A signal can be stretched (or compressed) in time by subjecting an analogue signal to time-stretching (or compression). Time stretching increases the input bandwidth and sampling rate of the ADC and is best implemented using optoelectronic techniques.

The continual proliferation of digital signal processing (DSP) in high performance systems underscores the need for major advances in analogue-to-digital (A/D) converter technology. One example of systems whose needs far exceed the performance of current A/D converters (ADCs) is the digital receiver. In such systems the A/D conversion is performed at IF or RF frequencies, placing stringent requirements on the sampling frequency and input bandwidth of the A/D. While the electronic A/D performance continues to improve, the rate of improvement is too slow to satisfy the requirements of advanced systems in the foreseeable future. Hence it is widely recognised that new concepts leading to major advances in A/D technology are a priority. In this Letter, we propose a new A/D converter based on time-stretching the analogue signal prior to sampling and quantisation. By reducing the signal bandwidth, the new concept alleviates problems associated with the limited input bandwidth of ADCs.

**Time-stretched analogue-to-digital conversion**

A.S. Bhushan, F. Coppinger and B. Jalali

A new concept for analogue-to-digital (ADC) conversion is proposed and demonstrated. The analogue signal is stretched in time prior to sampling and quantisation. Time stretching increases the input bandwidth and sampling rate of the ADC and is best implemented using optoelectronic techniques.

The continual proliferation of digital signal processing (DSP) in high performance systems underscores the need for major advances in analogue-to-digital (A/D) converter technology. One example of systems whose needs far exceed the performance of current A/D converters (ADCs) is the digital receiver. In such systems the A/D conversion is performed at IF or RF frequencies, placing stringent requirements on the sampling frequency and input bandwidth of the A/D. While the electronic A/D performance continues to improve, the rate of improvement is too slow to satisfy the requirements of advanced systems in the foreseeable future. Hence it is widely recognised that new concepts leading to major advances in A/D technology are a priority. In this Letter, we propose a new A/D converter based on time-stretching the analogue signal prior to sampling and quantisation. By reducing the signal bandwidth, the new concept alleviates problems associated with the limited input bandwidth of ADCs.

**References**

2. European Digital Cellular Telecommunications System (Phase 2), European Telecommunications Standard Institute, 1992
3. PCS 1900 Air Interface Specification, Telecommunication Industry Association, 1994
and (iii) dispersion [3]. For an amplitude modulated (AM) signal, the modulation waveform is scaled in time by the factor \( D_2/D_1 \), while the carrier becomes chirped. Here \( D_1 \) and \( D_2 \) are the total dispersion in the first and second dispersion stages, respectively. Therefore, the signal can be either stretched \( (D_2 > D_1) \) or compressed \( (D_2 < D_1) \).

Time-stretching can also be practiced in the optical domain [4]. Optical implementation is attractive since extremely broadband dispersion and ultrahigh chirp rates are available [5]. Here an optical carrier is intensity modulated by the electrical RF signal, and dispersion and chirping functions are performed in the optical domain. Both the electrical carrier and its modulation are slowed down in time. The optical carrier is chirped; however, this is of no consequence, as optical field oscillations are filtered out by the photodetector. If the chirp bandwidth is much greater than the input signal bandwidth, then the first dispersion stage can be avoided with minor penalty in the signal-to-noise ratio [5]. The ability to slow down the microwave carrier and its modulation is of paramount importance in digital receivers wherein the ADC must capture the received signal before down-convertion.

![Fig. 2 Experimental setup](image)

Fig. 2 shows a block diagram of our experimental setup. To stretch the analogue signal in time, we employ an optoelectronic technique in which a dispersed optical pulse is used to impose an ultra wideband (7.5THz) chirp on the analogue signal. A 160fs pulse from a mode-locked Erbium-doped fibre ring laser (MLFRL) is dispersed in length \( L = 1.1 \)km of single mode fibre (SMF) generating a linearly chirped optical signal. The pulse has bandwidth \( > 60 \)nm (7.5THz). The chirped signal is intensity modulated by the analogue waveform in an electro-optic (LiNbO,) modulator, producing an optically chirped copy of the analogue waveform. The latter is linearly stretched in time by a second dispersion stage consisting of \( L = 7.6 \)km of SMF. The stretch factor is given approximately by \( M \sim (L_1 + L_2)/L_1 \). The stretched waveform is digitised by a 1Gs/s electronic ADC. For comparison, the waveform is also recorded with a high speed sampling oscilloscope. An arbitrary analogue waveform was generated by biasing the modulator at \( f_T \) and applying a pulse to it. The solid line in Fig. 3 shows the analogue time stretched waveform captured by the sampling oscilloscope and the data points, spaced 1ns apart, represent the digitised output of the ADC. The inset shows the waveform prior to time stretching. The bandwidth of the analogue waveform has been reduced by the stretch factor, \( M \sim 7.9 \), allowing it to be captured by the ADC. The effective sampling rate of the 1Gs/s electronic ADC has been increased to 7.9Gs/s.

![Fig. 3 Experimental data](image)

Simulations shown in Fig. 4 were performed assuming that no jitter noise is introduced in the segmentation stage of Fig. 1a. It is critical to minimise the jitter introduced by this stage as it diminishes the improvements gained by time-stretching. In general, time-stretching is more effective and simpler to implement in ADC applications wherein the input signal is non-continuous, and hence no segmentation is necessary. An important example of such applications, and one in which time-stretching can have a significant impact, is the pulsed radar.

In summary, we have proposed and demonstrated an A/D system wherein the input analogue signal is stretched in time prior to sampling and quantisation. This system can potentially mitigate the A/D bottleneck in the digital receiver.

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References

ABR performance in presence of bursty TCP traffic

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The available bit rate (ABR) service class is a solution for the integration of data traffic in asynchronous transfer mode networks. Many algorithms have been proposed to implement ABR services. The authors present simulation results showing poor performance by a common ABR algorithm when supporting TCP bursty traffic. As a solution to this problem, the authors propose time averaging of the parameters calculated by the ABR algorithms.

Introduction: Asynchronous transfer mode (ATM) is the generally accepted switching technology for the support of broadband integrated services digital networks. One of the services to be integrated is data service, which conveys random and bursty traffic. To efficiently manage data traffic in ATM networks, the ATM Forum has proposed two service categories: UBR (unspecified bit rate) and ABR (available bit rate) [1]. ABR is currently considered the most promising solution because of its ability to share the changing available network bandwidth with a low cell loss rate. Nevertheless, it is not proven that ABR is the best choice for data applications because it has not yet been tested in completely realistic environments.

ABR performs congestion control by means of a closed loop of resource management (RM) cells. This loop starts at the source, goes forward to the destination along the virtual connection path, and returns to the source. Returning RM cells collect information about the network state. Depending on this information, the source must limit its cell rate emission, according to a standard protocol. Several algorithms have been proposed to compute the information to be written into the RM cells for the switches [2]. ABR performance has been evaluated under different conditions, including persistent and modulated sources, as well as TCP sources [3].

For every traffic burst, an ABR connection operates in two phases: open loop and closed loop. The loop is open from the start of transmission until the first RM cell arrives at the source. From this moment, the connection is in closed loop. ABR algorithms are designed and tested to operate in closed loop, but the assumption that ABR connections will remain in closed loop most of the time is not always realistic. If an ABR connection conveys traffic bursts with a shorter transmission time than the connection round trip time (RTT), then it will operate in open loop. With this kind of traffic, ABR algorithm behaviour may be very different from what is seen in closed loop, with its performance worse, as we show below.

ABR with TCP traffic: The bursty profile of data traffic can be caused not only by the application demand patterns, but also by congestion control mechanisms in the upper layer protocols running in most legacy nets. In particular, TCP can cause this bursty traffic, leading the ABR connection to operate mostly in open loop.

TCP uses a window mechanism to control flow and avoid congestion. It does not use the network state information from lower layers, but obtains an estimation of the network state from ACK messages. Thus, network congestion is perceived by TCP only when packets are lost, which means a delay of at least one RTT. To cope with this uncertainty about the current network state, TCP implements a number of preventive mechanisms. The most important of these from the point of view of ABR, is the slow start mechanism. Slow start closes the transmission window when a loss segment is detected, and opens it gradually after the reception of new ACKs. In the slow start phase, TCP generates short traffic bursts with RTT millisecond periodicity.

ABR instability: During the open loop phase, an ABR source has no information about the network state. Thus, if some connections operate for a long time in open loop, the ABR algorithm can become unstable. This situation arises when sources emit bursty traffic, as TCP sources do during the slow start phase. If the bursts are shorter than RTT, then when a source receives RM cells, the burst has already ended. According to the ABR source behaviour, however, this information can be used erroneously in the next burst. This is what causes instability, because this information does not refer to the present state of the network.

This instability can be observed in the simulation results shown below. They correspond to a configuration of two switches, connected by a 100km trunk link at 150Mbit/s, and five terminals connected to each switch by 100km links. At the terminals, TCP is running and using ABR connections. There is a TCP connection between each pair of terminals on one side and the other. All of these connections convey unidirectional data traffic. The TCP transmission window is set to equal the product delay bandwidth, so the transmission rate is limited only by the link rate.

In the switch, ABR is implemented by the ERICA algorithm [4]. This algorithm computes the arrival cell rate to each output port (IR), measuring the arrival time of 60 cells, and counts the number of active connections N. A target rate TR of 90% of available bandwidth is defined. For each port, it calculates an ‘overload factor’ of = IR/TR, and a ‘fair cell rate’ CR = TR/N. When a backward RM cell arrives, then its explicit rate information is clipped to the maximum of CRC and CR. CRC is the connection current cell rate, carried by the forward RM cells. This algorithm leads, in most situations, to an accurate fair sharing of the target rate between the active connections.