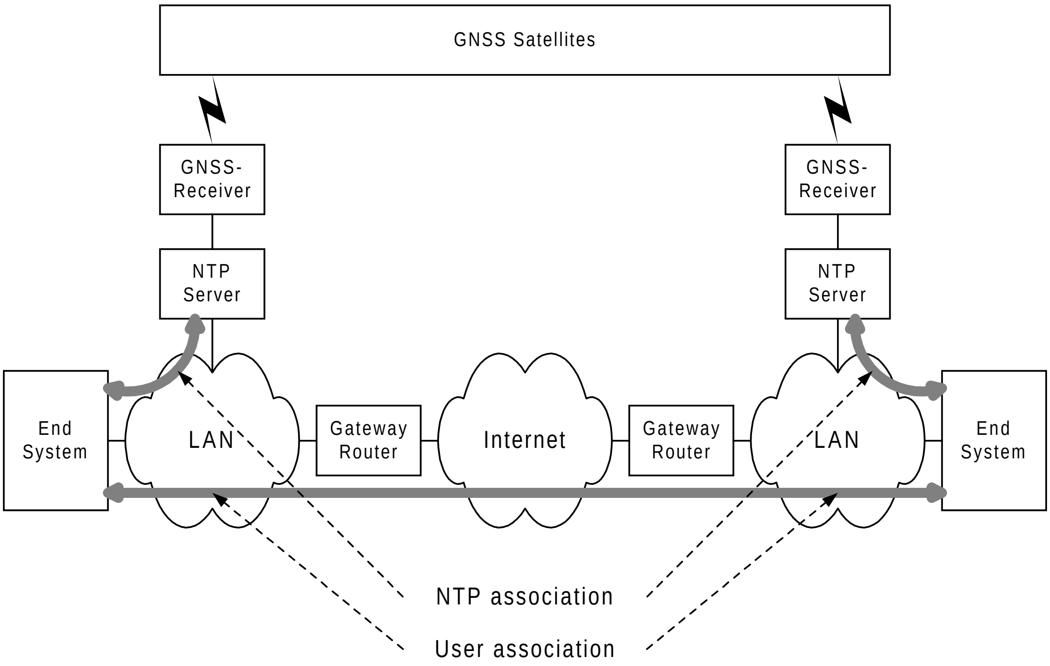

Figure 1: Communication architecture

2. TECHNICAL CONCEPT

The basic idea is to make use of a common time reference in order to improve the end-to-end-transfer of the timing information contained in real-time data streams. There are two important questions:

- Where does the common time reference come from and how is it distributed to the user equipment?
- What mechanism is used to exploit the common time reference for the transfer of timing?

Concerning the first question, the proposal here is to use a combination of GNSS and NTP (Network Time Protocol) for the distribution of a common time reference down to the end systems.

In a GNSS, a number of satellites orbiting around the Earth constantly broadcast information about their position in orbit and their on-board-clock's time. A GNSS-receiver receiving signals from at least four satellites, is able to determine its own position and time. In the application discussed herein, the accuracy of the time information is the important parameter. Fortunately, all existing and planned GNSS distribute time with a very high accuracy (see [10]). GNSS-receivers specifically designed for time distribution can be expected to deliver time with an accuracy of $\pm$ 50 ns or better relative to the international time scale UTC (Universal Time Coordinated).

NTP is a protocol standardized by the IETF for the distribution of time over an IP network. One of the NTP's operating modes is based on the client-server model. In this mode, a client, e.g. our end system, sends a request message to an NTP-server and the NTP-server sends back a response message containing the actual time. During the message exchange, the network's transfer delay is measured and compensated for. The accuracy of the distributed time information depends on the accuracy of the NTP-server's clock and the accuracy of the delay measurement. The latter depends mainly on the packet delay variation introduced by the underlying IP network. The typical time error introduced by NTP in wide-area networks is in the order of several tens of milliseconds (see [12]). However, in Local Area Networks (LAN), the error introduced by the network is in the millisecond and sub-millisecond region (see [13]). Hence the idea to install an NTP-server driven by a GNSS-receiver in the LAN, so that the end systems in that LAN get accurate time from that NTP-server.

The transmitter's and the receiver's common knowledge of time can be used to transfer the application data-stream's timing from end to end. In the transmitter (see Figure 2), the generation instants e_{DT} *(k)* (time instant when the *k*-th data unit is generated) are measured relative to time signal derived from the common reference clock. The measured values are transmitted as timestamps together with the data units to the receiver. The receiver (see Figure 3) first puts the data units into a play-out buffer. The play-out time $e_{DR}(k)$ for each data unit is calculated by adding a constant value to the timestamp value. This constant value reflects the delay experienced by the data units on their way through the network. When the receiver's time signal reaches the calculated value $e_{DR}(k)$, the k-th data unit is removed from the play-out buffer and sent to the real-time application. This technique resembles RTP. The main difference is, that the receiver's time signals is now derived from the common reference clock, whereas with RTP it was reconstructed from the received timestamps. The new technique is suitable for periodic data-streams (constant bit rate services) as well as for certain kinds of non-periodic data-streams (variable bit rate services where there is one data-unit per packet or where the data-units are segmented into several packets).

3. SIMULATION MODEL

3.1. Overall Model Structure (see Figure 4)

- The bloc "Transmitting End System Model" generates a sequence of packets. Only important features of the packets are modeled, i.e. the packet transmit times (time when a packet is leaving the transmitter) and two of the SyRTP header fields, namely the Timestamp field and the Sequence Number field. The transmit times are imported from the bloc "Time Error of the Transmitter's System Clock".
- The two blocs "Time Error of the Transmitter's / Receiver's System Clock" generate sequences of pseudo-random numbers which model the phase-time (or time error) affecting the end system's internal clock. This error is mainly caused by the packet delay variation of the NTP association which locks the system clock to the GNSS-receiver.
- The bloc "Transfer Delay" generates a sequence of pseudo-random numbers which model the user association's packet transfer delays. For details see section 3.3 below.
- The bloc "Receiving End System Model" calculates the play-out times of the restituted real-time data units (time instant when a data units is sent from the play-out buffer to the real-time application).

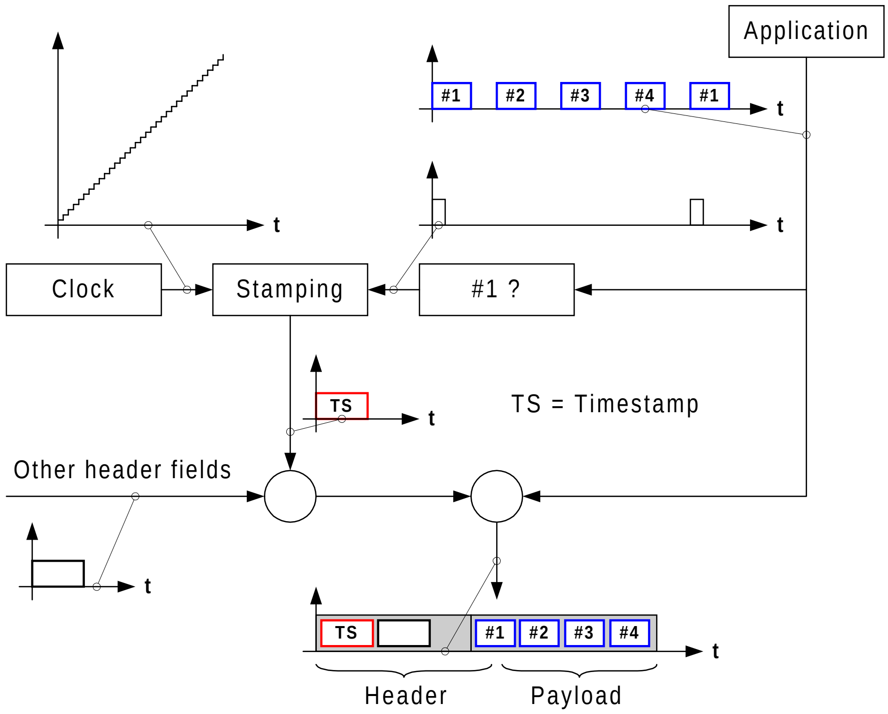

Figure 2: Application Layer process, transmitter side

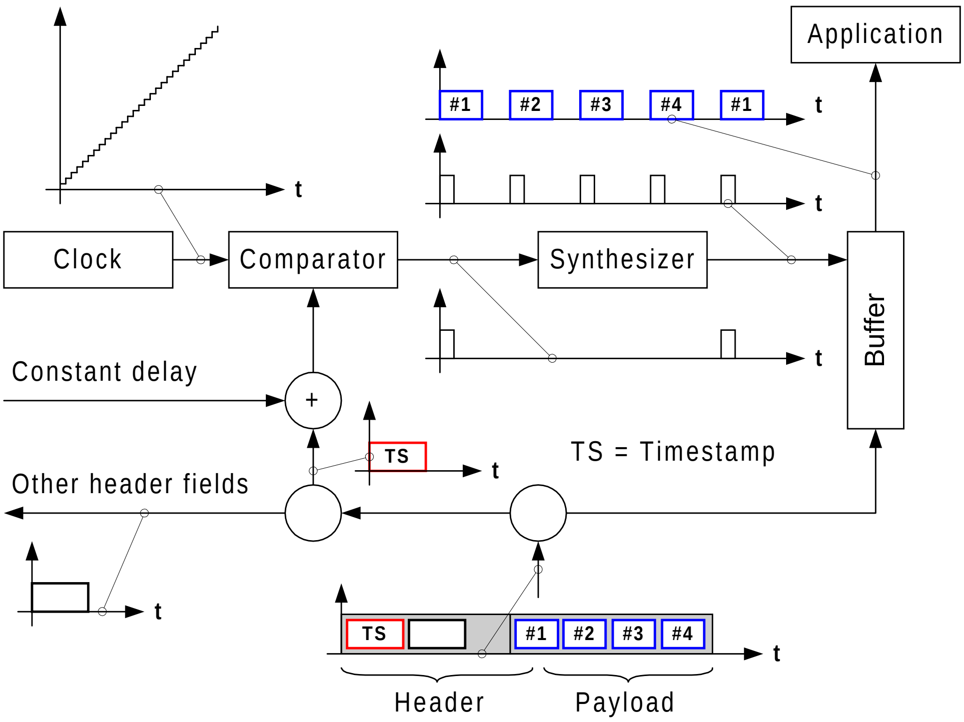

Figure 3: Application Layer process, receiver side

3.2 Time Error of the System Clocks

The main causes for the instabilities and inaccuracies of a clock synchronized to an NTP server are the packet delay variation of the network and the oscillator's sensitivity to temperature changes. These effects have been studied extensively by D. Mills (see [12], [13]). His results for typical Ethernet LANs show that the clock's phase-time closely matches a Gaussian White Phase Modulation noise model with a root-mean-square value x_{RMS} of 220 µs.

3.3 Transfer Delay on the User Association

The user association's transfer delay is defined as the sum of the delay contributions due to the internet, the two LANs, and the lower layer protocol entities (Physical and Data Layers) in the transmitter and the receiver.

The proposed model for the user association delay d_I consists of two terms, i.e. a constant propagation delay D_P = 25 ms, and a variable queueing delay d_Q:

$$d_I = D_P + d_Q$$

The queueing delay d_Q is modeled as a sequence of pseudo-random numbers with a GAPP probability distribution. The GAPP distribution proposed by Pitts and Schormann is a discrete variable distribution, i.e. the random variable d_I can only take a countable number of values: $d_I \in \{0, T_P, 2T_P, 3T_P, \ldots, n \cdot T_P, \ldots\}$, where T_P is the nominal period between consecutive packets. The following equation gives the GAPP cumulative distribution function (c.d.f):

$$F_{GAPP}\left(d_Q = n \cdot T_P\right) = \sum_{i=0}^{n} \frac{\left(i + N_Q - 1\right)!}{\left(N_Q - 1\right)! \cdot i!} \cdot \eta^i \cdot \left(1 - \eta\right)^{N_Q}$$

with

$$\eta = \frac{\lambda \cdot \exp(\lambda) - \exp(\lambda) - \lambda^2 + \lambda + \exp(-\lambda)}{\lambda - 1 + \exp(-\lambda)}$$

N_Q is set to 15. The load parameter λ is adjusted until the standard deviation is equal to the 45 ms. This is the case with λ = 0.4.

In reality d_I is not a discrete, but a continuous random variable. For this reason a modified variant of the GAPP distribution is proposed. The GAPP distribution is turned into a continuous c.d.f. by simple linear interpolation. The result is a continuous c.d.f. composed of a multitude of linear segments. This distribution is called the IGAPP distribution ("I" from "interpolated").

3.4 Model for RTP

The simulation results presented in the next section compare the new SyRTP system with the IETF's existing RTP system. The simulation model used for the RTP system is very similar to the SyRTP model. The differences reflect the fact that with RTP the play-out timing is constructed solely from the received timestamps. This process makes use of a Phase Locked Loop (PLL) for the smoothing of instabilities caused by network packet delay variation.

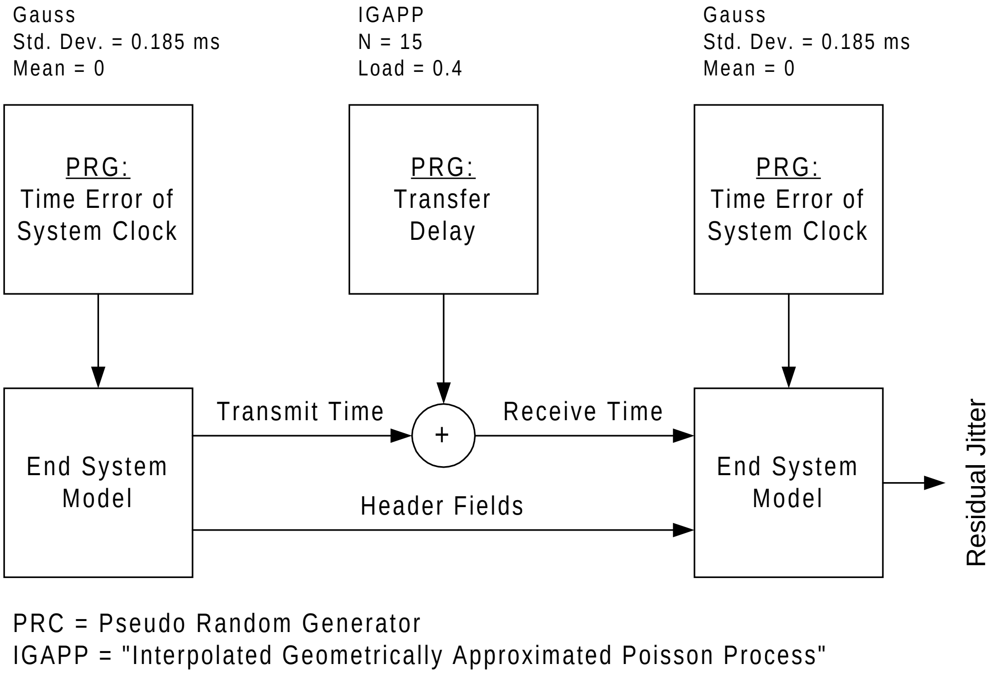

Figure 4: Structure of the simulation model

4. SIMULATION RESULTS

This section reports about a simulation run done for a particular real-time application, namely voice over G.729 codecs (see [6]).

Figure 5 shows how the application data units, i.e. the G.729 frames, arrive at and leave the play-out buffer of the receiver's SyRTP entity. The time intervals between two successive arrivals are very irregular due to the network's packet delay variation. The time intervals between two consecutive play-out deliveries is obviously far more regular than for the arrivals.

Figure 6 shows the phase-time for both the SyRTP and the RTP cases. The SyRTP trace shows smaller excursions than the RTP trace. The same tendency is confirmed by the comparison of the standard deviation calculated over the phase-time functions. With SyRTP, the standard deviation is 185 µs, with RTP, it is 3.0 ms. Figure 6 also reveals, that the differences between the two cases are more important for long-term phase-time variations than for short-term variations.

The simulation program also calculates the mean end-to-end delay for both systems. The mean delay is 199 ms for the SyRTP system, and 197 ms for the RTP system. In other words, they are nearly the same. Incidentally, these values are higher than the timing transparency requirements for telephony, which happens to be 150 ms (see [19]).

5. CONCLUSIONS

The simulations results lead to the following main conclusions:

- The results show clearly, that the synchronization of both the transmitting and the receiving end systems to a stable common reference clock has the potential to lower the residual jitter of real-time communication systems in the presence of high levels of packet delay variation in the network. In the case studied herein, standard deviation of the residual jitter with SyRTP is 15 times lower than with RTP.

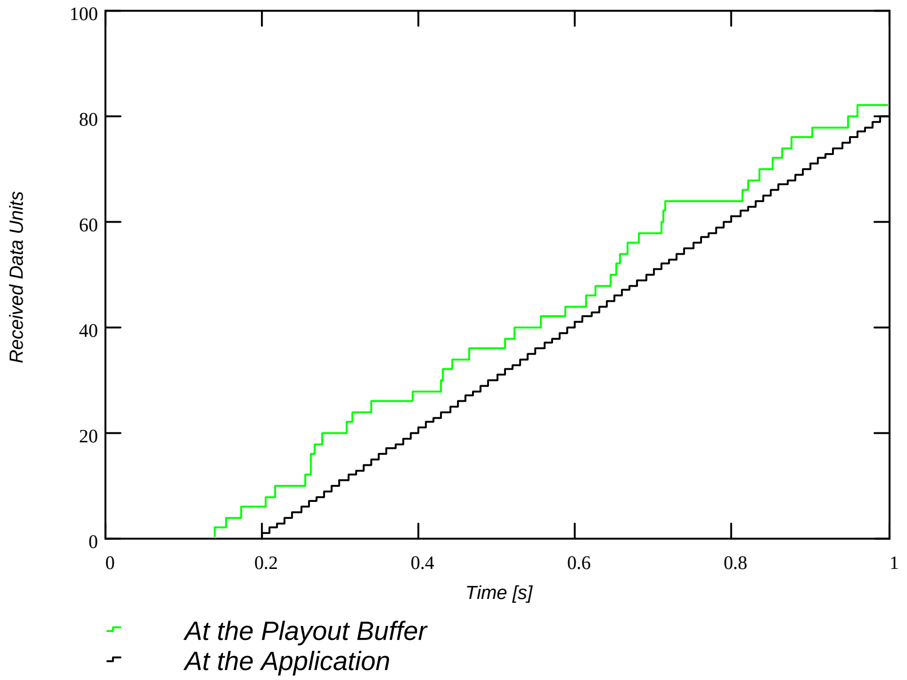

Figure 5: Arrival and Play-out of the data units (SyRTP)

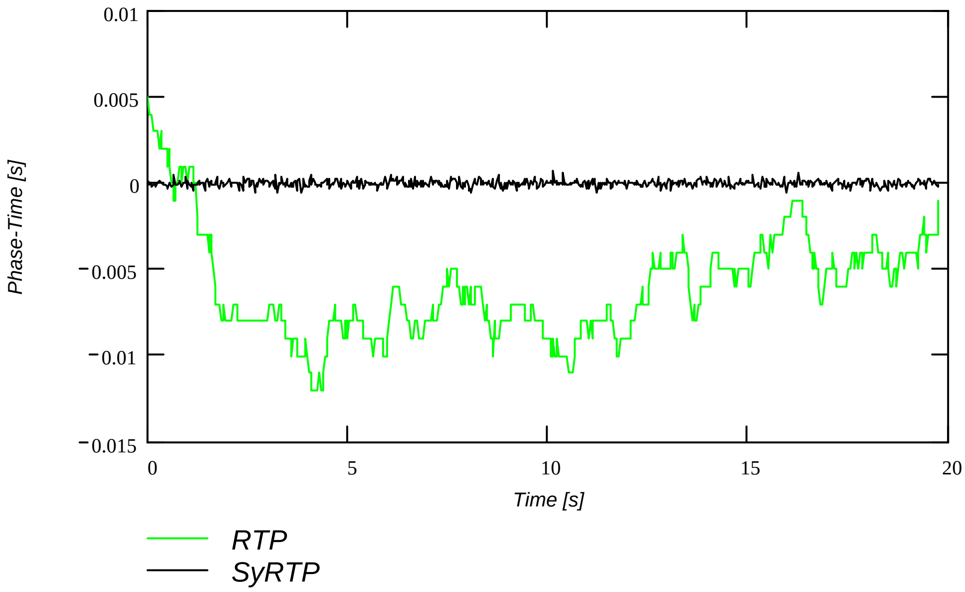

Figure 6: Phase-time of the restituted data units

- The simulations also show, that the jitter requirements of common multimedia applications can be met with state-of the-art RTP, at least with the simulated network scenario. These applications tolerate 4 to 5 ms of residual jitter (standard deviation). The RTP reference system generated only 3.0 ms in the simulation. The question whether the jitter performance will still be sufficient in other network scenarios than the one used in the simulation is for further study.
- The results obtained with SyRTP tend to show, that IP networks can be used for real-time applications with more stringent timing transparency requirements than those of common multimedia applications (4 to 5 ms of residual jitter). The residual jitter (standard deviation)

obtained with the SyRTP model is only approximately 200 µs. From the point of view of jitter performance, it is therefore conceivable to transmit some kind of high fidelity audio or high definition video over an internet or over the global Internet, if end systems are synchronized to a stable common clock, e.g. provided by a GNSS.

6. REFERENCES

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