BUFFER CONTROL TECHNIQUE FOR TRANSMISSION FREQUENCY RECOVERY OF CBR CONNECTIONS OVER ATM NETWORKS

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ABSTRACT

The transmission of audio-video coded signal in real time applications over ATM networks requires sophisticated techniques for synchronization and buffer control. The presence of ATM cell delay variation (CDV) represents the major jitter source affecting the reconstruction of the time reference signal associated to the actual real-time service.
In this paper is presented a buffer control technique and the related implementation aspects, based on both measure and utilization of CDV statistic or alternatively making use of buffer occupation statistic. The system allows setting target jitter attenuation in a way to have pre-established buffer underflow and overflow probabilities and its optimal utilization. The presented technique can be further extended to any asynchronous network context and is particularly suitable for high demanding professional audio-video applications.

1. INTRODUCTION

The problem of clock recovery of a stream transported over an ATM network, that uses time stamps as syncronization technique [3], has been analyzed by many authors [4] [6]. Different assumptions can be made on parameters that play a fundamental role on recovered clock performance: the statistical characteristics of network jitter, or the precision and stability required by reconstructed clock.
The problem can be solved in different ways: using directly time stamps or working on ATM cells flow at AAL level. The second one, recovering the information of transmission frequency f_s (see fig. 1), permits to have a more general solution that can be used everytime CDV reduction in CBR connections is requested.
The CDV can be sufficiently reduced, according to the applications of interest, applying adaptive jitter filtering. It is possible to use a buffer where the ATM cells are temporarily stored and than extracted with a frequency $\tilde{f}_s \approx f_s$ (see fig. 1).
Jitter is difficult to eliminate for two reasons [1] [5]:

- it has spectral components at very low frequency;
- it is of difficult statistical characterization, depending on the kind of background traffic and on the overall network load, that is not stationary. We only know the mean value that is zero.

For these reasons is necessary to use filter with a bandwidth as narrow as possible according to the finite dimension of the buffer, in a manner to avoid overflow and underflow conditions with high probability.

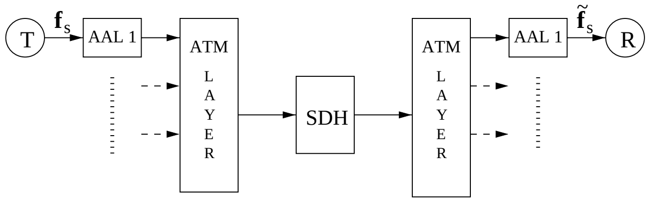

Figure 1: Transmission network: T transmitter, R receiver

The buffer control technique presented regulates the bandwidth of the filter using information about the buffer occupation and the statistical characteristic of the jitter. It permits to avoid overflow and underflow conditions with pre-established probabilities and optimal utilization of the buffer indipendently of its dimension.

2. MODELS

2.1. ATM cell jitter model

There have been many attempts to characterize the ATM cell delay statistics for transmission of constant bit rate streams [1] [5]. Our aim is to formulate a cell jitter reference model to be used for the design of an implementation of the buffer control technique presented in the next section.
A commonly accepted ATM network model [2] consists in a path of nodes (multiplexers and switching units), loaded with background traffic which enters and exits each node. Every ATM cell waits some time in the buffer queue of each node of the considered path. The load conditions of consecutive ATM nodes and the wait-time in the buffer of each node are correlated. For both this kind of wait-time correlation a first order Markov model can be used. A possible model for the wait-time in a buffer node is an exponential random variable. So the model for the transmission delay is the sum of n correlated exponential random variable, if n is the number of crossed ATM nodes.
The resulting transmission delay statistic distribution looks like an nth order Erlang PDF with a little higher dispersion (due to the wait-time correlation introduced).

We define the phase at istant t at the receiver (transmitter) as the number of bit received (transmitted) in the [0,t] interval. If the transmission system is transparent, the phase at the receiver would be the same as the phase at the transmitter, indicated in fig. 2 by the dashed line, where the slope is determined by the transmission

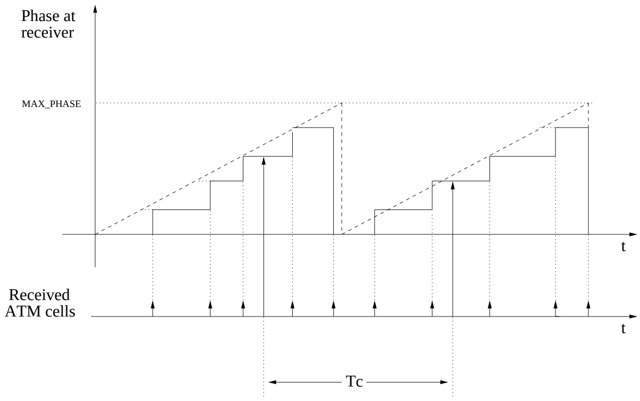

Figure 2: Ideal phase module max_phase (dashed line) and real phase module max_phase (solid line) at receiver.

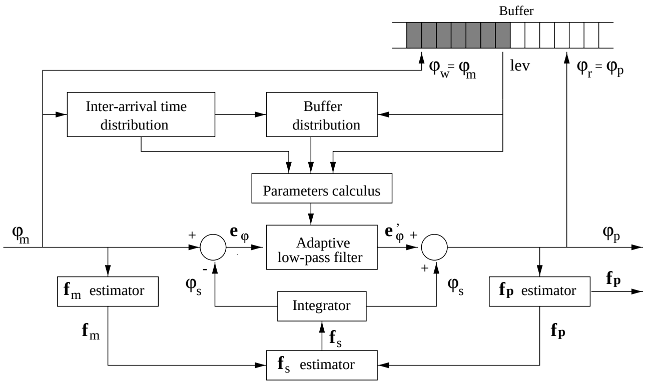

Figure 3: Buffer control technique model.

frequency f_s.
If we neglect the cell time duration T with respect to the inter-arrival time of consecutive ATM cells and consider the aleatority of the inter-arrival time due to ATM network, the phase at receiver assumes the aspect indicated in fig. 2 by the solid line. The difference between the real phase and the ideal phase at receiver is the jitter phase (according to our model).
Our duty is to reduce as much as possible the jitter phase, according to the finite dimension of the buffer used by the filtering system, avoiding overflow and underflow conditions with the pre-established probabilities.

2.2. Buffer control architecture

The buffer control architecture is showed in fig. 3.
f_m and φ_m are respectively the frequency and the phase measured by the filtering system; f_p and φ_p are respectively the phase and the frequency pre-processed (see section 5) by the filtering system; f_s and φ_s are respectively the estimated transmitter phase and frequency; at steady state f_s is practically equal to the transmission frequency and φ_s to the ideal phase at the receiver indicated in fig. 2. So at steady state e_φ and e'_φ are respectively the input jitter and the residual output jitter of the filtering system. φ_w and φ_r are respectively the phase of writing and reading of the buffer[1].
An initial inter-arrival time statistical distribution is updated at run-time by the inter-arrival time between consecutive ATM cells at the input of the filtering system. An initial buffer statistical distribution is updated at run-time by the buffer occupation level lev.
The parameters to regulate the bandwidth of the adaptive low-pass filter are calculated by the inter-arrival time distribution, the buffer distribution and the buffer occupation level.
The f_s-estimator produces f_s using the information about the inter-arrival time of consecutive ATM cells at input and output, where the jitter has been nearly eliminated, of the filtering system.

[1] In a first approximation f_m-estimator and f_p-estimator are derivators.

3. FILTERING SYSTEM DESIGN

The initial inter-arrival time statistical distribution can be obtained using the model described in par. 2.1.
The initial buffer statistical distribution can be obtained by simulation in the following way:

1. bulding a first buffer statistical distribution in the hypothesis to have an infinite dimension buffer, using nominal transmission frequency as costant reading frequency;
2. truncating the obtained distribution, eliminating the same area on the left and on the right sides, with the left and right queues respectively, to include a number of possible buffer occupation levels equal to which containable in the filtering system buffer;
3. normalizing the resulting distribution.

The center of the buffer is defined as the position corrisponding to the mean of the distribution obtained at step 1, thinking to lay it on the filtering system buffer[2] (see fig. 4). Because of the variations of the buffer distribution, at run-time the center can change position.

3.1. Adaptive low-pass filter

The low-pass filter used is a first order IIR filter. The in/out relation is:

$$\varphi_{pre}(n) = p \cdot \varphi_{pre}(n-1) + (1-p) \cdot \varphi_{mis}(n) \qquad (1)$$

where φ_{mis} and φ_{pre} are respectively the phase measured and pre-processed by the low-pass filter[3]; p is the pole of the filter. It must be $p \approx 1$ to reduce as much as possible the input jitter. In this case the normalized cut frequency is equal to $\frac{1}{2\pi} \cdot \frac{1-p}{p}$ (very low) and the normalized time costant is equal to $\frac{p}{1-p}$ (very high).
The bandwidth depends on the pole value. The pole expression is:

$$\frac{P}{1 + k \cdot W}$$

[2] In general, because of the asymmetry of the buffer distribution, the buffer center is not dislocated in the middle of the buffer.
[3] φ_{mis} and φ_{pre} coincide respectively with e_φ and e'_φ of fig. 3.

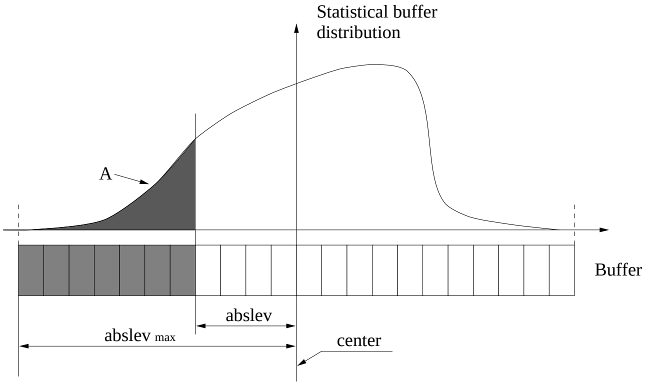

Figure 4: Defination of buffer parameters related to a generic distribution.

where P is the nominal pole value (very narrow bandwidth); W is the band modulation factor and k is determined in a manner to have pre-established probabilities of overflow and underflow.
The factor W is so defined:

$$W = lev^{esp} \cdot (A_{min} + A) \cdot c_B \tag{2}$$

where the factor lev^{esp} depends on the buffer occupation level: lev is defined as the distance of the buffer occupation level from the buffer center position, $abslev$, normalized to the maximum value that it can assume $abslev_{max}$, $lev = \frac{abslev}{abslev_{max}}$, in order to make the factor indipendent of the buffer dimension (see fig. 4). esp is a positive number in the set $\{2, 4, 8, 16, 32\}$, higher is the value and narrower is the bandwidth of the low-pass filter (under the same others conditions). The $A_{min} + A$ factor depends on the buffer statistical distribution: A is the area of the smallest buffer distribution queue determined by the buffer occupation level (see fig. 4). So the more probably the buffer occupation level can move to the nearest extreme of the buffer, that is towards an overflow or an underflow condition[4], the wider the low-pass filter band is let. A_{min} is a positive costant. c_B is a positive power of 2.
With this defination of W the bandwidth is very narrow either when the buffer occupation level is near to the buffer center position, far from overflow or underflow conditions, or at the extreme of the buffer distribution because in this case the buffer occupation level should move towards the buffer center position. The A_{min} factor is introduced to have a proper margin for buffer statistical distribution changes.
The esp and c_B parameters, from an initial value, are updated in a manner to obtain an optimal utilization of the buffer. When the buffer utilization in one direction from the center position is near to 100% (that is buffer nearly empty or full) the low-pass filter band must be wide to avoid the underflow or overflow condition with high probabilities. So to filter more on everage the input jitter is better to contain the buffer utilization between 60% and 80% (in both directions).
To determine the k factor periodically, not at every computational step, the W value is modify, with parabolic law, to a costant Wc from 80% to 85% of the buffer utilization.

[4] We suppose that at the begining the buffer is full till the center.

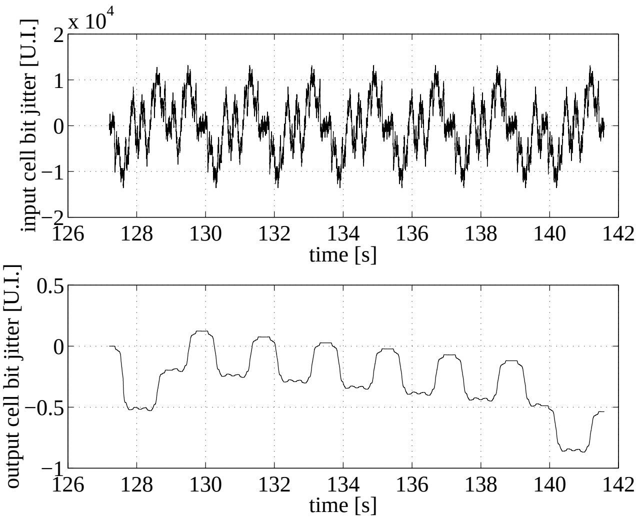

Figure 5: Input and output jitter at steady state.

The probability to lose cells in a ATM network is of the order of 10^{-9}; so it is rationale to guarantee underflow and overflow probabilities of the same order. We analyze the overflow case calculating the factor k_o (the underflow case with factor k_u, is analogous).
We want to determine the maximum input addition value X_{max} of the low-pass filter with probability at least of $1 - 10^{-9}$. Compensating it in condition of full buffer, we avoid the overflow condition with the same probability.
To determine X_{max}, we approximate the sum of $N = Tc/T$ inter-arrival time[5], where T is the nominal duration of an ATM cell at the transmitter (depending on the bit-rate) and Tc is the filtering system sampling time indicated in fig. 2, with a gaussian distribution of the same mean and variance, both calculated at run-time by means of the inter-arrival time distribution. The minimal sum of N inter-arrival time T_{min}, with probability at least of $1 - 10^{-9}$, can be calculated considering an interval of 5 times the variance on the left of the mean value. So the searched value X_{max} is:

$$\frac{Tc}{T_{min}/N} - X_{id}$$

where X_{id} is the input value due to the ideal phase, substracted to the input of the filtering system (see fig. 3). The factor k_o is calculated as the smallest positive real k that satisfy the relation:

$$\begin{aligned} \varphi_{pre}(n) \quad &= \quad p(k) \cdot \varphi_{pre}(n-1) + \\ &\quad\;\; (1 - p(k)) \cdot (\varphi_{pre}(n-1) + abslev_{max}) \\ &\geq \quad \varphi_{pre}(n-1) + X_{max} \end{aligned} \tag{3}$$

$p(k) = \frac{P}{1+Wc \cdot k}$, where it is imposed an addition output of the low-pass filter at least of X_{max}, when the buffer is full. In fact, in case of overflow, $abslev_{max}$ is the maximum exceeding phase that can be contained in the buffer, that is the maximum late of φ_{pre} with respect to φ_{mis} avoiding overflow.
Far from possible overflow condition the value of k can be smaller than k_o.

[5] Tc multiple of T.

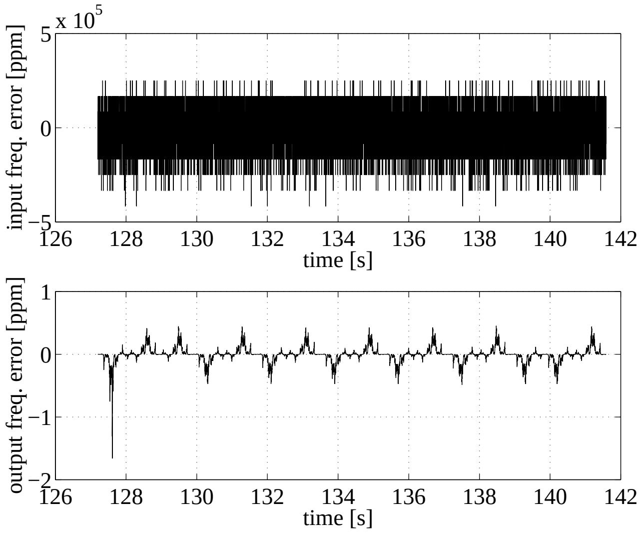

Figure 6: Input and output frequency error at steady state.

3.2. Transmission frequency estimation

The estimation of the transmission frequency f_s can be obtained by filtering the frequency mean value calculated on consecutive time interval at the input of the filtering system with some possible adjustements determined by checking the system filtering output. We use before a FIR filter to reduce the transient duration and then an IIR filter to have a narrower bandwidth. The FIR filter is a moving average system and the IIR filter is an adaptive low-pass filter with pole in $1 - \alpha$, where the factor α is updated at each filtering step with the law:

$$\alpha(n) = (1 - \delta) \cdot \alpha(n - 1) + \delta \cdot \alpha_\infty. \qquad (4)$$

δ determines how fast $\alpha(\cdot)$ tends to the steady value α_∞ ($0 \le \delta \le 1$).

4. SIMULATION RESULTS

In this section are presented some simulation results produced by adopting the following system parameters: 8 Mbit/s CBR source stream (so $T = 53\ \mu s$), ATM cell jitter generated as described in par. 2.1, filtering system sampling time $Tc = 12 \cdot T$, buffer dimension of 6 kbytes and a suitable choice of the others filtering system parameters.

The buffer dimension chosen is due to the intention of recovering with high precision the transmission frequency in spite of the considerable amount of jitter present at the input of the filtering system.

The transient state of the filtering system corrispond to the period necessary to estimate with sufficient precision the transmission frequency; it does not take long time so it is not important for evaluating the system filtering performance.

The input and the residual output jitter at the steady state are shown in fig. 5. The high frequency components of the jitter have been cut away and with the buffer dimension chosen also very low frequency components have been eliminated. You can have an approximative idea of the buffer utilization with respect to the buffer center at each computational step, considering the difference between the input jitter and the residual output jitter.

In fig. 6 is presented the input and the residual output frequency error at the steady state with respect to the transmission frequency f_s. The output/input frequency variance ratio is about of 10^{-12}.

By analizing several simulation results you could see that the system performance strongly depends on jitter correlation and jitter statistical power.

5. CONCLUSIONS

The presented innovative buffer control tecnique, based on either measure and utilization of CDV statistic or buffer occupation level, permits to solve, in a very efficient way, the problem of synchronization for audio-video coded signals, in real time applications, transmitted over ATM networks.

Using information about statistical properties of CDV you can better regolate the system filter bandwidth, in order to eliminate the input jitter, than using a pure deterministic based regulation. Moreover the buffer occupation checking with possible parameters updating allow an optimal utilization of the buffer itself. The technique is totally indipendent of the buffer dimension and assures overflows and underflows with pre-established probabilities.

The performance of the filtering system described above depends on the buffer dimension. The presented simulations have been produced with a large enough buffer to obtain the ordinary performance requested by video applications.

If the performance in recovering transmission frequency are particularly strict, it is convenient to introduce a PLL system, in a manner that the output phase of the filtering system φ_p constitutes the input phase of the PLL system. The PLL system requested results much less critic with respect to the PLL system used for solving syncronization problem in video applications over ATM networks, since it has attenuated jitter at its input.

Last but not least, the system proposed can be used in any case it is necessary to recover the transmission frequency in jittered networks.

6. REFERENCES

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