DOWNLOAD PDF DESIGN OF DIGITAL VIDEO CODING SYSTEMS (SIGNAL PROCESSING AND COMMUNICATIONS, NUMBER 12)

Chapter 1 : Data compression - Wikipedia

*Design of Digital Video Coding Systems: A Complete Compressed Domain Approach (Signal Processing and Communications) [Jie Chen, Ut-Va Koc, KJ Ray Liu] on racedaydvl.com *FREE* shipping on qualifying offers.*

Audio codec and Audio coding format Audio data compression, not to be confused with dynamic range compression , has the potential to reduce the transmission bandwidth and storage requirements of audio data. Audio compression algorithms are implemented in software as audio codecs. Lossy audio compression algorithms provide higher compression at the cost of fidelity and are used in numerous audio applications. These algorithms almost all rely on psychoacoustics to eliminate or reduce fidelity of less audible sounds, thereby reducing the space required to store or transmit them. The acceptable trade-off between loss of audio quality and transmission or storage size depends upon the application. For example, one MB compact disc CD holds approximately one hour of uncompressed high fidelity music, less than 2 hours of music compressed losslessly, or 7 hours of music compressed in the MP3 format at a medium bit rate. A digital sound recorder can typically store around hours of clearly intelligible speech in MB. Lossless compression is unable to attain high compression ratios due to the complexity of waveforms and the rapid changes in sound forms. Many of these algorithms use convolution with the filter [-1 1] to slightly whiten or flatten the spectrum, thereby allowing traditional lossless compression to work more efficiently. The process is reversed upon decompression. When audio files are to be processed, either by further compression or for editing , it is desirable to work from an unchanged original uncompressed or losslessly compressed. Processing of a lossily compressed file for some purpose usually produces a final result inferior to the creation of the same compressed file from an uncompressed original. In addition to sound editing or mixing, lossless audio compression is often used for archival storage, or as master copies. A number of lossless audio compression formats exist. Shorten was an early lossless format. See list of lossless codecs for a complete listing. Some audio formats feature a combination of a lossy format and a lossless correction; this allows stripping the correction to easily obtain a lossy file. Other formats are associated with a distinct system, such as: The lossy spectrograms show bandlimiting of higher frequencies, a common technique associated with lossy audio compression. Lossy audio compression is used in a wide range of applications. In addition to the direct applications MP3 players or computers , digitally compressed audio streams are used in most video DVDs, digital television, streaming media on the internet , satellite and cable radio, and increasingly in terrestrial radio broadcasts. Lossy compression typically achieves far greater compression than lossless compression data of 5 percent to 20 percent of the original stream, rather than 50 percent to 60 percent , by discarding less-critical data. Most lossy compression reduces perceptual redundancy by first identifying perceptually irrelevant sounds, that is, sounds that are very hard to hear. Typical examples include high frequencies or sounds that occur at the same time as louder sounds. Those sounds are coded with decreased accuracy or not at all. Due to the nature of lossy algorithms, audio quality suffers when a file is decompressed and recompressed digital generation loss. This makes lossy compression unsuitable for storing the intermediate results in professional audio engineering applications, such as sound editing and multitrack recording. Coding methods[edit] To determine what information in an audio signal is perceptually irrelevant, most lossy compression algorithms use transforms such as the modified discrete cosine transform MDCT to convert time domain sampled waveforms into a transform domain. Once transformed, typically into the frequency domain , component frequencies can be allocated bits according to how audible they are. Audibility of spectral components calculated using the absolute threshold of hearing and the principles of simultaneous masking â€”the phenomenon wherein a signal is masked by another signal separated by frequencyâ€”and, in some cases, temporal masking â€”where a signal is masked by another signal separated by time. Equal-loudness contours may also be used to weight the perceptual importance of components. Models of the human ear-brain combination incorporating such effects are often called psychoacoustic models. LPC may be thought of as a basic perceptual coding technique: In such applications, the data must be decompressed as the data flows, rather than after the entire data stream has been transmitted. Not all audio codecs can be used for streaming applications, and for such applications a codec designed to stream data effectively will usually be chosen. Some codecs will analyze a longer segment of the data to optimize efficiency, and then code it in a manner that requires a larger segment of data at one time to decode. Often codecs create segments called a "frame" to create discrete data segments for encoding and decoding. The inherent latency of the coding algorithm can be critical; for example, when there is a two-way transmission of data, such as with a telephone conversation, significant delays may seriously degrade the perceived quality. In contrast to the speed of compression, which is proportional to the number of operations required by the algorithm, here latency refers to the number of samples that must be analysed before a block of audio is processed. In the minimum case, latency is zero samples e. Time domain algorithms such as LPC also often have low latencies, hence their popularity in speech coding for telephony. In algorithms such as MP3, however, a large number of samples have to be analyzed to implement a psychoacoustic model in the frequency domain, and latency is on the order of 23 ms 46 ms for two-way communication. Speech encoding[edit] Speech encoding is an important category of audio data compression. The perceptual models used to estimate what a human ear can hear are generally somewhat different from those used for music. The range of frequencies needed to convey the sounds of a human voice are normally far narrower than that needed for music, and the sound is normally less complex. As a result, speech can be encoded at high quality using a relatively low bit rate. If the data to be compressed is analog such as a voltage that varies with time , quantization is employed to digitize it into numbers normally integers. If the integers generated by quantization are 8 bits each, then the entire range of the analog signal is divided into intervals and all the signal values within an interval are quantized to the same number. If bit integers are generated, then the range of the analog signal is divided into 65, intervals. This relation illustrates the compromise between high resolution a large number of analog intervals and high compression small integers generated. This application of quantization is used by several speech compression methods. This is accomplished, in general, by some combination of two approaches: Only encoding sounds that could be made by a single human voice. Throwing away more of the data in the signalâ€”keeping just enough to reconstruct an "intelligible" voice rather than the full frequency range of human hearing. History[edit] Solidyne While there were some papers from before that time, this collection documented an entire variety of finished, working audio coders, nearly all of them using perceptual i. Twenty years later, almost all the radio stations in the world were using similar technology manufactured by a number of companies. Video coding format and Video codec Video compression is a practical implementation of source coding in information theory. In practice, most video codecs are used alongside audio compression techniques to store the separate but complementary data streams as one combined package using so-called container formats. Although lossless video compression codecs perform at a compression factor of 5 to 12, a typical MPEG-4 lossy compression video has a compression factor between 20 and Such data usually contains abundant amounts of spatial and temporal redundancy. Video compression algorithms attempt to reduce redundancy and store information more compactly. Most video compression formats and codecs exploit both spatial and temporal redundancy e. Similarities can be encoded by only storing differences between e. Inter-frame compression a temporal delta encoding is one of the most powerful compression techniques. It re uses data from one or more earlier or later frames in a sequence to describe the current frame. Intra-frame coding , on the other hand, uses only data from within the current frame, effectively being still- image compression. A class of specialized formats used in camcorders and video editing use less complex compression schemes that restrict their prediction techniques to intra-frame prediction. Usually video compression additionally employs lossy compression techniques like quantization that reduce aspects of the source data that are more or less irrelevant to the human visual perception by exploiting perceptual features of human vision. For example, small differences in color are more difficult to perceive than are changes in brightness. Compression algorithms can average a color across these similar areas to reduce space, in a manner similar to those used in JPEG image compression. Highly compressed video may present visible or distracting artifacts. Other methods than the prevalent DCT-based transform formats, such as fractal compression , matching pursuit and the use of a discrete wavelet transform DWT , have been the subject of some research, but are typically not used in practical products except for the use of wavelet coding as still-image coders without motion compensation. Interest in fractal compression seems to be waning, due to recent theoretical analysis showing a comparative lack of effectiveness of such methods. Individual frames of a video sequence are compared from one frame to the next, and the video compression codec sends only the differences to the reference frame. If the frame contains areas where nothing has moved, the system can simply issue a short command that copies that part of the previous frame into the next one. If sections of the frame move in a simple manner, the compressor can emit a slightly longer command that tells the decompressor to shift, rotate, lighten, or darken the copy. This longer command still remains much shorter than intraframe compression. Usually the encoder will also transmit a residue signal which describes the remaining more subtle differences to the reference imagery. Using entropy coding, these residue signals have a more compact representation than the full signal. In areas of video with more motion, the compression must encode more data to keep up with the larger number of pixels that are changing. Commonly during explosions, flames, flocks of animals, and in some panning shots, the high-frequency detail leads to quality decreases or to increases in the variable bitrate. Hybrid block-based transform formats[edit] Processing stages of a typical video encoder Today, nearly all commonly used video compression methods e. They mostly rely on the DCT, applied to rectangular blocks of neighboring pixels, and temporal prediction using motion vectors , as well as nowadays also an in-loop filtering step. In the prediction stage, various deduplication and difference-coding techniques are applied that help decorrelate data and describe new data based on already transmitted data. Then rectangular blocks of residue pixel data are transformed to the frequency domain to ease targeting irrelevant information in quantization and for some spatial redundancy reduction. The discrete cosine transform DCT that is widely used in this regard was introduced by N. In the last stage statistical redundancy gets largely eliminated by an entropy coder which often applies some form of arithmetic coding. In an additional in-loop filtering stage various filters can be applied to the reconstructed image signal. By computing these filters also inside the encoding loop they can help compression because they can be applied to reference material before it gets used in the prediction process and they can be guided using the original signal. The most popular example are deblocking filters that blur out blocking artefacts from quantization discontinuities at transform block boundaries. Entropy coding started in the s with the introduction of Shannonâ€”Fano coding [31] on which the widely used Huffman coding is based that was developed in ; [32] the more modern context-adaptive binary arithmetic coding CABAC was published in the early s. Compression of Genomic Re-Sequencing Data Genetics compression algorithms are the latest generation of lossless algorithms that compress data typically sequences of nucleotides using both conventional compression algorithms and genetic algorithms adapted to the specific datatype. In , a team of scientists from Johns Hopkins University published a genetic compression algorithm that does not use a reference genome for compression.

Chapter 2 : Digital Signal Processing (DSP) - MATLAB & Simulink Solutions - MATLAB & Simulink

Discusses a compressed-domain approach for designing and implementing digital video coding systems, which is drastically different from the traditional hybrid approach.

Definitions[edit] Definitions specific to sub-fields are common. For example, in information theory , a signal is a codified message, that is, the sequence of states in a communication channel that encodes a message. In the context of signal processing , arbitrary binary data streams are not considered as signals, but only analog and digital signals that are representations of analog physical quantities. For example, the words " Mary had a little lamb " might be the message spoken into a telephone. The telephone transmitter converts the sounds into an electrical voltage signal. The signal is transmitted to the receiving telephone by wires; at the receiver it is reconverted into sounds. In telephone networks, signaling , for example common-channel signaling , refers to phone number and other digital control information rather than the actual voice signal. Signals can be categorized in various ways. The most common distinction is between discrete and continuous spaces that the functions are defined over, for example discrete and continuous time domains. Discrete-time signals are often referred to as time series in other fields. Continuous-time signals are often referred to as continuous signals even when the signal functions are not continuous ; an example is a square-wave signal. A second important distinction is between discrete-valued and continuous-valued. Particularly in digital signal processing a digital signal is sometimes defined as a sequence of discrete values, that may or may not be derived from an underlying continuous-valued physical process. In other contexts, digital signals are defined as the continuous-time waveform signals in a digital system, representing a bit-stream. In the first case, a signal that is generated by means of a digital modulation method is considered as converted to an analog signal, while it is considered as a digital signal in the second case. Another important property of a signal actually, of a statistically defined class of signals is its entropy or information content. Analog and digital signals[edit] A digital signal has two or more distinguishable waveforms, in this example, high voltage and low voltages, each of which can be mapped onto a digit. Characteristically, noise can be removed from digital signals provided it is not too large. Two main types of signals encountered in practice are analog and digital. The figure shows a digital signal that results from approximating an analog signal by its values at particular time instants. Digital signals are quantized , while analog signals are continuous. Analog signal An analog signal is any continuous signal for which the time varying feature variable of the signal is a representation of some other time varying quantity, i. For example, in an analog audio signal , the instantaneous voltage of the signal varies continuously with the pressure of the sound waves. It differs from a digital signal , in which the continuous quantity is a representation of a sequence of discrete values which can only take on one of a finite number of values. For example, an aneroid barometer uses rotary position as the signal to convey pressure information. In an electrical signal, the voltage , current , or frequency of the signal may be varied to represent the information. Any information may be conveyed by an analog signal; often such a signal is a measured response to changes in physical phenomena, such as sound , light , temperature , position, or pressure. The physical variable is converted to an analog signal by a transducer. For example, in sound recording, fluctuations in air pressure that is to say, sound strike the diaphragm of a microphone which induces corresponding fluctuations in the current produced by a coil in an electromagnetic microphone, or the voltage produced by a condenser microphone. The voltage or the current is said to be an "analog" of the sound. Digital signal A binary signal, also known as a logic signal, is a digital signal with two distinguishable levels A digital signal is a signal that is constructed from a discrete set of waveforms of a physical quantity so as to represent a sequence of discrete values. Other types of digital signals can represent three-valued logic or higher valued logics. Alternatively, a digital signal may be considered to be the sequence of codes represented by such a physical quantity. Digital signals are present in all digital electronics , notably computing equipment and data transmission. A received digital signal may be impaired by noise and distortions without necessarily affecting the digits With digital signals, system noise, provided it is not too great, will not affect system operation whereas noise always degrades the operation of analog signals to some degree. The resulting stream of numbers is stored as digital data on a discrete-time and quantized-amplitude signal. Computers and other digital devices are restricted to discrete time. Time discretization[edit] Discrete-time signal created from a continuous signal by sampling One of the fundamental distinctions between different types of signals is between continuous and discrete time. In the mathematical abstraction, the domain of a continuous-time CT signal is the set of real numbers or some interval thereof , whereas the domain of a discrete-time DT signal is the set of integers or some interval. What these integers represent depends on the nature of the signal; most often it is time. If for a signal, the quantities are defined only on a discrete set of times, we call it a discrete-time signal. A simple source for a discrete time signal is the sampling of a continuous signal, approximating the signal by a sequence of its values at particular time instants. A discrete-time real or complex signal can be seen as a function from a subset of the set of integers the index labeling time instants to the set of real or complex numbers the function values at those instants. A continuous-time real or complex signal is any real-valued or complex-valued function which is defined at every time t in an interval, most commonly an infinite interval. Amplitude quantization[edit] Digital signal resulting from approximation to an analog signal, which is a continuous function of time If a signal is to be represented as a sequence of numbers, it is impossible to maintain exact precision - each number in the sequence must have a finite number of digits. As a result, the values of such a signal belong to a finite set ; in other words, it is quantized. Quantization is the process of converting a continuous analog audio signal to a digital signal with discrete numerical values. Examples of signals[edit] Signals in nature can be converted to electronic signals by various sensors. The motion of an object can be considered to be a signal, and can be monitored by various sensors to provide electrical signals. A motion signal is one-dimensional time , and the range is generally three-dimensional. Position is thus a 3-vector signal; position and orientation of a rigid body is a 6-vector signal. Orientation signals can be generated using a gyroscope. Since a sound is a vibration of a medium such as air , a sound signal associates a pressure value to every value of time and three space coordinates. A sound signal is converted to an electrical signal by a microphone , generating a voltage signal as an analog of the sound signal, making the sound signal available for further signal processing. Sound signals can be sampled at a discrete set of time points; for example, compact discs CDs contain discrete signals representing sound, recorded at 44, samples per second ; each sample contains data for a left and right channel, which may be considered to be a 2-vector signal since CDs are recorded in stereo. The CD encoding is converted to an electrical signal by reading the information with a laser , converting the sound signal to an optical signal. A picture or image consists of a brightness or color signal, a function of a two-dimensional location. It can be converted to voltage or current waveforms using devices such as the charge-coupled device. A 2D image can have a continuous spatial domain, as in a traditional photograph or painting; or the image can be discretized in space, as in a raster scanned digital image. Color images are typically represented as a combination of images in three primary colors , so that the signal is vector-valued with dimension three. A video signal is a sequence of images. A point in a video is identified by its two-dimensional position and by the time at which it occurs, so a video signal has a three-dimensional domain. Analog video has one continuous domain dimension across a scan line and two discrete dimensions frame and line. The value of the signal is an electric potential "voltage". The domain is more difficult to establish. Some cells or organelles have the same membrane potential throughout; neurons generally have different potentials at different points. These signals have very low energies, but are enough to make nervous systems work; they can be measured in aggregate by the techniques of electrophysiology. Other examples of signals are the output of a thermocouple , which conveys temperature information, and the output of a pH meter which conveys acidity information. Signal processing Signal transmission using electronic signals A typical role for signals is in signal processing. A common example is signal transmission between different locations. The embodiment of a signal in electrical form is made by a transducer that converts the signal from its original form to a waveform expressed as a current I or a voltage V , or an electromagnetic waveform , for example, an optical signal or radio transmission. Once expressed as an electronic signal, the signal is available for further processing by electrical devices such as electronic amplifiers and electronic filters , and can be transmitted to a remote location by electronic transmitters and received using electronic receivers. Signals and systems[edit] In Electrical engineering programs, a class and field of study known as "signals and systems" S and S is often seen as the "cut class" for EE careers, and is dreaded by some students as such. Depending on the school, undergraduate EE students generally take the class as juniors or seniors, normally depending on the number and level of previous linear algebra and differential equation classes they have taken. Time, Frequency, s and z. Since signals and systems are both studied in these four domains, there are 8 major divisions of study. As an example, when working with continuous time signals t , one might transform from the time domain to a frequency or s domain; or from discrete time n to frequency or z domains. Systems also can be transformed between these domains like signals, with continuous to s and discrete to z. Although S and S falls under and includes all the topics covered in this article, as well as Analog signal processing and Digital signal processing , it actually is a subset of the field of Mathematical modeling. The field goes back to RF over a century ago, when it was all analog, and generally continuous. Today, software has taken the place of much of the analog circuitry design and analysis, and even continuous signals are now generally processed digitally. In past EE curricula S and S, as it is often called, involved circuit analysis and design via mathematical modeling and some numerical methods, and was updated several decades ago with Dynamical systems tools including differential equations, and recently, Lagrangians. The difficulty of the field at that time included the fact that not only mathematical modeling, circuits, signals and complex systems were being modeled, but physics as well, and a deep knowledge of electrical and now electronic topics also was involved and required. Students are expected to understand the tools as well as the mathematics, physics, circuit analysis, and transformations between the 8 domains. Because mechanical engineering topics like friction, dampening etc. Dynamical systems that involve noise, filtering and other random or chaotic attractors and repellors have now placed stochastic sciences and statistics between the more deterministic discrete and continuous functions in the field. Deterministic as used here means signals that are completely determined as functions of time. Signals and Systems Current loop - a signaling system in widespread use for process control.

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Chapter 3 : Digital Video Broadcasting for Cable Systems DVB-C

Design of Digital Video Coding Systems is a top-shelf reference for electrical and electronics, signal, image, video processing, computer circuit and systems, digital design, and communication engineers, and an exceptional text for upper-level undergraduate and graduate students in these disciplines.

The following document describes the basic concepts of Digital Signal Processing DSP and also contains a variety of Recommended Reading links for more in-depth information. What is a DSP? Digital Signal Processors DSP take real-world signals like voice, audio, video, temperature, pressure, or position that have been digitized and then mathematically manipulate them. A DSP is designed for performing mathematical functions like "add", "subtract", "multiply" and "divide" very quickly. Signals need to be processed so that the information that they contain can be displayed, analyzed, or converted to another type of signal that may be of use. In the real-world, analog products detect signals such as sound, light, temperature or pressure and manipulate them. From here, the DSP takes over by capturing the digitized information and processing it. It then feeds the digitized information back for use in the real world. It does this in one of two ways, either digitally or in an analog format by going through a Digital-to-Analog converter. All of this occurs at very high speeds. During the recording phase, analog audio is input through a receiver or other source. This analog signal is then converted to a digital signal by an analog-to-digital converter and passed to the DSP. During the playback phase, the file is taken from memory, decoded by the DSP and then converted back to an analog signal through the digital-to-analog converter so it can be output through the speaker system. In a more complex example, the DSP would perform other functions such as volume control, equalization and user interface. Signals may be compressed so that they can be transmitted quickly and more efficiently from one place to another e. Signals may also be enhanced or manipulated to improve their quality or provide information that is not sensed by humans e. Although real-world signals can be processed in their analog form, processing signals digitally provides the advantages of high speed and accuracy. You can create your own software or use software provided by ADI and its third parties to design a DSP solution for an application. For more detailed information about the advantages of using DSP to process real-world signals, please read Part 1 of the article from Analog Dialogue titled: A DSP contains these key components: Stores the information to be processed Compute Engine: Serves a range of functions to connect to the outside world Recommended Reading Digital Signal Processing is a complex subject that can overwhelm even the most experienced DSP professionals. Although we have provided a general overview, Analog Devices offers the following resources that contain more extensive information about Digital Signal Processing:

Chapter 4 : Signal Coding and Processing - Graham Wade - Google Books

This book has some good background material on speech coding plus some material on new speech processing and coding techniques. In order to lay the foundation of speech coding technology the book reviews sampling, quantizations, and then the basic nature of speech signals and the theory and tools applied in speech coding.

And in fact, the impact of digital technology has been rather dramatic. And the indications are that it will be even more so in the future. One of the primary advantages to digital as opposed to analog signal processing techniques is the tremendous flexibility that digital techniques and digital signal processing offers. And because of this flexibility, digital signal processing techniques have found application in a rather large or wide variety of areas. Speech processing, for example, has represented one of the major areas of application of digital signal processing for, at least, the past decade. Both analysis of speech and synthesis of speech rely very heavily on the notions of, for example, digital filtering, other notions, such as the fast Fourier transform algorithm, and a variety of the other digital signal processing techniques and algorithms. More generally, in communication systems, digital signal processing is being used for coding, for multiplexing and, in fact, there is a considerable amount of work being done at present directed toward, basically, replacing all of the present filtering in communications and telephone systems by digital filters instead of analog filters. Seismic data processing represents another very important area in which the flexibility of digital signal processing is very heavily exploited. In fact, seismic and speech processing have probably been the two major catalysts for most of the important developments in digital signal processing. In audio recording and processing, digital signal processing provides an opportunity for some very sophisticated processing and enhancement. And in fact, fairly recently, Professor Thomas Stockham at the University of Utah has been applying some sophisticated digital signal processing techniques to the restoration of old Caruso recordings. The singer, of course, singing into the recording horn. And the output of the recording horn being stored on a recording disk. The problem in that particular application is the fact that the frequency response of the recording horn is not flat. And what this tended to do is give the resulting recording a, sort of, megaphone type of distortion. And one of the objectives in enhancing or restoring some of these old Caruso recordings is to compensate, in a sense, for the frequency distortion introduced by the recording horn. What Professor Stockham has done, basically, is to use digital signal processing techniques to, first of all, estimate the frequency response of the recording horn. And second, to compensate for that frequency response. And all of the work that he carried out was done digitally, primarily, as I indicated previously, primarily to capitalize on the flexibility that digital signal processing offers. And some of the results that he obtained are rather dramatic. And let me just illustrate in a very short passage what some of this has sounded like. What I borrowed from Professor Stockham is a recording of the restoration that he generated on a digital computer. And this particular recording is a two track recording with the original segment recorded on channel 1 and the process segment recorded on channel 2. And that will allow us to switch back and forth between these. The particular piece that is illustrated here is a section from the famous aria "Vesti La Giubba" as sung, of course, by Enrico Caruso. So let me just quickly illustrate this as an example of some of the type of processing that is currently being done using digital signal processing techniques. Let me begin, first of all, by playing a little bit of the original. And then, switch back and forth a few times to present a comparison. So we begin, first of all, with the original. Primarily, the megaphone type of quality in the original has been, essentially, eliminated. Now to go even further in illustrating some of the flexibility of digital signal processing. And so one of the things, obviously, that we would like to do is remove this background noise. In fact, using some rather sophisticated signal processing techniques, Professor Stockham, together with Neil J. Miller, have not only removed the background noise but, with the same processing, removed also the orchestral accompaniment. Now this is, first of all, rather dramatic. Second of all, somewhat useful in the sense that in carrying out a complete restoration one can imagine then redubbing a new orchestra on top of the restored recording. So let me just play a little bit of this to, in fact, show you that it really has been possible to not only remove the background noise but also to remove the orchestral accompaniment. So first let me move forward on the tape to the right place. And on channel 2 is the result of further processing, the restored recording, to eliminate, both, the background noise and also the orchestra. And then, the orchestra removed. Another area in which digital signal processing has tremendous potential is in the area of digital image processing. And in that case, the basic notion is to treat an image as a two dimensional signal digitized, of course. And one is then afforded the possibility of applying digital signal processing techniques to the two dimensional signal. For example, in a very simple signal processing environment, we might be interested in low pass filtering a digital image. For example, if the image has considerable grain noise, grain noise, in fact, tends to be high frequency. And low pass filtering then will tend to reduce or eliminate noise of that type. Or we might be interested in high pass filtering. For example, if we wanted to enhance edges in a picture, the procedure would be or one procedure might be to apply a two dimensional high pass filter. More elaborately, we could consider some processing, which is directed at general image enhancement. Generally, photographically, these are conflicting requirements. And one of the things that we notice about that image is that it has a rather high dynamic range. For example, the snow outside the boiler room is rather bright. The inside of the boiler room is dark. And of course, the contrast inside the boiler room is relatively low because of the fact that the illumination inside the boiler room is relatively low. So one type of processing that we could consider is the simultaneous enhancement of contrast, and reduction of dynamic range, and applying some two dimensional signal processing. You can notice that the dynamic range, in fact, is reduced. The snow is darker than it was in the original. The boiler room is brighter than it was in the original, suggesting reduced dynamic range. But also, the contrast is very clearly enhanced. Just as another example of the same type of processing. And then, a more dimly illuminated area. The details in the right hand corner with the trees and leaves. And as a result of processing to, again, increase the contrast and reduce the dynamic range, we see in the resulting processed image that the detail in the dimly illuminated areas, in fact, is brought out rather dramatically. So this is one example of some rather sophisticated digital signal processing applied to two dimensional signals. And I should mention incidentally that the type of processing that was applied for this image enhancement is discussed in considerable detail in chapter 10 of the text. Now there are, of course, a long list of other applications of digital signal processing. In the biomedical area, for example, digital signal processing techniques are playing a very important role. In radar and sonar, those are two additional areas in which digital signal processing is extremely important. But I will not be assuming any specific background in discrete time signals and systems in Z-transforms, et cetera. In fact, in the next lecture, lecture two, we will begin from the beginning. Namely, we will begin with a definition of discrete time signals and systems. And if you feel a little rusty on basic linear system theory for continuous time systems, it might be well before then to do just a little bit of reviewing. And I suggest some possible books in the study guide. I would also suggest, before beginning the next lecture, that you read the introduction to the text. And perhaps, also browse through the text and the study guide to get a general impression of the scope of the course, and some of the objectives of the course. As I indicated, next time, we will begin with the definition of discrete time signals and systems. I sincerely hope that you find this set of lessons to be interesting and worthwhile.

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Chapter 5 : A Beginner's Guide to Digital Signal Processing (DSP) | Design Center | Analog Devices

A discussion of a compressed-domain approach for designing and implementing digital video coding systems, which is drastically different from the traditional hybrid approach. It demonstrates how the combination of discrete cosine transform (DCT) coders and motion compensated (MC) units reduces power consumption and hardware complexity.

We will address how to efficiently represent multimedia data, including video, image, and audio, and how to deliver them over a variety of networks. In the coding aspect, state-of-the-art compression technologies will be presented. Emphasis will be given to a number of standards, including H. In the networking aspect, special considerations for sending multimedia over ATM, wireless, and IP networks, such as error resilience and quality of service, will be discussed. Current research results in multimedia communications will be reviewed through student seminars in the last weeks of the course. Therefore, this course emphasizes computer assignments and projects. Students will complete individual assignments, and form small groups to complete projects in audio, image and video coding. If you plan to use other programming languages, please discuss it with the instructor. Haskell, Atul Puri, Arun N. Compression Standard, Joan L. Mitchell Editor , William B. Pennebaker Editor , Chad E. Principles and Applications to Speech and Video, N. Representation, Compression, and Standards, Arun N. A component of the homework will be computer assignments. These computer assignments mainly involve building components that may be used later in the projects. The homework policy is as follows: You can discuss the homework problems with any number of students currently taking the course, the TA, and the instructor. However, solutions and solution-techniques should not be exchanged. You should make sure that you understand the solution you turn in, and should of course write up every word of the solution by yourself. For problems which have an answer Problems such as "Find the output" or "Plot the result" , it is OK to compare your final answer with others currently enrolled in the course. But you should fix up any error by your own effort. If these sentences are still vague, just tell yourself "I shall not take undue advantage of any other student" and this should answer other policy-related questions you have in your mind. During the entire semester, do not look at the solutions to any homework or exams of previous years. In order to work out the homework assignments, you should not look at any sources such as books, solutions manuals, papers, and other articles that are not mentioned or handed out in the class. The only exceptions are mathematical tables, and standard texts which you have used in your past career, while taking courses at CMU or elsewhere. Even though this might sound counter-educational, it is not. We believe that it is much more educational to try out a proof or reasoning by yourself, rather than just grab a journal and read it up. In any case, you get the leisure to work on such tutorial exercises by yourself only in a University! All the answers you give must be fully justified. However, results proved in the class, in class-given handouts, in past assignments, or in text can be used without proof, provided you specifically cite the source. Sometimes you will find that a particular homework problem is apparently unrelated to the lectures. This is intentional, the aim being to get acquainted with additional material that cannot be covered in class. Each group will prepare a written report and present a seminar in class. These seminars will be scheduled in the second half of the semester. The final project will be due in the week of May the 17th week. During the week when the project is due, each group will schedule a time slot with the instructor and TA to demonstrate the project. The same policy for the homework should apply to the projects as well. If in doubt, please check with the instructor.

Chapter 6 : Lecture 1: Introduction | Video Lectures | Digital Signal Processing | MIT OpenCourseWare

1 Signal Processing for Wireless Communications and Multimedia: Design, Tools, Architectures Advanced Digital System Design Course , EPF-L Prof. Heinrich Meyr.

Chapter 7 : Lina Karam | iSearch

DOWNLOAD PDF DESIGN OF DIGITAL VIDEO CODING SYSTEMS (SIGNAL PROCESSING AND COMMUNICATIONS, NUMBER 12)

Digital Coding of Waveforms: Principles and Applications to Speech and Video, N.S. Jayant and Peter Noll, Prentice Hall, Englewood Cliffs, NJ Digital Video Processing, A. Murat Tekalp, Prentice Hall, Englewood Cliffs, NJ.

Chapter 8 : Multimedia Communications: Coding, Systems, and Networking

This system transmits an MPEG-2 or MPEG-4 family digital audio/ video stream, using a QAM modulation with channel coding through cable channels. Fig 1 DVB-C Transmitter The input signal is represented as a sequence of standard MPEG transport stream packets.

Chapter 9 : Signal - Wikipedia

Digital Signal Processing (DSP) techniques and methodology have been widely employed in many applications including video/audio/data communications and networking, medical imaging and computer vision, speech synthesis and coding, digital audio and video, and control of complex systems and industrial processes.