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REVIEW OF DIGITAL SIGNAL PROCESSING

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Introduction to digital signal processing

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Digital signal processing is a generic term used to describe many and varied approaches to signal manipulation particularly, but not exclusively, when the signal is presented in digital form.

A more general term that subsumes the digital part of signal processing is that of discrete-time signal processing by which term it is meant that operations are carried out in discrete-time i.e. signals are sampled. The implementation therefore of signal processing operations in which signals are sampled and quantised is referred to as digital signal processing. Processing of signals digitally, therefore, requires a sampling operation to convert the time axis from continuous to discrete. This operation however is only possible if the analogue or continuous signals are bandlimited. If this is the case then the sampling operation in effect produces a signal which can be construed as existing at the sampling instants $t_k = kT$, T = Sampling Period (or at least we can say that the signal is relevant only at these sampling points.

The operation of signals in the time domain can be translated into a transformed domain as for example the Fourier domain in the case of continuous signals. The transformed domain for sampled signals as that of the Z-transformation which is defined as

$$F(z) \triangleq \sum_{k=0}^{\infty} f(kT).z^{-k}$$

where f(kT) are the sample values of the bandlimited signal f(f) at the sampling instant t = kT.

A signal processing scheme can be conveniently described in the transformed V(z) domain by a transfer function where the Z-transform of the discrete time signal V(kT) produced as a result of processing (linearly) a given signal u(kT) of Z-transform U(z), can be divided to yield,

$$V(z)\ /\ U(z) = G(z)$$ - the transfer function of the operation.

To determine the frequency domain behaviour of such an operation we merely replace Z = exp (jwT) in the transfer function.

By careful analysis it is possible to show that the frequency response can be read at some regular frequency values and on

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this basis one can arrive at the Discrete Fourier Transform represented as DFT). For the purpose of computation of Fourier Transforms it is necessary to express the DFT in a way that would require as few as possible manipulations and computer storage; this can be achieved through a technique referred to as the Fast Fourier Transform or FFT. A generalisation of FFT can be achieved by moving away from a complex number domain (as it is required for the frequency response) into a real number field. This generalisation suggested recently has opened many and as yet unfathomed areas of enquiry.

The purpose of the lecture is to give a review of these areas and to provide an introduction to the subject which can serve as a basis for the subsequent talks.

DIGITAL PROCESSING FOR RADAR APPLICATION

A. Tonkin

Synopsis The rapid advance of digital circuit technology over the past few years has had its effects on radar system designs. Signal processing techniques can now be employed to enhance radar system performance which have in the past been impracticable using the available technology.

This paper will consider a number of areas where digital technology has greatly improved on existing techniques or has opened up entirely new methods of signal processing. The first area considered is the moving target indicator (MTI). Conventionally this has been implemented by means of a mercury or quartz delay line matched to the interpulse period. Subtraction of a pulse from its predecessor results in static targets echoes, which return the same size pulse, being cancelled leaving the echoes from moving targets. Digitally this is achieved by digitizing the data and storing for the interpulse period. The inherent simplicity of the digital system over its analogue counterpart has major advantages in terms of size, weight and reliability. Moreover when more advanced systems are considered, for example, with varying interpulse periods the digital system with its puesdo delays is not significantly increased in complexity whereas its analogue counterpart becomes increasingly unwieldy.

The Pulse radar with MTI is the simplest of systems which uses the doppler shift of a moving target to distinguish it from static echo clutter. More advanced techniques can be employed which provide considerable advantages over the simple MTI system. They require however much more complex signal processing methods. Two such techniques are described in this paper, the frequency modulated continuous wave radar and the ambiguous range pulse radar. Both techniques rely on determining the spectrum of the received radar return to separate the moving target from the static clutter returns but differ in the method of ranging.

The frequency modulated continuous wave system uses a linear frequency modulation to determine target range. There is however a basic ambiguity between range and doppler shift which is resolved by further modulation phases with a different frequency slope. The spectral analysis can be achieved by means of digital spectrum analysers. It is then necessary to distinguish and eliminate clutter returns and to determine the range and radial velocity of the targets. Digital techniques to achieve this are outlined in the paper.

The ambiguous range pulse radar operates on a different principle relying on the delay from transmission to return of a pulse to determine range. However the p.r.f. is much higher than in a conventional pulse radar so that

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new pulses are transmitted before the pulse from distant targets is returned. Thus arrange ambiguity is obtained. The p.r.f. is not however sufficiently high for the Doppler shift of fast targets to be unambiguous. Again spectral analysis can be achieved digitally although this time on several range gates. The range and radial velocities are resolved by sending three or more bursts of pulses each at a different p.r.f. and comparing the alternative possible range and velocities on each.

Rather different applications of digital processing are the automatic target detectors. Basically they consist of two parts, an automatic theshold determinator and an integrator. The automatic theshold determinator operates by measuring the mean signal level over a number of independent but nearby cells and thesholding at a predetermined level above this. Several different approaches are discussed in the paper. With a conventional pulse radar several pulses are received from a given target as the beam sweeps through it. The integrator combines the returns from these pulses to maximise the effective signal to noise ratio on the target. Several types are considered in the paper ranging from the digital equivalent of the RC integrator to a binary counting integrator each having advantages in particular applications.

Some aspects of the application of digital techniques to radar displays are also considered in the paper. Scanning radars produce data naturally on a polar basis and, if the radar is on a moving platform, with a moving origin. The radar display picture using convential techniques is therefore, a polar scan system and when ground stabilized has a moving offset origin. Refresh rates are slow, typically once every 10 or 15 secs so that long persistence CRT's are required. Such systems are far from ideal for the operator. A system which converts the polar scan to a television type of radar scan would be a great improvement. A CRT type of scan converter can be used for this purpose or alternatively the conversion can be made using digital techniques.

Two main facilities are required by such a system, the co-ordinate conversion of the incoming data and the subsequent storage of the converted data for display. An outline of a means of achieving this digitally is given in the paper.

Similar equipment can be used in other radar roles. For example a sideways looking radar uses a fixed aerial array on the side of an aircraft and slowly builds up a radar map strip by strip as the aircraft moves over the ground.

The data is digitized and stored. It c an be output in a number of ways to suit the particular requirement. For example a 'rolling map' type of display can be presented with the whole picture moving slowly down the display as new data is received. This is equivalent to the view that can be seen by looking out of the side of the aircraft. Alternatively the picture can be ground stabilized so that new data slowly moves down the picture overwriting the oldest data.

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and phase modulation, is a good example of a flexible digital design. For v.s.b. modulation another method employs a straight digital multiplier for modulating a digital sinewave carrier with the digitally-filtered baseband data signal.
Filtering: Digital filtering in modems is usually restricted to spectral shaping functions in the pass-band, and the usual form is the finite-impulse-response or transversal filter structure which can be designed with a linear phase characteristic. However, there are some interesting variations. One manufacturer employs general purpose digital filter structures comprising three LSI circuits, each of which, at a 10 KHz sampling rate, provides up to 26 poles and 26 zeros of filtering which are distributed about the modem to provide filtering, all-pass delay equalization etc. The flexibility of this approach is further exploited by adaptively varying the filter coefficients to provide coarse equalization (in the frequency domain) of severe transmission line impairments. Another manufacturer uses an LSI circuit containing two independent second-order recursive structures with programmable coefficients to provide adaptive channel-switching for a multi-frequency modem.
Adaptive Equalization: The area where digital processing is usually applied is in adaptive equalization, although analogue or sampled-analogue adaptive equalizers are inexpensive and very effective for some applications. The equalizer is basically a transversal filter where the tap gains are controlled by an algorithm (usually a form of mean-square-error minimisation) to form the inverse filter for the impulse response of the transmission channel. Equalizers vary in structure and in length, depending upon the severity and the overall dispersion (echo) of the distortion to be equalized. V.s.b. modems employ one such equalizer operating on the demodulated base-band signal, whereas double-sideband multiphase modems require two equalizers with cross-correlation between the two. Although in analogue terms the latter represents a considerable increase in complexity, the use of multiplexed digital techniques substantially equates the processing for the two modulation schemes. Adaptive equalizers are implemented in many ways. There is a 4800 bit/sec modem which has a complete dual-channel adaptive equalizer on one chip. Several manufacturers have LSI equalizer processing circuits which can be cascaded, while others use off-the-shelf MSI and LSI circuits to provide maximum flexibility with low initial costs.
Other Modem Functions: Symbol timing recovery and carrier recovery are now usually performed by digital phase locked loops, although an analogue implementation is often preferred when a second-order loop is required. The digital property of indefinite storage is also employed to 'freeze' certain analogue functions like a.g.c. amplifier gain.

What lies in the future? As the modem market expands and these techniques become established, the use of LSI will become more widespread; but the next major step forward in this field is just outside the modem, namely, the "intelligent network processors". These equipments, which incorporate microprocessors and large storage capacity, seek to make far more efficient use of the data transmission capabilities of the modem by adaptive data concentration.

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TWO-DIMENSIONAL DIGITAL SIGNAL PROCESSING

V. Cappellini

The digital methods of two-dimensional (2-D) signal processing are of increasing interest due to several reasons: high efficiency which is obtainable permitting better signal processing and analysis; great expansion of standard computers or special type digital processors. Among the different methods and techniques of 2-D data processing, "digital filtering" and "data compression" are here mainly considered [1].

For digital filtering some non recursive and recursive types are described. Efficient 2-D non recursive digital filters, using suitable window functions[2] or frequency sampling algorithms, and 2-D recursive digital filters obtained through a suitable rotation and stabilization procedure are presented. In particular a new method is presented for designing in a fast way surely stable 2-D digital filters of recursive type with circular symmetry and zero phase, based on the transformation of the magnitude-squared frequency response of 1-D digital filter[3]. Efficiency comparisons are developed considering the main important parameters (in band peak ripple, out of band peak ripple, width of the transition band): tables of values are presented, useful for a practical selection of the desired digital filter. Some applications to the processing of 2-D data given in nuclear medicine (γ-camera) are shown; the application of low-pass and inverse 2-D digital filtering is described, performing also image enhancement and giving data in a form more suitable for pattern recognition and morphological analysis. Some examples of 2-D data processing relating to livers are in particular presented.

For data compression, some methods using prediction, interpolation, differential pulse code modulation, delta modulation, Fast Fourier Transform and Fast Walsh Transform are described[4]. In particular some methods using the Fast Walsh Transform alone or together other simple algorithms (threshold, prediction) applied on the sequences of the transformed image are described[5]. Efficiency comparisons are reported: results are given obtained by applying the different compression algorithms on standard images and on images received from earth resource satellities, giving the values of the compression ratio and of the errors after the reconstruction (r.m.s. error and peak error). The interest of these data compression methods both for image transmission and for archieval systems (solving information retrieval system) is outlined.

The application of the described methods of 2-D digital filtering and data compression to other types of 2-D data (facsimile, video-phone, X-ray scintigraphies, 2-D ultrasonic data in medicine) is also briefly described. Finally the problem of the implementation of the described methods is considered both in hardware and in software, with particular reference to on-line minicomputers and microcomputers.

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PICTURE CODING FOR VISUAL TELECOMMUNICATIONS

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1. Introduction Bandwidth compression, or redundancy removal, has appealed to academics since the very first visual communication systems were introduced for the reason that however sophisticated the encoding device it must inevitably fail to match the image processing efficiency of the human visual system. As integrated digital technology evolves, it becomes possible to significantly improve this match either in computer simulation or, to a lesser extent, in "real time" transmission systems. Research in the latter area is currently stimulated by such questions as "What digital capacity will be necessary in the evolving integrated digital network for the transmission of video signals?" "To what extent can data compression techniques economically exploit the properties of television and its recipient?"

This talk will seek to outline these properties and some techniques that are being applied in experimental systems for coding colour (broadcast standard) television, visual telephone and conference-television signals. In particular an examination will be carried out of the fundamental signal processing algorithms which are associated with exploiting the sample-to-sample, line-to-line and frame-to-frame similarities of television signals and associated perceptual redundancies of the viewer. Following a discussion of the 'irreversible' process of deriving a discrete representation of a continuous source, further reversible techniques of redundancy reduction will be described.

2. Intraframe source coding Two approaches to irreversible coding have become established - predictive and transform - but these are conceptually similar in that they seek to digitise the signal in such a way that quantising noise is assigned as a monotonically increasing function of spatial and/or temporal frequency. Thus the linear predictive encoder operates by differentiating the signal and quantising the result on a tapered (companded) characteristic. In the case of composite colour television signals which have a multimodal spectral distribution (having a second peak at subcarrier frequency); the signal can be simultaneously differentiated to determine the changing components at both dc (ie, luminance) and at subcarrier frequency (ie, chrominance), which are again submitted to a companded quantiser. Frame to frame prediction can also be used but is currently costly in terms of storage. Non-linear techniques are more effective in non-broadcast applications.

3. Interframe source coding For subscriber applications where the type of picture material is restricted, non-linear techniques can be used with effectiveness to transmit only the moving parts of each frame, stationary parts being re-iterated from a frame store at the decoder. Signal processing for movement detection attempts to solve the classical problem of detecting signals in noise. Having identified the moving area, linear predictive coding is used for transmission.

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4. Reversible (statistical) source coding While the ideal source coder produces the best match to perceptual requirements in terms of sampling and quantising, and generates a digital signal with no statistical redundancy, practical considerations of equipment complexity (for television at least) require these objectives to be separated. Typically a reversible variable length coding stage can exploit the non-uniform probability distribution of the output of an irreversible (predictive) source coder. This comprises a Read Only Memory and a buffer store of the largest practicable length to cater for statistical non-stationarity, together with some feedback mechanism to minimise effects of buffer underflow or overflow.

5. Applications The concepts introduced in previous sections will be discussed in the context of a number of practical applications.

While 7 bits/sample and a sampling rate in excess of the Nyquist rate for the defined signal bandwidth may be necessary for Analogue/Digital conversion of television (for a single coding operation), linear predictive techniques can reduce this to 5 for (colour) 625 line signals or as little as $3\frac{1}{2}$ bits/sample by additional variable length coding using a buffer of (compressed) field capacity. For video telephone signals, an average rate of 1 bit/sample can be achieved through the addition of movement detection.

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