FASAC Technical Assessment Report

NON-US DATA COMPRESSION AND CODING RESEARCH

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ABSTRACT

This assessment of recent data compression and coding research outside the United States examines fundamental and applied work in the basic areas of signal decomposition, quantization, lossless compression, and error control, as well as application development efforts in image/video compression and speech/audio compression. Seven computer scientists and engineers who are active in development of these technologies in US academia, government, and industry carried out the assessment.

Strong industrial and academic research groups in Western Europe, Israel, and the Pacific Rim are active in the worldwide search for compression algorithms that provide good tradeoffs among fidelity, bit rate, and computational complexity, though the theoretical roots and virtually all of the classical compression algorithms were developed in the United States. Certain areas, such as segmentation coding, model-based coding, and trellis-coded modulation, have developed earlier or in more depth outside the United States, though the United States has maintained its early lead in most areas of theory and algorithm development. Researchers abroad are active in other currently popular areas, such as quantizer design techniques based on neural networks and signal decompositions based on fractals and wavelets, but, in most cases, either similar research is or has been going on in the United States, or the work has not led to useful improvements in compression performance. Because there is a high degree of international cooperation and interaction in this field, good ideas spread rapidly across borders (both ways) through international conferences, journals, and technical exchanges. Though there have been no fundamental data compression breakthroughs in the past five years—outside or inside the United States—there have been an enormous number of significant improvements in both places in the tradeoffs among fidelity, bit rate, and computational complexity. Several international standards for compression based on mature technologies have been adopted recently, and strong competitors for the future standards based on newer algorithms are emerging in several parts of the world.

In the United States and abroad, compression algorithms should continue to improve at a slow pace, as algorithms are optimized for specific applications or made more generally robust, as new hybrids combining better features of various approaches are developed, and as promising but complicated algorithms are simplified and refined. The best future non-US (and US) systems are likely to incorporate improvements in all compression system components, including signal decomposition by transform or subband/wavelet filtering, quantization and bit allocation, entropy coding, and error control coding. As data volumes continue their exponential growth, compression of some form will become an implicit component everywhere of most systems that store, communicate, or process data.

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NON-US DATA COMPRESSION AND CODING RESEARCH
TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

LIST OF FIGURES AND TABLES

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FOREWORD

This report, *Non-US Data Compression and Coding Research,* is one in a series of technical assessment reports produced by the Foreign Applied Sciences Assessment Center (FASAC), operated for the Federal government by Science Applications International Corporation (SAIC). These reports assess selected fields of foreign basic and applied research, evaluate and compare the state of the art in the country or area of interest with US and world standards, and identify important trends that could lead to future applications of military, economic, or political importance. This report, like others produced by the Center, is intended to enhance US knowledge of foreign applied science activities and trends, to help reduce the risk of technology transfer, and to provide a background for US research and development decisions. Appendix C lists the FASAC reports completed and in production.

This report was prepared by a panel of internationally recognized scientists and engineers who are active in data compression and coding research:

• Dr. Robert M. Gray (Panel Chairman)	Professor of Electrical Engineering Information Systems Laboratory, and Acting Vice-Chairman Department of Electrical Engineering Stanford University
• Dr. Martin Cohn	Senior Research Associate/Senior Lecturer Computer Science Department Brandeis University
• Dr. Larry W. Craver	Senior Electronics Engineer Department of Defense
• Dr. Allen Gersho	Professor/Director Center for Information Processing Research Electrical & Computer Engineering Department University of California/Santa Barbara
• Dr. Thomas Lookabaugh	Vice President Research and Business Development DiviCom
• Dr. Fabrizio Pollara	Technical Group Leader Communication Systems Research Section Jet Propulsion Laboratory/Pasadena

- Dr. Martin Vetterli — (Acting) Associate Professor
  Department of Electrical Engineering and
  Computer Sciences
  University of California/Berkeley, and
  Associate Professor of Electrical Engineering
  Department of Electrical Engineering
  Columbia University

On a part-time basis over the period from August 1992 to April 1993, each panel member devoted a substantial amount of time assessing published foreign literature on data compression and coding research.

EXECUTIVE SUMMARY

Data compression is becoming an increasingly important and ubiquitous component of systems that store, communicate, or process information, including systems that deal with speech or video signals, digital images, telemetry data, and facsimile transmissions. This assessment evaluates recent data compression research performed at academic, industrial, and government institutions located outside the United States. Based primarily on expert critical reading of technical literature published during the past five years (including refereed journals, conference proceedings, technical reports, and popular technical and scientific magazines) and direct personal contact with non-US researchers through technical meetings and visits, the assessment considers significant recent non-US accomplishments and trends in fundamental theory, algorithms, and applications.

Research on compression draws on ideas from many technical disciplines, including communication and information theory, signal processing, optimization, statistics, and algorithmic complexity. Contemporary developers of compression algorithms borrow methods freely from traditional and recent statistical and mathematical research areas, including clustering, neural nets, simulated annealing, linear prediction, dynamical systems, fractal geometry, and wavelet expansion. Most of this work is done by electrical engineers, with significant minority participation by statisticians, mathematicians, and (recently) computer scientists. Because the range of contributors and applications is so broad, research on compression is reported in an unusually wide variety of publications.

This assessment examines the following basic components of general compression systems:

- Signal decomposition,
- Quantization,
- Lossless coding, and
- Error control coding.

An additional component—sampling—is not discussed, because the maturity of its theory and uniformity of its practice discourage active research.

The assessment also examines two important and challenging application areas:

- Speech and audio compression, and
- Image and video compression.

Compression is commonly applied to transmission and storage of other copious sources of information, including remote sensing systems and medical diagnostic equipment, but the quest for systems that compress signals that are heard or seen by human observers dominates research and commerce.

Pervasive themes that run through many of the detailed topical discussions that follow include:

- Steady—cumulatively dramatic—progress in the development of compression algorithms and implementations has provided and continues to provide steadily increasing fidelity, steadily declining bit rates, and steadily declining compression system computational or hardware complexity. Excellent contributions to this evolution have come from Western Europe, Israel, and the Pacific Rim (especially Japan), as well as the United States. The primary goals of these ongoing efforts are the development of simpler, cheaper, faster systems with quality comparable to the most sophisticated systems currently available, and the development of systems that (at acceptable cost) provide extremely high fidelity of the kind desired for applications like high-definition television or digital radiography. Progress is certain to continue.

- Some compression techniques, such as segmentation coding, model-based coding, and coded modulation, have been developed earlier or in more depth outside the United States, particularly in Western Europe. Europe also leads the United States in audio (as distinct from speech) compression technology. Non-US researchers have been quite active in other currently popular areas, including quantizer designs based on neural networks and signal decomposition based on fractals and wavelets, but, in most of these cases, either work of equal or better quality is done in the United States, or the work has not produced significant performance improvements. There is rough parity among Western Europe, Japan, and the United States in compression product development. The United States maintains a lead or is competitive in almost all areas of theory and algorithm development.

- Because there is a high level of international interaction and cooperation (through conferences, visits, and publications) in data compression research and development, good ideas tend to spread fast and technical leads tend to be short-lived. The primary exception is proprietary algorithms developed by industrial institutions, which restrict their publication.

- Eastern Europe is practically invisible in the data compression field, with the exception of theoretical work done in Moscow, Budapest, and Prague. Researchers from that part of the world publish virtually no original work on compression algorithms for speech, audio, images, video, or telemetry.

- The techniques embodied in international standards for data compression are generally quite good, and almost all commercial hardware and software products for data compression comply with existing standards. Notable examples include the joint ISO/CCITT[1] JPEG[2] standard for still image compression (based on the well-understood fast discrete-cosine transform for signal decomposition, uniform scalar quantization, and lossless coding) and code-excited linear prediction algorithms for speech compression systems, which now provide quality at low bit rates that would have seemed incredible 10 years ago. Much international research in data compression is driven by fierce competition among developers of algorithms to have their products incorporated into the next generation of standards, in the hope that the successful competitor will enjoy a lead in fielding commercial products based on the new standard.

- In general, Western Europe and the Pacific Rim have been competitive with the United States in research aimed at enhancing and improving interna-

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1 International Standards Organization/International Consultative Committee on Telephone and Telegraphy

2 Joint Photographic Experts Group

tional standards and in developing the best candidates for future standards. Although the number of possible combinations of signal decomposition, quantization, lossless coding, and error control is very large, specific combinations of techniques have become either *de facto* or official international standards for important applications (against which other combinations are judged). In a few areas, such as audio standard development, the primary advances and leadership have come from outside the United States.

- Despite occasional claims in the popular and scientific press, there have been no significant breakthroughs or major surprises in data compression—anywhere in the world—in the past five years. For example, though there have been significant advances in the mathematics of wavelets (especially in France), these advances have not led to corresponding improvements in compression systems relative to what has been achieved using the related, older quadrature mirror filter subband codes. Though novel neural-network-based compression codes (some of the most interesting of which were developed in England and Finland) have useful structural properties, this has not yet led to significant improvements in performance or complexity. It has, however, added to the growing variety of choices that may provide good matches to specific problems. For example, work in Finland, the United Kingdom, and elsewhere suggests that quantizer designs based on Kohonen's algorithm are particularly well suited to the design of compression codes that are robust against bit errors in transmission and storage. Fractal-based compression is highly touted in the United Kingdom, the United States, and, to some extent, in Norway, but to date it has proved capable of providing compression comparable only to the better standard methods—at the price of greatly increased encoding complexity. The very real gains realized internationally during the last five years have come primarily from variations on and improvements of existing methods. Algebraic geometry codes originally developed in the former Soviet Union have created much interest in the area of error control coding, but it may be several years before such codes translate into useful products.

In reading the discussions that follow of current research and development activities outside the United States, the reader should be aware that during recent years the primary research activity outside the United States has been in algorithm development and implementation. The basic theoretical approaches to data compression have reached maturity and stability after 40 years of development since their formulation by Shannon and Bennett at AT&T Bell Laboratories. Many open problems remain, but they have thus far resisted solution, and progress has been slow in extending the classical results to more general code structures and source models. In contrast, algorithm development has been explosive worldwide.

Most of the research on fundamentals and on algorithms concentrates on individual components of compression systems: signal decomposition, quantization, lossless coding, and error control coding. This focus is generally reasonable, because both theory and practice demonstrate that improvements in individual component performance lead directly to improvements in overall system performance. Much of the discussion of non-US data compression research and development work that follows is grouped into sections corresponding to important compression system components. The issues of overall system design will be considered under the applications headings.

Signal Decomposition

The primary non-US research on subband filter banks is currently taking place in Australia, Belgium, Brazil, France, Japan, Norway, and Taiwan. The most important practical application of such work is the recent development of the MUSICAM audio coding standard in Western Europe using subband filters designed to match human hearing. Although comparable research is conducted in the United States, the development of applications is more advanced in Western Europe. Similar applications matching filter design to perceptual criteria for image compression have been developed in Switzerland.

Important research on wavelet-based subband decompositions in specific combination with quantization has been done in France and Germany. Although several groups have claimed significantly superior performance for wavelet-based subband coding over traditional transform coding using the discrete cosine transform, the traditional methods still dominate practice and standards, because of their good performance, relative maturity, and simplicity.

Germany, Poland, and France lead the worldwide search to develop better basis functions and filter bank implementations for the decompositions, including low-complexity finite-precision implementations, quantitative measures of performance when structural constraints are imposed, and deeper understanding of the underlying mathematics. Work also continues, especially in Japan, on means of speeding up the traditional discrete cosine transform. France remains the leader in fundamental theoretical studies of wavelet decompositions, but activity and quality are spreading throughout technically advanced countries.

Lossless Coding

Israel, Russia, Finland, and Japan dominate recent theoretical work in universal coding and modeling. Israelis, often with US co-authors, have been a powerful influence on ergodic theory and its practical consequences for lossless coding for over two decades. Russia has a strong tradition in probability, information theory, and ergodic theory, but practical design has not been emphasized. Western Europe, including Finland, emphasizes algebraic sources such as automata and semigroups and, along with Israel, has developed the combinatoric properties of lossless codes. Japan is more a follower than a leader in fundamentals, but increasingly applies theory to knowledge systems and artificial intelligence problems.

The current emphasis in code design is on source estimation, on codes that obey restrictions, and computational enhancement. The primary non-US research is found in Israel and Japan, along with Australia, Canada, and New Zealand. Western Europe dominates research on cryptology and lossless coding. Lossless coding enhances security and compressors can analyze the security of key streams.

A variety of application-specific lossless codes have been researched outside the United States, including image and numerical data in Japan and word/phrase/document storage and retrieval in Australia, Canada, Finland, and Israel. Text and character compression remain the primary application, but applications to more sensitive data such as medical images and scientific data have been gaining importance in Japan.

Despite very active efforts in this area outside the United States, US developers lead in most practical aspects of lossless compression, including new hardware and architectures for faster compression.

Quantization

The best non-US theoretical work relating to the fundamental performance bounds and convergence properties of quantizers and, more generally, source coding is done in Hungary, Russia, and Israel, with fewer but notable contributions from Japan and Western Europe. Although this work does not immediately suggest improved algorithms and implementations, it does provide insight into the basic performance limitations for general compression systems, and it is likely that some of these ideas will at least guide future design methods and code structures.

Researchers in Western Europe (especially Switzerland, Germany, France, and Italy) and the Pacific Rim (especially Japan and South Korea) are as active (at all quality levels) as their US colleagues in quantization and related lossy compression technique development. Their work has included many variations, extensions, and hybrids of the traditional techniques of predictive coding, block truncation coding, and transform coding. They have also investigated optimized vector quantizers with constrained structure, and systems that match the quantization to the specific signal decomposition used (for example, a variety of transforms and subband [wavelet] filter banks). Little new fundamental theory has been developed. The work of the best non-US institutions is competitive with US work in this area.

Error Control Coding

The two error control coding areas that are receiving the most international attention at present were both developed outside the United States. Coded modulation, a technique for achieving nearly optimal error control over noisy narrow band communication channels, was developed at IBM–Zurich. Algebraic geometry codes, a new class of codes (currently primarily of theoretical interest) that promises performance better than existing codes once efficient coding and decoding algorithms are developed, were developed at two institutes in Moscow. West European, Japanese, and US developers have rapidly caught up in coded modulation, which has revolutionized modem design and is incorporated in the best currently available high-speed modems. US developers now lead in producing hardware systems incorporating coded modulation. Denmark, France, and Japan have been increasingly active in algebraic geometry code research. US researchers have been slow to follow suit, partially due to skepticism about the eventual performance and practicality of error control coding systems based on algebraic geometry codes.

Speech and Audio Compression

The primary non-US work in speech coding is being done in Japan, France, the United Kingdom, Italy, Germany, Canada, and Norway. Smaller efforts can be found in The Netherlands, Spain, Sweden, Denmark, Belgium, South Korea, Australia, Singapore, Ireland, Hong Kong, Brazil, Australia, China, Taiwan, Turkey, Greece, India, and elsewhere. The substantial ongoing engineering research and development in speech coding are motivated by the move to digital transmission for cellular mobile systems and other wireless applications. Very good work in the development and testing of speech coding algorithms and implementations is taking place in geographically dispersed locations, particularly in Western Europe, Japan, and Canada. Relatively little of this work is of a fundamental and innovative nature compared to the dominance of innovative algorithmic approaches originating in the United States. Much of the non-US work consists of a large variety of complexity reduction methods and minor algorithm modifications with slight quality improvements over the more basic versions of the algorithms previously developed in the United States. A few very innovative and imaginative coding algorithms or components of coding algorithms have appeared in the foreign literature in recent years, but none have had a major impact on the current state of the art in speech coding.

In contrast, the best work in wideband audio coding has come largely from outside the United States, and primarily from Western Europe, particularly Germany, The Netherlands, and France. Standardization of speech (and audio) coding algorithms is the driving force for a large fraction of the activity in speech and audio coding.

Japan (Nippon Telephone and Telegraph) leads the world in the area of compressed speech quality assessment, the development of objective measures of speech quality that are good predictors for subjective appraisals of quality.

Image and Video Compression

Currently, international image compression standards are almost entirely based on the relatively old technique of discrete cosine transform coding combined with lossless coding, while research has focused on competing systems with subband or wavelet signal decomposition, vector quantization, and additional lossless compression in cascade. French and German developers are particularly active in subband and wavelet coding systems, though several US groups are competitive in application of these ideas. Swiss and Japanese developers have pioneered alternatives based on image segmentation or modeling. These computationally complex methods can provide a higher degree of image compression than any other known technique. Canadian, British, Dutch and US developers have recently been producing work in this area comparable in quality (if not quantity) to the Swiss and Japanese work. Significant reductions in complexity will be required before such techniques will be serious candidates for useful implementation.

Western Europe, especially The Netherlands, Belgium, and Denmark, and Japan are the present leaders in research on compression of medical images and sensor signals for use in hospital information systems. This work is motivated by the desire to minimize the expensive image and signal acquisition device costs by making the results widely available through digital storage and transmission networks.

Japan and Western Europe, especially Switzerland, have led in research on high-definition television, but the recent change in emphasis to digital high-definition television in the United States and the resulting importance of digital compression techniques have resulted in a marked increase in compression research for commercial video. Japan and South Korea have been the closest competitors in video compression research and development for lower-bandwidth systems such as twisted wire and digital line video telephone. Japan, South Korea, and The Netherlands lead in compression for digital video recording, and Japan is the primary source of research on compression for filmless cameras producing digital images for disk storage.

There is dramatic activity overseas in application-oriented research, both in universities and in industrial research laboratories. Research oriented at generating good implementations of image and video compression standards and the development of high-volume products, such as digital video cassette recorders, digital still cameras, video phone and videoconferencing, and others, is as active abroad as in the United States. As a result, the technology lead implied by fundamental research efforts in the United States is less likely to result in leads in manufacturing and selling products based on image and video compression technology.

Although there is always the possibility for breakthroughs and surprises, the general trends of research and development are likely to continue to provide a steady improvement of performance and a reduction of computational complexity in software and cost in hardware. The best work outside the United States is almost certain to be produced at those institutions already active or by researchers trained at those institutions who then start new efforts. Expanding university research efforts are particularly visible in speech and image compression in South Korea, Hong Kong, Taiwan, and other Asian countries.

The best lossless compression algorithms will become faster, cheaper, and ubiquitous in many communication, storage, and computing environments. There are not likely to be any significant improvements in compression ratios, because these codes already are operating near their theoretical limits. The theory of signal decompositions, especially transforms and filter banks such as wavelets, will continue to advance, and implementations of decompositions will become simpler and cheaper in hardware and software. It is not likely, however, that these advances alone will yield significant improvements in compression, as the existing decompositions are already quite good. Optimized quantization algorithms will improve, but simpler alternatives will remain competitive in many applications because of their lower cost, in spite of inferior performance. Error control coding methods have stabilized for the next few years, but, in the long run, algebraic geometry codes may provide significant improvements in reliability and robustness to transmission and storage bit errors.

The major improvements in compression in the near future are most likely to come from improved engineering of overall systems, matching the diverse components of compression and finding good overall balances of system complexity and cost with system performance. It is expected that research activity outside the United States will continue to grow with a continuing emphasis on speech, audio, image, and video applications and an increasing emphasis on multimedia applications.

CHAPTER I
ASSESSMENTS

A. OVERVIEW

Overwhelming acceleration of the rate of production of digital information during the past two decades has prompted widespread research and development on data compression techniques: methods for speeding data transmission, and reducing data storage requirements without loss of important information. Currently important sources of large volumes of information include remote sensing systems, speech and audio signals, digital images for medical and entertainment applications, facsimile transmission systems, photographs, multispectral imaging systems, radiometric systems, and video signals. Compression of information makes more efficient use of storage and communication resources and (in some applications) eases the computational burden of subsequent signal processing. Typical compression systems include components that perform one or more of the following operations:

- Lossless compression: compression of a file in a perfectly reversible fashion by using knowledge about the different frequencies of occurrence of different symbols;

- Quantization: mapping of continuous or large discrete spaces into relatively small discrete spaces;

- Signal decomposition: decomposition of a signal into different components (such as frequency bands or basis functions) so that each can be separately quantized;

- Error control: use of error-control coding techniques (in addition to or in combination with compression) to prevent catastrophic reconstruction errors resulting from bit errors encountered in communication or storage.

These fundamental aspects of compression are each given a chapter in this report and a section in this assessment, along with additional chapters and sections for the two primary applications of data compression: speech/audio compression and image/video compression.

This assessment has not uncovered non-US research thrusts substantially different from US activities. Dominance of the field by US journals and conferences makes unnoticed advances by non-US researchers unlikely. It is not surprising that countries like the United States and Japan, with enormous installed computing bases and device-manufacturing capabilities, will be most active in implementation, while researchers in less well endowed countries (like Russia, or even the United Kingdom) will be drawn to theory and specific applications. The present flood of computer science and engineering graduate students to the United States from China, India, Pakistan, and the Pacific Rim, make it likely that research will eventually prosper in those places (if the students actually return home).

In the near future, it is very likely that all aspects of compression will begin to take advantage of the increasing availability of massively parallel computation. Parallel engines are evolving and becoming more widely available in Japan and Western Europe as well as in the United States. Some archival compression tasks can be subdivided readily, as can array computations (such as those used extensively in image processing). The expected increases in speed will facilitate real-time video compression.

The theoretical roots of compression and virtually all of the classical compression algorithms (including Huffman coding, optimal quantization, predictive coding, and transform coding) were developed in the United States, but there are now strong industrial and academic research groups throughout Western Europe and the Pacific Rim. The United States maintains a lead in many areas of theory and algorithm development, although certain techniques (such as segmentation coding, model-based coding, and trellis-coded modulation) have been developed earlier or in more depth at non-US institutions.

Active cooperation among US and non-US researchers, dominance of the field by a few US-based publications, and regular interaction between US and non-US researchers at international conferences tend to diminish early national leads in application of good ideas quickly once the ideas are announced. A common interest in developing, adopting, and improving international standards has also tended to equalize the quality and subject matter of research and development worldwide. However, diverse efforts to develop novel algorithms and implementations continue to challenge the dominant techniques, and the standards are likely to continue to evolve in the direction of new techniques that provide better tradeoffs between performance (the quality of the decompressed data at a given bit rate) and cost. This report and this summary chapter assess many promising (and some not so promising) non-US research and development activities in data compression.

Several general themes recurred in our examinations of various specific topics. They can be distilled into the following observations:

- There has been steady progress in development of algorithms and implementations that provide better fidelity, lower bit rates, and lower computational or hardware complexity. This progress will continue.

- There have been no significant breakthroughs or major surprises in the past five years.

- International standards are generally quite good, but serious competition for the next generation of standards exists.

- The United States leads or is competitive in most (but not all) theoretical areas; US and non-US capabilities in product development are more equal.

B. LOSSLESS COMPRESSION

Lossless data compression utilizes two major paradigms: *statistical* coding, which estimates input statistics and replaces frequent patterns with short codes, infrequent patterns with longer codes; and *dictionary* methods, which code patterns by citations to their earlier occurrences. The former method is exemplified by arithmetic and Huffman coding, the latter by Lempel-Ziv coding.

Researchers from the Commonwealth nations of Australia, Canada, and New Zealand are prominent in estimating input statistics, as well as in losslessly compressing large bodies of text, such as data bases, concordances, and natural-language corpora. Israeli researchers are also prominent in compression of large bodies of text.

West Europeans, especially the French and Finns are prominent in treatment of linguistic data.

Theoretical aspects of lossless compression, including work on the classical problems of classification and estimation, are dominated by Israel, Russia, Finland, Japan, and Western Europe, in that order. Work on practical aspects of coding—the design of new algorithms and the modification of existing algorithms—is concentrated in Japan, Israel, Australia, Canada, and New Zealand (with little work in Russia or Western Europe).

Transmission-error-resistant coding is a research focus in Japan. Interesting work on the application of lossless compression to images and medical data (often in tandem with a lossy stage) is going on in Japan and Western Europe. The integration of error control is modest up to now, but certain to grow. Great Britain, India, China, South Korea, Taiwan, and Singapore are conspicuously inactive in this area, considering its potential importance to their commerce.

The enhancement and implementation of cryptographic systems by compressors are well known and likely to continue to be an incentive for further research. Notable work connecting compression and cryptography is found in Western Europe.

Arithmetic coding achieves the best compression for sources with low-order dependencies; Lempel-Ziv coding achieves the highest compression for higher-order sources, and is typically faster in both hardware and software. The quest for speed in lossless compression is driven by two major applications: user-transparent compression, as applied in computer operating systems, and real-time video compression. The most important speed improvements are expected to come from advances in device technology and architecture, especially systolic and parallel processing. The United States seems almost alone in practical aspects of this hardware, while European researchers are contributing to the theory.

The use of parallel searching and string matching can certainly accelerate lossless compressors for all applications. A number of clever compromises have tweaked more speed out of arithmetic coders without seriously diminishing compression, but, among L/Z compressors, apart from Welch's decade-old version, there has been incremental but not dramatic acceleration. Something like probabilistic hashing or associative retrieval might yield still more speed with negligible performance loss, but improvements in device speeds will likely be dominant.

A national or international standard for lossless text compression would stimulate the market for, and hence the development of, hardware realizations. Such standard devices could then be built into disk drives, data-transfer buses, and modems, permitting the universal exchange of data in less time, bandwidth, and space. Likewise, the adoption of standards for reversible compression would facilitate the inclusion of compression in commercial cryptographic protocols, yielding enhanced security along with the usual advantages of economy.

Inductive tools like neural nets, genetic algorithms, and fuzzy logic are likely to see increasing use in solving the matching problems and in designing the contents of dictionary-based lossless compressors. Present algorithms, such as those of Lempel and Ziv, Elias, and Ryabko, already employ more conventional heuristics to populate their dictionaries. None of them can be proven "ideal," and schemes involving inductive tools may prove to have advantages in convergence rate or other measures of performance.

C. QUANTIZATION

Non-US development activity in quantization algorithms emphasizes traditional scalar quantization combined with predictive coding, transform coding, or block truncation coding; it is comparable in quality and content to analogous work in the United States. Non-US developers in this field have devised a variety of minor improvements and hybrid techniques in recent years, but have made no more significant breakthroughs in terms of implementation complexity or distortion/rate tradeoffs than have their US colleagues. Foreign and US activities in vector quantization techniques are also of comparable quality and scope. These include applications of Hopfield and Kohonen neural net ideas to the design and implementation of vector quantizers. In both scalar and vector quantization algorithm development, the primary non-US work is being done in Japan, South Korea, Germany, the United Kingdom, and Italy, with notable smaller efforts in Taiwan, Hong Kong, France, and Spain.

Western and Eastern Europe, Israel, and Japan have made significant contributions to the theoretical aspects of quantization research. However, recent non-US developments in the traditional asymptotic theories due to Shannon (information theory) and Bennett (high-rate quantization theory) have been primarily of mathematical interest, and have not suggested new fundamental insights or supported fundamental breakthroughs in code design. Research outside the United States has led to no theoretically motivated practical improvements comparable to lattice quantizer and trellis-coded quantizer improvements recently demonstrated in the United States.

The area of segmentation coding has been considerably more active outside the United States. A variety of promising results have been obtained by Swiss and Canadian researchers in this area, which has received little attention in the Pacific Rim or in the United States. These codes are computationally intensive, but they yield the highest verifiable compression ratios reported in the literature.

Non-US research on fractal-based compression techniques has largely followed US work and, as in the United States, has been promoted chiefly in the popular scientific press (with occasional tutorials in technical journals). The United Kingdom has been the primary site of non-US activity on fractals for image compression, although, recently, good work has been reported in Norway. Most papers referring to fractal coding are descended from M. Barnsley's "iterated function system," or IFS approach. An alternative technique called "fractal geometry codes" was introduced by researchers from IBM-Israel in 1986. These codes are simpler than the IFS approach and are more in the spirit of Mandelbrot's original fractal geometry work in that they use the idea of a "traveling yardstick" to fit a piecewise-linear approximation to one-dimensional data. These codes have received little attention in the literature, but the simplicity of the technique and its good performance and theoretical performance bounds that compare well with simulation all suggest that the technique may be well suited to low-complexity applications.

It seems unlikely that there will be major breakthroughs (in the United States or abroad) in the theory or practice of quantization in the next five years comparable to those seen during the last 10 years, which included refinement of transform codes into JPEG, development of a variety of vector quantization algorithms, achievement of high compression ratios using segmentation codes, development of powerful and promising subband and wavelet codes, development of simple lattice and trellis quantizers that are effective against overload quantization noise, and intelligent use of signal processing techniques (such as prediction and classification) to improve performance in any quantization algorithm. Non-US developers, like those in the United States, are likely to continue to make minor improvements in a variety of algorithms, and to find that various combinations of the general approaches are particularly suitable for certain applications. As in the United States, non-US researchers who study the nonlinear theory that describes the behavior of quantizer error and quantifies the bit rate/distortion tradeoffs in quantization systems are likely to continue slow development of that theory and to use it to suggest new architectures, as those who studied Shannon and Bennett theories once did. Surprising breakthroughs are always possible, but it seems unlikely that they will come from any of the currently known methods (including the fads).

D. SIGNAL DECOMPOSITIONS

Research on signal decompositions useful for data compression is active worldwide. The focus is on finding better transforms, subband and wavelet expansions, on reducing implementation complexity, and on building complete compression systems taking advantage of inherent features of the decomposition. Good non-US work is done in Europe (France, Germany, England, Scandinavia, Belgium, Switzerland), the Far East (Japan, South Korea, Taiwan), Australia, and Brazil.

Some key ideas in subband coding and wavelets have come out of Europe. In particular, France has been at the forefront of the work on wavelets, but mostly from a theoretical point of view rather than applications. Several groups are active in filter design for subband coding in Europe (France, Belgium, Norway, Switzerland) and in the Far East (Japan, Taiwan).

Compression of speech, music, images, and video based on various signal decompositions is active worldwide, with interesting contributions from Europe (France, Germany, Norway, The Netherlands, Switzerland) and the Far East (Japan, South Korea). The compression work done abroad is comparable to the work done in the United States.

No major changes are expected in the near future. Good theoretical work will continue, for example, in France. Good applications work, especially related to complete systems, will be done, for example, in Germany and The Netherlands. The Far East, while producing probably less theoretical work, will lead in implementations. The United States will remain in a good position in new ideas and theory, and in high-end applications.

E. SPEECH AND AUDIO COMPRESSION

Substantial engineering research and development in speech coding are being done outside the United States, much of it motivated by the move to digital transmission for cellular mobile systems and other wireless applications. Very good work in the development and testing of speech coding algorithms and implementations are taking place in geographically dispersed locations, particularly in Western Europe, Japan, and Canada. Relatively little of this work is of a fundamental and innovative nature compared to the dominance of innovative algorithmic approaches originating in the United States. Much of the non-US work consists of a large variety of complexity reduction methods and minor algorithm modifications with slight quality improvements over the more basic versions of the algorithms previously developed in the United States. A few very innovative and imaginative coding algorithms or components of coding algorithms have appeared in the non-US literature in recent years, but none have had a major impact on the current state of the art in speech coding. In fact, there is a major gap between academic and industrial research in most countries. The primary non-US work in speech coding is being done in Japan, France, the United Kingdom, Italy, Germany, Canada, and Norway. Smaller efforts can be found in The Netherlands, Spain, Sweden, Denmark, Belgium, South Korea, Australia, Singapore, Ireland, Hong Kong, Brazil, Australia, China, Taiwan, Turkey, Greece, India, and elsewhere.

In contrast, the best work in wideband audio coding has come primarily from Western Europe, particularly Germany, The Netherlands, and France. Standardization of speech (and audio) coding algorithms is the driving force for a large fraction of the activity in speech and audio coding.

The action in speech coding is at steadily decreasing bit rates. For example, since adoption of the ADPCM[1] standard of 32 kb/s in 1983, research on such high-rate coding has more or less died. Adoption by CCITT[2] of a 16-kb/s algorithm (which may also become the standard for personal communication system applications) severely dampened research activity at this bit rate (other than implementation activities related to the g.728 standard). Recently, the RPE LPC[3] standard has specified a source rate of 13 kb/s, and the VSELP[4] standard has specified a source rate of just 8 kb/s. Emerging mobile applications are using second-generation "half-rate" digital speech compression, with rates half of the above values.

Following the imminent "half-rate" standards for speech coding in Europe and Japan, future efforts in the next five years are likely to concentrate on coding algorithms for 2.4 kb/s, with some continued activity at 4.8 and 8 kb/s for some specialized applications. Increased focus on vocoder approaches is expected for 2.4 kb/s due to the inherent limitation of code-excited linear prediction and other algorithms at this rate, essentially waveform coding methods. Research is likely to become increasingly dispersed as more Asian countries (South Korea, Hong Kong, China, Malaysia, etc.) expand their university-based research in speech coding. We see little indication of increased emphasis on basic research worldwide, although some significant and highly innovative work could possibly come out of academic institutions.

F. IMAGE AND VIDEO COMPRESSION

Non-US research in image and video compression has tended to the most part to be of comparable quality to that in the United States, but lagging in key areas such as transform coding, frequency decompositions, motion compensation, and vector quantization. Two notable exceptions are boundary and segmentation-based coding and model-based coding, both of which originated and are largely championed outside the United States. Model-based coding especially represents an important long-term research direction.

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1 adaptive differential pulse-coded modulation

2 International Consultative Committee on Telephone and Telegraphy

3 regular pulse excitation linear predictive coding (Groupe Speciale Mobile)

4 vector -sum-excited linear prediction (Telephone Industry Association)

Non-US industrial concerns have shown their ability, however, to generate good applied research aimed at commercializing compression technology, particularly in the area of consumer electronics. There is little indication that any US lead in basic compression research will manifest itself in significant economic reversals.

Little research is in progress in the United States or abroad on compression algorithms specifically tailored towards binary images such as facsimile or images produced by halftoning gray-scale images. Halftoning itself is a form of compression, but again little research has appeared in the literature. Error diffusion and dithering techniques remain the dominant approach to halftoning, and generally accepted standards exist for binary image compression, both lossy and lossless, that are not being seriously challenged by new methods.

With the growth of Picture Archiving and Communication Systems and Hospital Information Systems, there is a great deal of research and development aimed at developing algorithms, validation procedures, communication and storage systems, and standards for medical images. Many believe that the future will see nearly all-digital radiology departments, with the one possible exception being analog-acquired X-rays. So far, most compression schemes considered have been lossless, but there is a growing effort to develop lossy schemes such as transform codes, especially for archiving, recall, and educational purposes. It is unlikely that lossy images will be used for primary diagnosis for several years, but the use of Computer-Aided Diagnosis techniques combined with compression may someday change this. The large majority of the work on medical image compression and on PACS in general is taking place in Japan and in Western Europe, especially in The Netherlands, Belgium, and Denmark. The driving force is to make maximal use of existing imaging centers and ease the sharing of images among such centers and the hospitals, clinics, and individual practitioners that they serve.

Geographically, Japan and Western Europe have the largest concentration of research on general image and video compression systems, with many university and industrial laboratories participating. South Korea and Taiwan also show a surprising amount of research activity for their size. Pockets of important research activity can also be found in other countries, such as Australia, Singapore, and Canada.

The United States conducts leading research in most areas of image and video compression fundamental techniques. The most significant exception is model-based coding, a technique that is being actively pursued in Japan and Europe, but which will not likely see important application in the next 10 years.

In application-oriented research, however, there is dramatic activity overseas, both in universities and in industrial research laboratories. Research oriented at generating good implementations of image and video compression standards and the development of high-volume products, such as digital video cassette recorders, digital still cameras, videophone and videoconferencing, and others, is as active abroad as in the United States. As a result, the technology lead implied by fundamental research efforts in the United States is less likely to result in leads in manufacturing and selling products based on image and video compression technology.

Facsimile compression will be guided primarily by groups that will be able to directly influence international standards. While this will promote interoperability of equipment, it will, and apparently has, stifled innovative research. Thus, for this decade, most compression techniques are expected to use standards except for minor variations in proprietary implementations. The future use of compression standards for facsimile is uncertain since it will be affected by the continued development of Integrated Services Digital Network (ISDN) and the coming High-Definition Digital Video standard. However, proprietary techniques will continue to compete with international standards, especially among large-scale vendors who have a high probability of communicating with their own equipment.

Because investigations have centered on different aspects of compression, it seems likely that if facsimile images can be segmented accurately into different types of features, then the most promising solution is to compress each feature separately with the most suitable compression algorithm. Such features as text, line drawings, and gray-scale images could each be compressed with optimum procedures.

G. RELIABLE COMMUNICATION

In spite of the fact that US research in coding theory has been strong and productive, the two major developments in the last decade, trellis-coded modulation and algebraic geometry codes, originated in foreign countries, Switzerland and Russia. Both developments were unexpected, if not fortuitous. What was not unexpected is that the most theoretical development happened within the strong school of information and coding theorists that exists in Russia today, and the more practical development happened in a research laboratory of a large US company (IBM), albeit based in a foreign country (Switzerland).

Research in trellis-coded modulation for bandwidth-limited channels started after the publication (1982) of work by G. Ungerboeck, a researcher at IBM-Zürich, and has since then received considerable attention in Europe and Japan, as well as in the United States. Significant non-US contributions to theoretical and practical aspects of trellis-coded modulation were made by researchers in Western Europe (mainly, Germany, Switzerland, and Italy). The development of trellis-coded modulation had enormous practical implications due to its applicability to high-speed data transmission over bandlimited channels, such as the common twisted pair telephone link, in an era when the demand for data exchange between computers is continuously increasing. The fact that the discovery happened abroad did not impede its quick assimilation by US researchers and industry, since IBM was the original sponsor of Ungerboeck's research, and AT&T Bell Labs and Codex Corporation (Motorola) rapidly made their own contributions to this field. Several smaller companies, both in the United States and abroad, were also able to include trellis-coded modulation in their modem products, since the implementation is not particularly complex. The need for modem standardization led to inclusion of these techniques into CCITT standards. IBM-Zürich is currently still at the forefront of research in high-speed modems. Other contributions from industry come from DLR and AEG in Germany, and Alcatel and Telecom in France. Japan (T. Kasami at Osaka University and I. Oka at Osaka City University), Italy (a group of researchers at the Politecnico of Torino and the University of Pisa), ETH-Zürich (Federal Institute of Technology) in Switzerland, and Australia (B. Vucetic at the University of Sydney) made several important theoretical contributions.

Research in algebraic geometric codes, which are currently only of theoretical interest, is based on the seminal work of three researchers in the former Soviet Union, M. Tsfasman, S. Vladut, and T. Zink, and received later contributions by researchers in Europe, Japan, India, and China. The original research in the former Soviet Union was done at the Problems of Information Transmission Institute (USSR/Russian Academy of Sciences) and at the Central Economics and Mathematics Institute, both in Moscow. The main contributors in Europe are J. Justesen's group at the University of Denmark, another group at Aarhus University, R. Pellikaan at Eindhoven University, D. Lebrigand at the University of Paris, D. Rotillon at the University of Toulouse (which is also the location of workshops on algebraic coding theory), and M. Perret and G. Lachaud, who work for the French Centre National de Recherché Scientifique (CNRS). Several Japanese universities have researchers who contributed to this field. US researchers did not appear to show much interest in this field until recently, perhaps due to a yet unclear practical advantage of these codes.

The most active area of research in trellis-coded modulation will probably be the construction of trellis codes with the same gain of those already known, but with additional useful properties (for example, rotational invariance, ease of synchronization, variable rate applications). In this respect, the use of isometries will be the most promising tool for such constructions. These methods have been recently introduced in the United States and are pursued actively in Western Europe and in Italy, in particular. The most challenging area of research in algebraic geometric codes will be to find fast decoding algorithms that decode to the full error-correcting capability of the codes, to find better and less complex derivations of the codes, and possibly to find similar binary codes that exceed the Gilbert-Varshamov bound. The likelihood for success in this field is currently centered in Denmark, France, and Russia.

The future of coding research promises more developments in new applications than in theoretical breakthroughs. Progress in non-US applications of coding will be made most noticeably in Japan, Germany, and France. The situation will be similar to the historical collaboration between Sony and Philips that led to the inclusion of Reed-Solomon codes in the compact-disk recording technology. Non-US high-definition television research will also be a driver for new developments in coding for compressed data.

Error-correcting codes are finding increasing application in a broad range of systems developed in the United States, Europe, and Japan, as hard disks for computer storage and audio compact discs. The use of error-correcting codes on hard disks can render some of the defective sectors usable and the trade-off between the redundancy of the code, the cost of implementing error correction, and the fraction of the disk restored to use can be advantageous. Similarly, compact disk technology, pioneered by Sony in Japan and Philips in Europe, contains very sophisticated signal processing that includes Reed-Solomon coding and interleaving to protect against burst errors caused by surface scratches. In addition, error correction techniques are being applied, more widely in the United States than in Japan, to memory chips to overcome defects and improve process yield.

When new applications for coding are found, the codes need to be modified to take into account special constraints. This is a situation where countries having large volumes of data exchange and processing, notably, the United States, will generate the need for new research. In this respect, Russian research will probably remain more theoretical. As much as applications can be the driver for new developments, advances in microelectronics will also play a major influence on the design of future coding and decoding algorithms. In this respect, Japan and the United States will probably take the lead. In general, error correction algorithms will become more and more an integral part of an overall signal processing scheme rather than a separate and specialized component.

CHAPTER II
INTRODUCTION

A. DATA COMPRESSION

Data compression is the conversion of a signal, such as speech, an image, or a set of sensor measurements, into a representation requiring less digital storage space to save (and less digital communication system capacity to transmit) than does the original signal. Data compression allows faster transmission of data through digital communication links and more efficient use of digital storage devices. It makes practical a wide range of applications, from transmission of high-definition television (HDTV) over constrained-bandwidth channels to storage and transmission of massive quantities of image data in modern medical center radiology departments. In some applications, compression is an important prelude to other signal processing algorithms; by reducing the number of bits required to describe a signal, further signal processing such as enhancement, classification, detection, and estimation can be greatly simplified (if the relevant information has not been lost in the compression process).

Most data compression systems have three identifiable components that accomplish sampling, quantization, and digital compression, respectively. Some of these components may not be required for certain types of input signals, and sometimes the operations of the components may overlap or be combined.

- Sampling

  A sampling operation converts a continuous-time signal into a discrete-time signal by extracting the signal values at a discrete set of points. For example, sampling a speech signal 8,000 times each second or sampling a photograph at a square grid of 256 x 256 points (called *picture elements* or *pixels* or *pels*) converts these one- or two-dimensional waveforms into sequences or matrices of data points. The Nyquist theorem demonstrates that—provided one samples fast or finely enough—no information need be lost, that is, that the original signal can be reconstructed from the samples with arbitrary accuracy. A separate sampling is not necessary if the input signal already consists of values at a discrete set of points, as it does in the output of a sampling sensor or in a digital image created by computerized tomography (CT) or magnetic resonance imaging (MRI).

- Quantization

Quantization maps analog numbers describing signal values at the sample points onto a defined set of possible outputs. Since the outputs are chosen from a finite set, they can be specified by a finite sequence of binary integers (bits). A component or system that maps an input signal by sampling, quantization, and coding into a binary representation is often referred to as an *analog-to-digital converter* (*ADC* or *A/D converter*). Decoding of the digital signal to reconstruct a replica of the original waveform inverts the procedure, to perform a digital-to-analog conversion (DAC or D/A conversion).

The most common type of quantizer performs a scalar mapping of scalar real values onto a finite set; this variety of ADC is called *pulse-coded modulation (PCM)*.[1] More generally, quantization is the mapping of any space into a finite space. If the input space is a multidimensional vector space, the quantizer is called a *vector quantizer* (VQ). Quantizer input spaces may also be discrete. For example, digitally acquired CT scan may originate as a raster of pixel intensities each taking on one of 2^{12} possible values (12 bits per pixel or bpp).

- Digital Compressor

The function of a digital compressor is to map discrete spaces into other discrete spaces so that the output spaces specify (allow reconstruction of) the input signals using fewer bits than the original input signals contained. This allows the output signals to be stored in smaller files or communicated more rapidly or over narrower bandwidths than can the raw input signals. The mapping can be performed in a variety of ways, including the simple application of a second, specialized quantizer. In such cases, the second and

1 B. M. Oliver, J. Pierce, and C. E. Shannon, "The Philosophy of PCM," *Proc. IRE, 36*, 1948, 1324–1331.

third stages (quantization and compression) are often combined, at least in theoretical treatments, and treated as a single mapping. Most practical implementations, however, have separate stages to perform initial quantization and true compression.

PCM has revolutionized communication. In trade for the small amount of distortion introduced by a high-rate quantizer, the resulting signal can be communicated over a noisy channel virtually error free by using digital error control techniques. By accepting this small amount of controlled error in the overall communication, one protects against uncontrollable and potentially large random errors caused by noise in the communication channel (due to thermal noise, atmospheric noise, fading, interference, etc.).

Sampling and quantization to perform ADC expands the effective bandwidth of the original signals (although mathematically there has still been data compression, since an analog signal of infinite resolution has been reduced to a digital signal of finite precision). For example, if a telephone-quality speech signal occupying a 4-kHz bandwidth is sampled at the Nyquist rate, quantized to 8 bits per sample, and then transmitted using a simple binary modulation technique such as phase-shift keying (PSK, in which each binary symbol is modulated by a cosine or a phase-shifted cosine), the bandwidth of the resulting signal is increased from 4 to approximately 64 kHz! This disadvantage is usually outweighed by the advantages of digital communication and digital signal processing, but highlights the need for an explicit compression step subsequent to quantization to achieve practical compression in many applications.

Practical data compression systems frequently include additional stages that carry out a variety of other signal processing operations, including various forms of preprocessing and postprocessing before or after sampling, quantization, or compression. The most common type of preprocessing is some form of signal decomposition by linear transform or linear filtering. The goal of such decompositions is typically to concentrate energy, to extract important features, or to decorrelate the signal. Finally, compression may be followed by or combined with error control coding to protect the bit stream against errors in transmission and storage. A block diagram depicting a general compression system is shown in Figure II.1.

Decompression reverses the process, except, of course, for the quantization step that can only be approximately reconstructed.

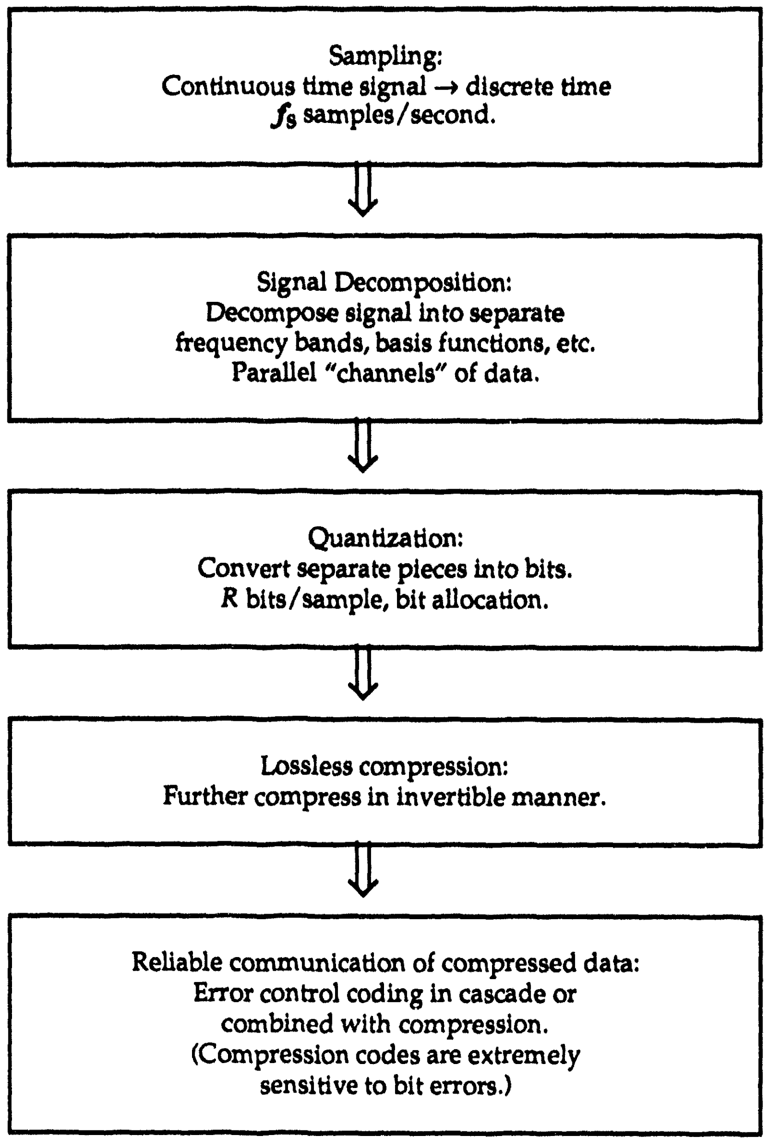

Figure II.1
Basic Components of Data Compression

B. LOSSLESS COMPRESSION

There are two quite distinct types of compression: lossless compression and lossy compression. Lossless compression, which also goes by names like noiseless coding, entropy coding, invertible coding, and data compaction, permits perfect reconstruction of the original signal. One of the oldest examples of lossless coding is the Morse code, which converts the 26 letters of the English alphabet, the 10 Arabic digits, and various other symbols into binary words (or vectors) that are short for the often used symbols and long for the rarely used symbols. The average length of these binary words is less than if all letters, numbers, and symbols were given equal-length binary codes, as they are in the ASCII (American Standard Code for Information Interchange). Another old but still popular lossless encoding technique is run-length coding, where bits are saved when storing and transmitting black and white documents by counting the number of times 1s (corresponding to black) or 0s (corresponding to white) occur in runs. If the binary data contain long runs of one or the other symbol (for example, the white space in an ordinary letter), such a representation can greatly reduce the number of bits required to reproduce the document.

The best known modern examples of lossless codes are Huffman codes, Lempel-Ziv codes, and arithmetic codes. To meet the demands of perfect reconstruction, these codes are limited in their ability to compress data. (Commercially available lossless coding products typically promise an average of 2:1 compression of arbitrary computer files, which they sometimes deliver.) However, because perfect reconstruction is necessary in applications like compression of computer programs or transmitting bank data, these codes are very popular. In fact, the words "data compression" in the computer science literature virtually always mean lossless compression.

Lossless codes are mature and highly developed. The literature on the theory and design of such codes is extensive, and a variety of commercial and public-domain software products and commercial hardware products are available.

General treatments of lossless codes may be found in books by Lynch,[2] Bell et al.[3] and Storer.[4] A brief survey of the basic types of lossless compression may be found in the work by Gersho and Gray,[5] and a treatment of lossless coding in the context of Shannon information theory can be found in the book by Cover and Thomas.[6] Lossless coding research addresses two primary topics:

- What are the fundamental limits on performance; that is, what is the maximum compression that can be achieved while maintaining perfect reconstruction?

- How does one design fast codes that perform in a nearly optimal fashion, that is, close to the theoretical limits?

The first question is answered in terms of Shannon's information theory. Digital signals are characterized by a measure of their complexity called entropy, and signals can be losslessly compressed only to the point where the average number of bits per sample is close to the entropy, and essentially no farther.[7] Many code constructions perform close to the Shannon bound when the statistical characteristics of the input signal are known in advance. Theory and practice continue to seek means of finding implementable codes that can perform well on arbitrary input signals.

Lossless compression is only possible when the original signal is digital. An analog-source signal, such as a common TV signal or the audio output of a microphone, must first be converted (in an inevitably lossy fashion) to a digital signal before it can be compressed in a lossless fashion. In theoretical terms, truly analog sources have infinite entropy.

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2 T. J. Lynch, *Data Compression: Techniques and Applications*, Belmont, Calif: Lifetime Learning, Wadsworth, 1985.

3 T. C. Bell, J. G. Cleary, and I. H. Witten, *Text Compression*, Englewood Cliffs, NJ: Prentice Hall, 1990.

4 J. Storer, *Data Compression*, Rockville, MD: Computer Science Press, 1988.

5 A. Gersho and R. M. Gray, *Vector Quantization and Signal Compression*, Boston: Kluwer Academic Publishers, 1992.

6 T. M. Cover and J. A. Thomas, *Elements of Information Theory*, New York: Wiley, 1991.

7 C. E. Shannon, "A Mathematical Theory of Communication," *Bell System Tech. J.*, *27* (1948), 379–423, 623–656.

C. LOSSY COMPRESSION

As the name implies, lossy compression results in a representation that contains less information than the original (and therefore in a reconstruction that is not identical to the original). If properly done, however, the differences are imperceptible, or have little or no effect on the intended application of the reconstructed signal. In some applications, perceptible and even severe degradation may be tolerable. For example, even poor-quality compressed representations might permit effective, rapid browsing of a data base containing many megapixel visual image files.

The oldest lossy compression technique is simple quantization. In the crudest version, binary quantization or hard limiting, all numbers above a certain threshold (for example, all positive numbers, above the threshold 0) are mapped into one reproduction number, a, and all numbers below the threshold (in the example, all negative numbers) are mapped into another reproduction number, b, usually $b = -a$. Only one binary number is required to represent each output reproduction number: 1 implies a, and 0 implies b. Clearly, much quality is lost if this representation is used for an arbitrary string of real numbers. Methods for choosing the best a, b, and threshold, and extension of this technique to non-binary quantizers, are the classical focus of the theory of quantizer design, begun in papers by Lloyd[8] and Max.[9] As one allows more and more representation levels, input signals can be reconstructed to ever increasing accuracy. Use of additional levels, however, means more bits must be used for each quantized sample. This tradeoff of bits against quality is fundamental to the theory and design of quantizers: research intended to quantify the tradeoffs between bit count and distortion in quantizers with a large number of bits was initiated by Bennett.[10]

In practice, the quest for optimality is often tempered by a desire for simplicity of implementation. Simple techniques, notably, uniform quantizers that approximate

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8 S. P. Lloyd, "Least Squares Quantization in PCM," unpublished Bell Laboratories Technical Note, 1957. Portions presented at the Institute of Mathematical Statistics Meeting, Atlantic City, NJ, Sept 1957. Published in the Mar 1982 special issue on quantization of the *IEEE Trans. Inform. Theory*.

9 J. Max, "Quantizing for Minimum Distortion," *IEEE Trans. Inform. Theory*, (1960), 7–12.

10 W. R. Bennett, "Spectra of Quantized Signals," *Bell Systems Tech. J.*, 2, (Jul 1948), 446–472.

real numbers by a finite number of equally spaced reproduction values, have dominated both practice and the supporting theory.

As has been noted previously, simple quantization alone can expand rather than compress data. Often, it does not provide sufficient compression to meet application requirements. Hence, it is generally combined with further compression by cascading it with separate lossless or additional lossy compression stages. Combination of scalar quantization with entropy coding is simple and extremely popular. More sophisticated techniques can improve the bit rate versus distortion tradeoff. The designer must decide whether or not the performance improvement is worth the added complexity. The remainder of this section describes several of the principal popular approaches to lossy compression. These categories of compression techniques often overlap, and they are often combined with lossless compression.

Many traditional compression techniques combine simple quantization with other signal processing methods that attempt to remove unneeded information from the signal, or to decompose the original signal into components that can be efficiently compressed separately. Detailed descriptions of the techniques are provided by Jayant and Noll[11] and in Chapters 1–8 of Gersho and Gray.[12] More specialized treatments for speech compression can be found in O'Shaughnessy,[13] and for image compression in Netravali and Haskell[14] and Rabbani and Jones.[15] Davisson and Gray[16] and Jayant[17] contain many of the classic papers in the area.

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11 N. S. Jayant and P. Noll, *Digital Coding of Waveforms*, Englewood Cliffs, NJ: Prentice-Hall, 1984.

12 Gersho and Gray, 1992, op. cit.

13 D. O'Shaughnessy, *Speech Communication*, Reading, Mass: Addison-Wesley, 1987.

14 A. N. Netravali and B. G. Haskell, *Digital Pictures: Representation and Compression*, New York: Plenum Press, 1988.

15 M. Rabbani and P. W. Jones, *Digital Image Compression Techniques, Tutorial Texts in Optical Engineering*, Bellingham, Wash: SPIE Optical Engineering Press, 1991.

16 L. D. Davisson and R. M. Gray, *Data Compression, 14*, Benchmark Papers in Electrical Engineering and Computer Science, Stroudsburg, Penn: Dowden, Hutchinson & Ross, 1976.

17 N. S. Jayant, Editor, *Waveform Quantization and Coding*, Piscataway, NJ: IEEE Press, 1976.

1. Predictive Coding

Instead of quantizing the signal samples themselves, one can compute errors between predicted (from past quantized values) and actual next-sample values, and then quantize that error. This strategy is based on the assumption that the residual errors will have less variation than the original signal, and hence will be easier to quantize. Alternatively, prediction removes the "redundancy" from the signal and permits the quantizer to focus on new information. This approach is embodied in techniques called *differential PCM* (DPCM), delta modulation, and predictive quantization. In each of these, the quantization operation is still performed on scalars, but one attempts to first remove information redundant with past reconstructed values. Reconstruction combines prediction with correction by the quantized residual to accomplish reproduction of the original signal. A variation on this idea is to code only a subset of the available samples and then interpolate the missing samples using straight lines, polynomials, or splines.

2. Transform Coding

Prior to quantization, signals can be transformed using any of many available linear transforms, including Fourier, Karhunen-Loeve, Hotelling, discrete cosine, Hadamard, Walsh, Hartley, and wavelet. The basic strategy is to concentrate the important information in a few transform coefficients and to reduce correlation or memory in the quantized signal. For example, input signals such as speech and images will often have similar statistical behavior over a wide collection of samples, but appropriate transforms can produce sample coefficients with well-differentiated behavior; lower-order coefficients often have more energy and are more important for accurate reconstruction. When the transformed coefficients are quantized, more bits can be assigned to the quantizers of certain coefficients than to others, allowing optimal bit allocation.

Although many transforms (including all those mentioned at the beginning of the preceding paragraph) have been considered in the research literature, the discrete cosine transform (DCT) dominates in practice, because it is simple and can approximate many of the special qualities of more complex transforms.[18] The

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[18] D. R. Rao and P. Yip, *Discrete Cosine Transform*, San Diego: Academic Press, 1990.

ISO/CCITT p x 64, JPEG, and MPEG standards are all based on a cascade of DCT coding and lossless compression. Decompression algorithms must invert both the compression and the transform to obtain the final reproduction.

A general survey of applications of transform coding to image compression may be found in Clarke.[19]

3. Subband Coding

Subband codes are a close cousin of transform codes. Subband codes decompose time- or space-domain signals into frequency-domain signals. The input signal is decomposed into separate frequency bands, and the resulting bandpass signals are downsampled to preserve the total number of samples. The samples are separately quantized, with attention paid to bit allocation (as in transform coding). The primary difference between transform codes and subband codes is that the former perform matrix multiplication on disjoint blocks of data, while the latter apply filters or sliding-block codes to overlapping blocks of data. Wavelet-based codes can be viewed as either transform codes (operating on large blocks of data) or as subband codes, applying short filtering operations through the large blocks. Decompression algorithms for subband codes pass the quantized data through an inverse filter bank to reconstruct the original signal. Originally developed for speech applications, subband codes are now popular for image compression as well.[20]

4. Block Truncation Coding

The earliest one-bit-per-sample image compression scheme having reasonably good quality was Delp and Mitchell's block truncation code (BTC).[21] The original scheme was a binary quantizer (one bit per sample) that scaled the threshold and the reproduction so as to force the first two moments, the mean and variance, of the reproduction to match those of the original. A quantized mean and variance had to be transmitted as side information to do the scaling, but, for a large image, the bit rate was only slightly larger than one bit per sample. The scheme remains popular in

19 R. J. Clarke, *Transform Coding of Images*, Orlando, Fla: Academic Press, 1985.

20 J. W. Woods, *Subband Image Coding*, Boston: Kluwer Academic Publishers, 1991.

21 E. J. Delp and O. R. Mitchell, "Image Compression Using BTC," *IEEE Trans. Commun., COM-29*, (1979), 1335–1342.

the research literature, primarily because of its simplicity and good ability to reproduce edges. A number of variations have appeared that generalize the basic property that the reconstructed image preserves the lower-order moment (like mean and variance) of the original image.

5. Segmentation Codes

Segmentation codes divide an input signal, such as a full image, into subsets exhibiting common behavior. For example, an image might be decomposed into pieces corresponding to background-relatively-constant subimages, edges, and highly textured pieces. The segments might be chosen with the constraint that they have common shapes (squares, for example), or they might be chosen without such constraints. Separate quantizers or other codes are then applied to each subset of images. Typically, these processors extract edges or contours and interpolate to form low-resolution approximations of the subimages. If the reproduction quality is insufficient, residual errors can then be coded by any number of methods. Segmentation codes are also called "two-source" or "two-component" or "second-generation" coding. These codes have yielded some of the highest compression ratios reported and verified (100:1 and higher), but they come at the cost of significant computational complexity. As computing power becomes faster and cheaper, these algorithms will become more competitive.[22]

6. Multi-Resolution and Hierarchical Codes

These codes construct a representation of the input signal that has several layers, successive layers providing reproductions of increased resolution (and fidelity) in time or space by sampling faster or more densely. The successive reproductions are specified in terms of quantized parameters that describe each higher-resolution layer in terms of changes from the next lower-resolution layer. The classical example of this technique is the Burt-Adelson pyramid coder,[23] but the same general approach has also been combined with transform codes, subband codes, and coding schemes that decompose input signals into wavelets.

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22 M. Kunt, M. Benard, and R. Leonardi, "Recent Results in High Compression Image Coding," *IEEE Trans. Circuits, Systems, CAS-34,* (Nov 1987), 1306–1336.

23 P. J. Burt and E. H. Adelson, "The Laplacian Pyramid as a Compact Image Code," *IEEE Trans. Commun., COM-31,* (Apr 1983), 552–540.

7. Vector Quantization

The quantizers considered so far perform conversions from analog to digital (or from high-rate digital to low-rate digital) on scalars, that is, on individual real numbers. It is a basic tenet of the Shannon theory of information[24] that one can do better in the sense of having the best-quality representation for a given average bit rate if one codes vectors (groups of samples) as units instead of separately as individual scalars. Further theoretical support comes from extension of Bennett's asymptotic high-rate quantization theory to vectors, as pioneered by Shutzenberger[25] and Zador.[26] The achievable performance of scalar quantizers, special cases of vector quantizers constrained to look at only one sample at a time, should clearly be less than that of vector quantizers that are free of this constraint.

Transform codes are one form of vector quantizer, operating on vectors of samples before quantizing individual coefficients. A variety of other techniques can also map real vectors into binary words, trading off bit rate and quality in effective or near-optimal fashion. Methods for development of vector quantizers (including the special case of scalar quantizers) fall into two general classes: lattice-based codes and codes designed by clustering. The former confine the possible reproduction values to the points of a regular lattice, as does the uniform quantizer mentioned above; the latter try to optimize the quantizer for a particular set of signals, as do the Lloyd-Max quantizers mentioned above. In addition to vector generalizations of the

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24 Shannon, 1948, op. cit.

C. E. Shannon, "Coding Theorems for a Discrete Source with a Fidelity Criterion," *IRE National Convention Record, Part 4*, 1959, 142–163.

R. G. Gallager, *Information Theory and Reliable Communication*, New York: John Wiley & Sons, 1968.

T. Berger, *Rate Distortion Theory*, Englewood Cliffs, NJ: Prentice-Hall Inc., 1971.

R. M. Gray, *Source Coding Theory*, Boston: Kluwer Academic Publishers, 1990.

25 M. P. Schutzenberger, "On the Quantization of Finite Dimensional Messages," *Information and Control, 1*, (1958), 153–158.

26 P. L. Zador, "Development and Evaluation of Procedures for Quantizing Multivariate Distributions," PhD Dissertation, Stanford University, 1963.

P. L. Zador, "Topics in the Asymptotic Quantization of Continuous Random Variables," Unpublished Bell Laboratories Memorandum, 1966.

P. L. Zador, "Asymptotic Quantization Error of Continuous Signals and the Quantization Dimension," *IEEE Trans. Inform. Theory, IT-28*, (Mar 1982), 139–148.

Lloyd algorithms (often called k-means in the statistical literature), modern techniques from neural networks have also been applied to the design and implementation of vector quantizers. Hopfield neural nets[27] minimize a quadratic functional and hence can be used to perform a minimum distortion or nearest neighbor search through a codebook. The codebook itself can be designed using backpropagation and other neural net design techniques such as competitive learning. Kohonen's clustering algorithm of "self-organizing feature maps"[28] resembles the original k-means clustering algorithm and Widrow's LMS (least mean squares) algorithm: a learning set is used to iteratively update the nearest neighbor selected as well as its close neighbors.

As with scalar quantization techniques, vector quantization techniques can be combined with prediction, transforms, and other signal decompositions, as well as with segmentation compression codes. They can also be cascaded with lossless codes. Discussions of vector quantization are provided in Jayant and Noll, [29] in Gersho and Gray,[30] in O'Shaughnessy (for speech),[31] and in Netravali and Haskell[32] and Rabbani and Jones (for images).[33] A collection of papers on the topic can be found in Abut.[34]

8. Fractal Codes

No compression algorithms have received as much attention in the popular press in recent years as those based on fractals. Mandelbrot's fractals (short for "fractional dimension") have had an impact on various aspects of image processing during recent years, particularly on image creation in computer graphics. Their replication properties have made them famous for creating coastlines and planet

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27 D. W. Tank and J. J. Hopfield, "Simple Neural Optimization Networks: An A/D Converter, Signal Decision Circuit and a Linear Programming Circuit," *IEEE Trans. Circuits, Systems, CAS-33*, 5(1986), 533–541.

28 T. Kohonen, *Self-Organization and Associate Memory*, 3rd Ed., Springer, 1989.

29 Jayant and Noll, 1984, op. cit..

30 Gersho and Gray, 1992, op. cit.

31 D. O'Shaughnessy, *Speech Communication*, Reading, Mass: Addison-Wesley, 1987.

32 Netravali and Haskell, 1988, op. cit.

33 M. Rabbani and P. W. Jones, *Digital Image Compression Techniques, Tutorial Texts in Optical Engineering*, Bellingham, Wash: SPIE Optical Engineering Press, 1991.

34 H. Abut, Editor, *Vector Quantization*, Piscataway, NJ: IEEE Press, May 1990.

surfaces in science fiction films. Many have seen in the popular press the intricate fern leaves and daisy fields that can be generated from low-complexity fractal descriptions. Given their undisputed ability to create complex images (of certain types) from simple descriptions, fractals have long been thought to have significant potential for image compression applications: if one can invert the image creation procedure and find for an arbitrary image a simple fractal description, then enormous compression could be achieved. Following Barnsley's original proposal to use fractals for image compression,[35] compression ratios such as 10,000:1 were frequently mentioned in the press during 1988–1989, compression at least two orders of magnitude better than the best high-complexity compression algorithms existing. Media hype has made it somewhat difficult to unravel the claims and verify the performance of compression systems based on fractals, a situation aggravated by the fact that the company making many of the claims keeps its basic algorithms proprietary. (It has even attempted to copyright the words "fractal transform".)

Two quite distinct approaches to data compression have adopted the name "fractal." The first approach is due to Barnsley and his disciples and is increasingly becoming known as iterated function systems (IFS), IFS coding, or attractor coding. The goal of an IFS encoder is to find for a given image a family of affine mappings that can approximate the image using a few subblocks, plus mappings that, when iterated, construct the rest of the blocks. The second approach was developed by Wallach and Karnin of IBM Science and Technology in Israel[36] and is based on Mandelbrot's fractal geometry. Rather than iterating block mappings as in an IFS system, the fractal geometry codes are essentially one-dimensional and code successive scan lines by using a "traveling yardstick" mechanism to construct low-rate piecewise-linear approximations for the sequence of pixel intensities. The technique is computationally simple, and the theory of fractal geometry leads to simple predictions of the code performance as a function of the complexity of an image, predictions that agree well with simulations. The apparent quality on ordinary monochrome images at bit rates of around .5 bits per pixel is quite respectable, although not as good as more sophisticated schemes.

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35 M. F. Barnsley and S. Demko, "Iterated Function Systems and the Global Construction of Fractals," *Proc. Royal Soc. London, A399,* (1985), 243–275.

M. F. Barnsley, *Fractals Everywhere,* New York: Academic, 1988.

36 E. Wallach and E. Karnin, "A Fractal Based Approach to Image Compression," *Proc. ICASSP,* (1986), 529–532.

Most of the lossy compression techniques can be considered as "quantization" in the general sense of converting a continuous or large discrete space into a relatively small discrete space, and Chapter IV of this assessment deals with fundamental research on these techniques and their supporting theory.

D. ADAPTIVE AND UNIVERSAL COMPRESSION

Much of the theory and design of compression systems is based on assumptions about the statistical nature of the input signals. This can allow compression systems to take advantage of the particular structure of a data source and theoretical studies to compare the performance of compression system designs with the optimal possible performance. Unfortunately, most important information sources do not have generally accepted complete statistical models. As a result, an active area of compression system theory has been extension of performance bounds to sources that are incompletely specified, and an active area of algorithm design has been development of compression schemes that do well for a variety of signal types. Two general approaches have been widely studied: *adaptive coding* and *universal coding*. Adaptive coding continually "learns" the necessary structural parameters of the input signals, and slowly changes its code parameters to better match those inputs. Universal coding constructs a collection of different compression algorithms for distinct signal types and then selects the best available system. The two approaches are variations on a common theme, but they have led to different types of practical implementations. The classical example of an adaptive code is a uniform quantizer with a step size that changes according the long-term variance of the input signal. The classical example of a universal code is the Rice machine, based on a collection of distinct Huffman codes. All of them are tried, and then the one yielding the smallest bit rate over a long time interval is picked for final use. Adaptive and universal codes play a major role in both lossless and lossy compression, and in both theory and practice.

E. SIGNAL DECOMPOSITION

The popular lossy compression techniques of transform and subband coding along with the newer extensions of multiresolution and wavelet coding systems all involve the decomposition of a signal into separate pieces that are individually coded. These separate pieces may be extracted from the original by Fourier or other transform analysis, by temporal or spatial filtering, by downsampling, and by other signal processing techniques. As a result, the compression application has fueled the development of the theory and implementation of signal decomposition algorithms, especially those that result in useful data structures or are computationally efficient or seem well suited to combination with quantization (or combinations of these attributes). Chapter V of this assessment is devoted to such decompositions and their role in compression.

F. SPEECH AND IMAGE COMPRESSION

Although many data types can be usefully compressed for storage and transmission, most published papers focus on two particular general information sources: speech and image data. Speech is taken in the general sense of human-produced sounds, including ordinary speech, as well as audio, music, and sound tracks. Images come in an enormous variety: photographs, digitally generated medical images, facsimile such as text and line drawings, multispectral images, monochrome and color, and sequences of images such as video and medical image movie loops. These two sources have many distinct structural and statistical features that can be used to advantage by compression algorithms. Most compression algorithms can be specifically tailored to either data type, and some algorithms are useful only for one or the other. Transform and subband techniques can be fine tuned to either data type, but there is little work on block truncation coding for speech or on code-excited linear-prediction methods for images. Because of the extreme importance of and interest in these two data types, each is given a chapter in this assessment.

G. RELIABLE COMMUNICATION OF COMPRESSED DATA

Fundamental results in information theory and coding theory obtained in the United States[37] and abroad during the last 45 years have provided a solid theoretical basis to the problem of reliable communications. Since the most important results of this theory are non-constructive in nature, a very large amount of research has been conducted to arrive at practical applications and to approach the theoretical limits.

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37 Starting with the work by C. E. Shannon, "A Mathematical Theory of Communication," *Bell System Tech. J.*, *27*, (Jul 1948), 379–343.

In general, the complexity of implementation increases very steeply while approaching those limits.

The most important quantity in establishing the amount of information that can be reliably transmitted over a channel or stored in a storage device is its *capacity*. To find the capacity of a channel, it is necessary to get a plausible statistical model of the channel—this may be relatively easy as in satellite or cable links, or more complicated, as in the case of a switched phone network. The *reliability* of the communication system is guaranteed by the fact that the probability of error can be made to approach zero for any transmission rate below the capacity of the channel, by using appropriate error-correcting codes.

If one is willing to accept some distortion or loss of quality in the reproduced messages, then results from the branch of coding theory called *rate distortion theory*[38] can be used to evaluate the maximum possible transmission rate that guarantees a certain required fidelity.

Coding theory also predicts that the problems of compressing the data generated by a source (source encoding) and of protecting the data from errors introduced by noisy channels (channel coding) can be treated separately without sacrificing anything in terms of the possible rate-distortion performance. This consideration does not preclude savings in *complexity* by treating source and channel coding as a joint problem.

Therefore, it is important to find efficient strategies to protect from channel errors data that have been compressed by a source encoder. The protection offered by error-correcting codes is particularly important in the case of compressed data, since single errors introduced by the channel may affect large portions of the message due to error propagation. Conversely, it is also important to devise compression methods that are less susceptible to error propagation.

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38 T. Berger, *Rate Distortion Theory*, Englewood Cliffs, NJ: Prentice-Hall Inc., 1971.

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CHAPTER III
LOSSLESS COMPRESSION

A. SUMMARY

Lossless data compression utilizes two major paradigms: *statistical* coding, which estimates input statistics and replaces frequent patterns with short codes, infrequent patterns with longer codes; and *dictionary* methods, which code patterns by citations to their earlier occurrences. The former method is exemplified by arithmetic and Huffman coding, the latter by Lempel-Ziv coding.

Researchers from the Commonwealth nations of Australia, Canada, and New Zealand are prominent in estimating input statistics, as well as in losslessly compressing large bodies of text, such as data bases, concordances, and natural-language corpora. Israeli researchers are also prominent in compression of large bodies of text. West Europeans, especially the French and Finns are prominent in treatment of linguistic data.

Theoretical aspects of lossless compression, including work on the classical problems of classification and estimation, are dominated by Israel, Russia, Finland, Japan, and Western Europe, in that order. Work on practical aspects of coding—the design of new algorithms and the modification of existing algorithms—is concentrated in Japan, Israel, Australia, Canada, and New Zealand (with little work in Russia or Western Europe).

Transmission-error-resistant coding is a research focus in Japan. Interesting work on the application of lossless compression to images and medical data (often in tandem with a lossy stage) is going on in Japan and Western Europe. The integration of error control is modest up to now, but certain to grow. Great Britain, India, China, South Korea, Taiwan, and Singapore are conspicuously inactive in this area, considering its potential importance to their commerce.

The enhancement and implementation of cryptographic systems by compressors are well known and likely to continue to be an incentive for further research. Notable work connecting compression and cryptography is found in Western Europe.

Arithmetic coding achieves the best compression for sources with low-order dependencies; Lempel-Ziv coding achieves the highest compressionfor higher-order sources, and is typically faster in both hardware and software. The quest for speed in lossless compression is driven by two major applications: user-transparent compression, as applied in computer operating systems, and real-time video compression. The most important speed improvements are expected to come from advances in device technology and architecture, especially systolic and parallel processing. The United States seems almost alone in practical aspects of this hardware, while European researchers are contributing to the theory.

B. INTRODUCTION

1. Background

Lossless data compression is also known as *noiseless compression*, *compaction*, and *entropy coding*. These terms refer to the translation of discrete (non-continuous) data into a more compact form, in such a way that the original can be reconstructed exactly. Exact reconstruction is an obvious necessity in compressing data such as computer programs and system specifications, where changes in individual digits can be catastrophic. Lossless coding is also essential for encrypted data or data already compressed by a lossy algorithm, because small changes in the compressed form can propagate into enormous differences during decryption or lossy reconstruction. Lossless compression often follows a lossy pass, removing some remaining redundancy, and thus easing the burden of the lossy algorithm. Recent work on vector quantization has addressed the problem of designing the lossy pass in conjunction with the anticipated entropy coding so as to optimize the overall compression achieved by the tandem combination.[1] Sometimes, two types of

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1 N. Farvardin and J. W. Modestino, "Optimum Quantizer Performance for a Class of Non-Gaussian Memoryless Sources," *IEEE Trans. Inform. Theory*, (1984), 485–497.

P. A. Chou, T. Lookabaugh, and R. M. Gray, "Entropy-Constrained Vector Quantization," *IEEE Trans. ASSP*, *37*(1989), 31–42.

R. M. Gray, A. H. Gray, Jr., G. Rebolledo, and J. E. Shore, "Rate-Distortion Speech Coding with a Minimum Discrimination Information Distortion Measure," *IEEE Trans. Inform. Theory*, *28*, 6(1981), 256–261.

J. Lin, J. A. Storer, and M. Cohn, "On the Complexity of Optimal Tree Pruning for Source Coding," *Proc. IEEE Data Compression Conf., Snowbird, Utah*, 1991, 63–72.

compression are applied in tandem to improve upon the performance of either alone.[2] Generally, compression *must* be lossless whenever data cannot be specified or limited in advance; for instance, it has been announced that forthcoming operating systems like MS-DOS 6.0 will include data compression utilities to double the virtual disk space as seen by the user; such compressors must be lossless.

Other applications currently require lossless compression, although this may relax in the future. For instance, users of medical imagery such as X-ray photographs, tomographs, and magnetic-resonance images have not commonly accepted lossy compression, even though much higher compression ratios are attained. There are studies in progress that may convince the cautious. Similarly sensitive applications include fingerprints[3] and X-ray scattering images.

The term "lossless" originally was used to mean that the probability of reconstruction error approaches zero as the data length grows large. However, the term is now used in the place of "strictly lossless" to mean that absolutely no change takes place, regardless of the amount of data processed. It is important to note that lossless compression by itself cannot cope with media or channel errors, which must be handled by error-correction/detection techniques. However, there are some recent attempts in the direction of error-resilient lossless compressors for which the effect of media/transmission errors can be shown to be bounded. It is also worth noting that no single lossless technique can compress all data. In order for an algorithm to assign short codes to some inputs, it must assign the remaining long codes to other inputs, so that *some* data must be expanded. The trick is to see that important, or commonly occurring, data get compressed. As a practical matter, most of the data produced by humans and their machines are compressible.

2. Theory Versus Practice

The study of lossless coding comprises two general areas, though they often mingle in research. One is the study of data *complexity*, an attempt to characterize what it is that makes data sources hard or easy to compress and how this might

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2 E. R. Fiala and D. H. Green, "Data Compression with Finite Windows," *Commun. ACM, 32,* 4(1988), 490–505.

3 T. Hopper and F. Preston, "Compression of Gray-Scale Fingerprint Images," *Proc. IEEE Data Compression Conf., Snowbird, Utah,* 1991, 309–318.

correspond to intuitive notions of complexity and randomness. Complexity measures may be statistical, as in Shannon's entropy measure,[4] or algorithmic,[5] or combinatorial.[6] Some of the most interesting work relates these seemingly different complexity measures, thus imparting a satisfying consistency to the study.

Beyond theoretical assessments of data complexity lie the practical problems of how actually to compress the data that are theoretically compressible. The criteria here are *compression efficiency*—how close to predicted compressibility is the actual performance—and *time or space requirements*—how fast can the algorithm be made to run, on how much digital logic, and with what amount of storage? Given historical increases in digital-logic speed, and decreases in component sizes and costs, absolute speed and cost are moving targets. To normalize this fluid situation, algorithmic running times are usually expressed in terms of the number of elementary operations to be executed (comparisons, shifts, adds); memory space and device counts are often neglected. Much of the research on lossless compression concerns the modification and enhancement of algorithms whose basic designs are well established. It is usually required that the processing time be *linear* in the input size; that is, that the number of elementary operations per input symbol be bounded by a constant. This permits the coder and decoder to perform "on-line" or "in real time," meaning that, in principle, they can process serial input without significant buffering or delay, and without any loss of data. The real-time requirement, in fact, may be inappropriate for some applications, where batch processing is applicable (think of archival storage). But almost all current research aspires to the linear time bound.

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4 C. E. Shannon, "A Mathematical Theory of Communication," *Bell System Tech. J.*, 27 (1948), 379–423, 623–656.

5 A. N. Kolmogorov, "On the Logical Foundation of Information Theory," *Prob. Inform. Transm.*, 5(1969), 3–7.

A. N. Kolmogorov, "Three Approaches to the Quantitative Definition of Information," *Prob. Inform. Transm.*, 1(1965), 1–7.

G. J. Chaitin, "A Theory of Program Size Formally Identical to Information Theory," *J. ACM*, 22:3(1975b), 329–340.

G. J. Chaitin, "Randomness and Mathematical Proof," *Scientific American*, (May 1975), 47–52.

G. J. Chaitin, *Algorithmic Information Theory*, New York: Cambridge U. Press, 1987.

6 A. Lempel and J. Ziv, "On the Complexity of Finite Sequences," *IEEE Trans. Inform. Theory*, 22, 1(1976), 75–81.

Some work even deals with *sub*linear processing time, using parallel computing on pre-loaded data.[7]

3. Basic Approaches

The scientific revolution in lossless compression began with Shannon's landmark 1948 paper.[8] Prior to this, there had been ad hoc attempts typified by the shipping codes and the Western Union codes in which a fixed selection of useful messages was assigned short replacements, so that a phrase or sentence ("Arriving tomorrow, arrange berth." "Happy Birthday, Sister dear!") could be transmitted by just a few taps of the teletype. For a short time after Shannon's breakthrough, general interest in designing codes for sources whose statistics were assumed to be known and fixed continued. Shannon-Fano coding[9] and Huffman's (static) code[10] were of this type. Interest then swung to the question of *universal* and *adaptive* codes. These two concepts are close in meaning, the first alluding to a family of codes designed to match a set or range of parameters that characterize a source, the second, to an algorithm that dynamically changes its parameters as it gains more experience with the source. The first well-known adaptive code was Gallager's modification of Huffman coding,[11] The study of lossless universal codes dates from Fano[12] and Ott[13] through the present time (Delogne and Macq, 1991; Han, 1990; Ryabko, 1988b; Tjalkens, 1988; Yokoo, 1988).[14] Some recent work is less general and is married to a specific application; for example, a battery of Huffman codes for

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7 T. Markas, J. Reif, and J. A. Storer, "On Parallel Implementations and Experimentations of Lossless Data Compression Algorithms," *Proc. IEEE Data Compression Conf., Snowbird, Utah,* 1992, 425.

S. De Agostino and J. A. Storer, "Parallel Algorithms for Optimal Compression Using Prefix Property Dictionaries," *Proc. IEEE Data Compression Conf., Snowbird, Utah,* 1992, 52–61.

8 Shannon, 1948, op. cit.

9 R. M. Fano, *Transmission of Information,* New York: Wiley, 1961.

10 D. A. Huffman, "A Method for the Construction of Minimum-Redundancy Codes," *Proc. IRE, 40* (1952), 1098-1101.

11 R. G. Gallager, "Variations on a Theme by Huffman," *IEEE Trans. Inform. Theory, 24,* 6(1978), 668–674.

12 R. M. Fano, PhD Thesis, Cambridge: MIT, 1949.

13 G. Ott, "Compact Encoding of Stationary Markov Sources," *IEEE Trans. Inform. Theory,* (1967), 82–86.

14 J. Rissanen, "Modeling by Shortest Data Description," *Automatica,* 14(1978), 465–471.

J. Rissanen, "A Universal Data Compression System," *IEEE Trans. Inform. Theory, 29,* 5(1983), 656–664.

deep-space telemetry,[15] or predictive pixel coding with prediction error assumed to be one of a parametric family of Laplacian distributions,[16] or a file compressor for personal computers that attempts several methods on each file and picks the one that seems best.[17] The adaptive lossless codes fall into two broad categories, *statistical-model* codes and *dictionary-based* codes. Well-known members of the first class include adaptive Huffman coding, Tunstall coding,[18] predictive coding,[19] and arithmetic coding,[20] although it is arguable that any statistical-model code is a form of prediction. Among these, arithmetic coding has an advantage that is not only theoretical but manifests itself in practice: the compressor need not output integral numbers of bits; it need not even output *any* bits on some frequent inputs. For this reason, arithmetic coding is the choice for most uses. The statistical-model codes differ from one another in how the model is formed and updated, as well as in how input characters are then encoded as a function of their predicted frequencies. Models typically are Markovian, of fixed or variable order (Horspool and Cormack, 1986). Updating strategies determine how stable or rapidly adaptive they are,[21] while coding strategies determine how closely the output length can approach the ideal entropic length and at what cost in arithmetic operations, which in turn are constrained by time and space requirements.

The dictionary-based codes relate remotely to the Western Union idea, insofar as they maintain a finite collection of strings (the dictionary) and they encode incoming data by pointing successively to matches in the dictionary. This class includes the

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15 R. F. Rice and J. R. Plaunt, "Adaptive Variable-Length Coding for Efficient Compression of Spacecraft Television Data," *IEEE Trans. Comm. Tech.*, *COM-19*, (1971), 889–897.

16 P. Howard and J. S. Vitter, "New Methods for Lossless Image Compression Using Arithmetic Coding," *Proc. IEEE Data Compression Conf., Snowbird, Utah*, 1991, 257–266; also in *J. Inform. Proc. Mgmt.*, (1992).

17 ARC Software, Systems Enhancement Associates, Inc., Wayne, NJ
MacRisken, US Patent 4,730,348

18 B. P. Tunstall, "Synthesis of Noiseless Compression Codes," PhD Thesis, Georgia Institute of Technology, 1968.

19 P. Elias, "Predictive Coding," *IRE Trans. Inform. Theory*, *IT-1*, 1(1955), 16–33.

20 N. Abramson, *Information Theory and Coding*, New York: McGraw-Hill, 1963.
J. Rissanen and G. Langdon, "Arithmetic Coding," *IBM J. Res. Develop.*, 23 (1979), 149–162.
I. H. Witten, R. M. Neal, and J. G. Cleary, "Arithmetic Coding for Data Compression," *Commun. ACM*, *30*, 6(1987), 520–540.

21 J. L. Mitchell and W. B. Pennebaker, "Optimal Hardware and Software Arithmetic Coding Procedure for the Q-coder," *IBM J. Res. Develop.*, 32 (1988), 727–736.

nomenclatures *citation*,[22] *textual substitution*,[23] *interval* and *recency-rank*,[24] *book-stack* (Ryabko, 1987a) and, most ubiquitously, the algorithms pioneered by Ziv and Lempel and known for short as *LZ*, *ZL*, and *LZW*, possibly suffixed with distinguishing letters or numerals.[25]

The differences among dictionary-based codes consist in how the dictionary is initialized, updated, and searched. Proper initialization can mitigate compression inefficiency during the early stages of adaptation. Just as with statistical models, updating can be tailored to track a rapidly changing source or, conversely, to smooth perturbations in a source known to be stable. Searching strategies relate to the data structures chosen to hold the dictionary, and affect the data rates attainable and the memory requirements.

The references cited in this chapter clearly show that the birth of rigorous lossless data compression and most of its early development took place in the United States, under the stimulus of academic workers such as Shannon,[26] Fano, Elias, and Huffman. The balance shifted somewhat in the mid-1970s, when Lempel and Ziv began the collaboration in Israel that led to the family of lossless codes bearing their names. (It is interesting, though, that both spent considerable parts of those years at research facilities in the United States, where early implementations were designed, tested, and patented.)[27] With that noteworthy exception, US universities and industrial laboratories have continued to dominate research and development in lossless algorithms until the contributions to arithmetic coding by Canadians in the mid-1980s.

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22 T. A. Welch, "High-Speed Data Compression and Decompression Apparatus and Method," US Patent 4,558,302 10 Dec 1985.

23 J. A. Storer, "Textual Substitution Techniques for Data Compression," *Combinatorial Algorithms on Words*, Eds. A. Apostolico and Z. Galil, Springer-Verlag, 1985, 111–129.

24 P. Elias, "Interval and Recency Rank Source Coding: Two On-Line Adaptive Variable Length Schemes," *IEEE Trans. Inform. Theory*, 33, 1(1987), 3-10.

25 J. Ziv and A. Lempel, "A Universal Algorithm for Sequential Data Compression," *IEEE Trans. Inform. Theory*, 23, 3(1977), 337–343.

J. Ziv and A. Lempel, "Compression of Individual Sequences Via Variable-Rate Coding," *IEEE Trans. Inform. Theory*, 24, 5(1978), 530–536.

T. A. Welch, "A Technique for High-Performance Data Compression," *IEEE Computer*, 17, 6(1984), 8–19.

26 Shannon's career was divided between AT&T Bell Laboratories and MIT.

27 W. L. Eastman, A. Lempel, J. Ziv, and M. Cohn, "Apparatus and Method for Compressing Data Signals and Restoring the Compressed Data Signals," US Patent 4,464,650, 7 Aug 1984.

C. NON-US ACTIVITY

This section surveys all aspects of non-US research and development in lossless data compression. The reader is cautioned that it is not often meaningful to partition activities into discrete, disjoint categories, because research often overlaps two or more areas. Theory papers might discuss concrete codes that are either illustrative of principles or are capable of being proven optimal in some sense by the theory. Clever modifications of known algorithms may be interesting in their own right but also motivated by and directed toward specific applications. Keep in mind that in almost every case where a national strength is cited, the words "besides the United States" should be understood.

1. Summary of Non-US Activity

a. Theory

Israel, Russia, Finland, and Japan dominate recent work in universal coding and modeling. Russian researchers have long been strong in probability and ergodic theory, and Kolmogorov's successors in the former Soviet Union have maintained interest and continuity, but practical design is not emphasized. Finland is disproportionately represented. For over two decades, Israelis, often with US co-authors, have been a powerful influence in ergodic theory and its practical consequences. Japanese researchers are more followers than leaders in fundamentals, but increasingly apply theory to *knowledge systems* and artificial-intelligence (AI) problems.

Researchers in Western Europe show interest in algebraic sources, like automata and semigroups; they are joined by Israelis in the study of combinatorial properties of codes.

b. Code Design

The current emphasis is on source estimation, codes that obey restrictions, and computational enhancement. Israel and Japan are joined by Australia, Canada, and New Zealand in this work.

c. Cryptology

Compression enhances security, and compressors can analyze the security of key streams. Western Europe dominates this literature. Pacific Rim countries and China encrypt characters.

d. Algorithmic Refinements

Japan, Australia, Canada, and New Zealand show enormous activity in improving the performance of standard compressors. Neural nets have been considered, and other AI tools such as fuzzy logic and genetic algorithms should follow. A large number of research papers concern modification in the service of application, taking advantage of "side information" to craft a compressor that is superior in some way to the general issue. Space-filling curves and hierarchical decompositions are routinely rediscovered and applied. All countries except those of the Pacific Rim, China, and Russia are active here.

e. Applications

Another common interest is the application of known compressors to novel or important applications. Japan emphasizes image and numerical data; Australia, Canada, Israel, and Finland concentrate on word/phrase/document storage and retrieval. Worldwide, text and character compression is the most common application of lossless coding. Lossless compression of images is the next most common application. (Lossless coding alone is inappropriate for video.) Compression of medical data and other numerical data is receiving increased attention in Japan; all the researchers use lossless compressors to "mop up" after lossy compression.

2. Theory

A *source* is an idealized version of a process that generates data. Much of lossless compression theory concerns the characterization of sources, usually probabilistically, but sometimes combinatorially. Such characterizations have several goals:

- They can lead to bounds on the best compression that can be expected of *any* general-purpose algorithm operating on such data.

- They can suggest *concrete* algorithms that will, in the limit of long inputs, attain the compression bound when applied to an appropriate source. For example, in a joint paper, Lempel and Ziv (1976) described a complexity measure that lies at the heart of the compressors called LZ-I (Ziv and Lempel, 1977), LZ-II (Ziv and Lempel, 1978), LZW,[28] and their myriad offspring.

- They permit estimates of how rapidly an associated algorithm will converge to predicted performance, or in other words, *adapt* as a function of input length.

- They can suggest empirical measurements that most naturally estimate the ideal source parameters for real data sources.

Outside the United States, Israel and Russia have dominated the elaboration of Shannon's theory of lossless compression. Finland, Holland, Japan, and France form a second tier. In Israel, the work of Ziv, his colleagues, and his students is outstanding and has most influenced the work (and citations) of all others (Merhav, 1991; Merhav et al., 1989; Plotnik et al., 1992; Weinberger et al., 1991; Wyner and Ziv, 1989, 1991; Ziv and Merhav, 1990). Russia and Eastern Europe show continuing interest in universal coding and ergodic theory, in which the work of Ryabko and Shtarkov frequently appears (Gerencser, 1991; Ryabko, 1987a–b, 1988a–b, 1989, 1990, 1991; Sauvagerd, 1988; Shtarkov, 1991a-b; Shtarkov and Tjalkens, 1990; Trofimov, 1987); universal codes are also investigated by researchers from The Netherlands (Tjalkens, 1988; Willems, 1987; Tjalkens et al., 1987), France (Hansel, 1989), and Japan (Kobayashi and Iwamoto, 1990). The latter work generally considers independent, identically distributed sources (which produce one symbol at a time, with probability unaffected by context) or Markov source models, which are simpler than the sources studied by Israeli and US researchers.

28 Welch, 1984, op. cit.

Related work from Israel, Western Europe, and Japan on the theoretical performance of abstract computational models (such as finite-state transducers and automata with bi-directional heads—Sheinwald et al., 1991) is more theoretical than practical, but relates the computation model to compression efficiency and rates of convergence. Israeli and West European researchers frequently have viewed compressors from the standpoint of grammatical parsers, factor automata, and combinatorially constrained systems (Shannon's "discrete noiseless channel"—Hansel et al., 1992; Jacquet and Szpankowski, 1989, 1991; Zipstein, 1991). Much of this interest derives from the practical problems of computational linguistics, discussed later, and the coding terminology has infiltrated that area. Somewhat closer to practical concerns are performance differences between codes whose input words have fixed length while output lengths vary versus the opposite situation (Ziv, 1990), which has implications in the amount of input and output buffering needed to avoid data loss in on-line compression (Krichevskiy and Trofimov, 1981). It has long been known that optimal compression can be achieved at the price of coding delay—the amount of input that must be buffered before the output can be computed. Some recent Israeli work refines the relationship (Weinberger et al., 1992; Merhav and Neuhoff, 1992).

Since its introduction in 1952, the Huffman code not only remains popular with users, but depicted as a weighted finite tree, has attracted combinatorial analysis. In recent years, the late Renato Capocelli and his Italian colleagues have been prominent in studying bounds and distributions of Huffman code word lengths, or equivalently, of path lengths in weighted trees. These studies have mainly mathematical interest, but in instances where compression optimality must be sacrificed to meet register-length constraints, these studies enable a designer to relate the sacrifice to the constraint (Capocelli and De Santis, 1988, 1990, 1991a-b, 1992a-b; Capocelli et al., 1986; De Santis and Persiano, 1991; Golic and Obradovic, 1987; Manstetten, 1991).

There seems to be no equivalent interest in the size and shape of the codes of Tunstall,[29] whose construction is dual to Huffman's. (Tunstall codes are variable-length to fixed-length, while Huffman codes are fixed-length to variable-length.) Even arithmetic coding, which has largely replaced Huffman coding among statisti-

[29] Tunstall, 1968, op. cit.

cal codes of choice, boasts not nearly so many analyses (Raita and Teuhola, 1989; Nakatsu, 1991).

3. Estimation

Information theory remains a powerful tool in classification, learning, and estimation theory, essential in pattern recognition and hypothesis testing. This section reviews the use of compression algorithms for estimating parameters of unknown sources. Proposals for using these ideas, but with explicit applications in mind, are discussed in Section III.C.5, "Applications of Lossless Codes."

Many non-US authors have remarked that measures such as entropy or mutual information can be estimated directly by compression, rather than computed from sampling in the classical way (Barlow et al., 1989; Burrows and Sulston, 1991; Feistauerova, 1988; Grassberger, 1989; Hansel, 1989). However, there is an even more general and powerful aspect to the theory of lossless compression, which is its application to classification and hypothesis testing. Intuitively, a compressor that has adapted to a source is a matched filter for that source. Suppose that an adaptive compression algorithm is "trained" by being exposed to enough source data to approach optimum compression, and then the adaptation is stopped. If the frozen compressor is now exposed to data from a source identical, or nearly so, to the original, it will continue to compress at about the same rate. If, on the other hand, the new input data are statistically or combinatorially distinct from the training data, the data will be poorly compressed or even expanded. In this way the compressor can be used to discriminate among sources or, in other words, to classify sources. This idea (Ziv, 1988; Weinberger et al., 1991) has been exploited by researchers in Japan, Israel, and elsewhere to study automatic inference and machine learning, thus extending and deepening the application of ideas from information theory in artificial intelligence. The term "minimum description length" (MDL),[30] the smallest number of bits that define a stochastic state process, has achieved wide influence in this field (Feder, 1991; Gabor et al., 1991; Gammerman and Bellotti, 1992; Matsushima et al., 1990; Moharir, 1991; Rissanen, 1983; Takimoto and Maruoka, 1991; Yokoo, 1988).

30 Rissanen, 1978, 1983, op. cit. The term has been traced back to R. A. Fisher.

4. Code Design

This section reviews non-US efforts in the enhancement and implementation of general-purpose coding techniques, not tied to any special application nor to any embodiment in software or hardware. These activities include the practical characterization of sources, improvement of computational efficiencies in established coding techniques, and modification of established techniques to accommodate restrictions. Despite the beliefs of occasional researchers, no truly new compressor designs have emerged in the past decade.

The practical modeling of real-world sources is essential for the performance of statistical compressors, as exemplified by arithmetic coding, Huffman coding, and Tunstall coding. The inventors of these coding schemes took for granted that source statistics were given, and showed how best to use them. Statistics *are* available *a priori* in certain well-studied cases: the zero-th order, and in some cases higher-order, statistics for popular languages such as English, Russian, French, and so on. However, in many other cases, such statistics are not available or are not accurate for the subset of language actually encountered in an application, so the encoder must collect them from the sample of data at hand. The decoder, which also needs to know the statistics used by the encoder, must either be given them explicitly, at a cost in compression ratio, or must collect them for itself. Practical concerns in collecting statistics include controlling the storage space necessary, the choice of data structures that permit rapid enough searching for real-time updating and encoding, and the order or complexity of the model being fitted to the data. Higher-order statistics give better compression, but are more difficult to estimate correctly because of their correspondingly small sample sizes. "Novel events," symbols not yet sampled by the compressor, raise another problem. All these concerns are, in fact, classical, but a disproportionate amount of recent work is due to a few authors (Bell, Cleary, Horspool, and Witten, from Canada, New Zealand, and Australia) concerned mainly with modeling for arithmetic coding (Bell et al., 1989, 1990; Witten and Bell, 1991; Witten and Cleary, 1983; Horspool and Cormack, 1986, 1987, 1992; Han, 1990; Cormack and Horspool, 1987). Japanese researchers have also addressed these problems (Witten and Bell, 1991; Vermeersch et al., 1991). Occasionally, inference devices such as Hopfield Nets are considered for estimating conditional probabilities (Hentschel and Barlow, 1991).

Sometimes an extraneous restriction, dictated by channel or memory constraints, is imposed on a standard compression algorithm. A common instance concerns alphabetical ordering, in which the code words assigned to events must display the same alphabetical ordering of the events they encode in order, for instance, to permit rapid searching of the compressed form. Non-US researchers consider this topic as well as the design of codes that are decodable either forwards or backwards for use in tape drives and deque (double-ended queue) data structures (Fraenkel and Klein, 1990; Nakatsu, 1991; Sheinwald, 1992). Of much wider interest are methods of enhancing the performance of standard algorithms by introducing new ways of computing necessary functions or new data structures for maintaining and searching the compressor's model or dictionary. This work can be collected naturally by the type of compressor it is intended to improve.

a. Arithmetic Codes

This area (which has already enjoyed a history of notable US contributions from Pasco,[31] Langdon,[32] Rissanen,[33] Vitter, and Howard[34]) has been explored by researchers in Japan, Israel, and Finland. Some work is concerned with accelerating the arithmetic computations necessary to generate code output from input probabilities. If approximate computations are employed, such as replacing explicit multiplications with register shifts, some small degradation in compression performance is a necessary consequence, but speed can be markedly increased (Chevion et al., 1991). In other cases, improvements are claimed by refining the storage and access of probability contexts via tree structures (Park et al., 1989; Ueda et al., 1991), possibly combined with Lempel-Ziv encoding of arithmetic-coder output (Ghanbari, 1991), or vice-versa (Park et al., 1988). The use of hashing (Teuhola and Raita, 1991; Raita and Teuhola, 1989) and of fixed-output lookup tables (Koch and Sommer, 1992) is also suggested.

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31 R. Pasco, "Source Coding Algorithms for Fast Data Compression," PhD Dissertation, Stanford University, 1976.

32 G. G. Langdon, "An Introduction to Arithmetic Coding," *IBM J. Res. Develop., 28*, 2(1984), 135–149.

33 J. Rissanen and G. Langdon, "Arithmetic Coding," *IBM J. Res. Develop.*, 23(1979), 149-162.
J. Rissanen, "Generalized Kraft Inequality and Arithmetic Coding," *IBM J. Res. Develop.*, 20 (1976), 198–203.

34 P. G. Howard and J. S. Vitter, "Practical Implementations of Arithmetic Coding," *Image and Text Compression*, Norwell, Mass: Kluwer Academic Press, 1992.

b. Huffman-Tunstall Codes

Because of their familiarity, simplicity, and implementation speed, Huffman codes hold their interest despite increasing use of arithmetic codes with better compression performance. Around the world, hashing, exotic tree structures, new indexing tricks, and dedicated hardware are devoted to the enhancement of our oldest practical compression algorithm (Llewellyn, 1987; Fitingof and Waksman, 1988; Basu, 1991; Pataricza, 1988; Sato et al., 1987; Lu and Chen, 1990; Ho and Law, 1991; Yokoo, 1991; Brent, 1987). The serial speed still does not match that of dictionary coders. One noteworthy development is a parallel algorithm that, given a set of symbol probabilities, produces in sublinear time a code equivalent to a Huffman code. This is of purely theoretical interest, unless it can be extended to adaptive modification and used to speed the coding process itself (Teng, 1987).

c. Lempel-Ziv Codes

Enhancements to the Lempel-Ziv algorithm aspire to speed the compression by fast searching techniques, improve the compression ratio by conserving output bits, and permit the tracking of non-stationary sources while living within fixed memory constraints. There is much duplication in this research. The two distinct embodiments for LZ compression require the matching of character strings. Among serial algorithms, there have been no remarkable improvements in the speed of LZ-II schemes since the "LZW" (UNIX Compress) version of 1984 introduced hashing.[35] Gzip, Implode, Stacker, and Microsoft DoubleSpace,[36] all based on LZ-I with hash searching, attain better compression at about one-quarter the speed of LZW, but with equivalent decompression speeds. A somewhat simpler British invention is reportedly even faster than the three aforementioned, but with inferior compression (Waterworth, 1987).

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35 Welch, 1984, 1985, op. cit.

36 Gzip, distributed as *freeware* by Free Software Foundation, 675 Massachusetts Ave., Cambridge, MA 02139.

PKZip et al., distributed as *shareware* by PKware, 9025 North Deerwood Drive, Brown Deer, WI 53223.

Stacker, marketed by Stac Electronics, 5993 Avenida Encinas, Carlsbad, CA 92008

DoubleSpace, marketed as part of DOS 6.0 by Microsoft Corporation

Earlier US work described a linear-time tree search for LZ-I that, in principle, permits linear-time searching.[37] This motivation has further stimulated the study of exact and approximate string matching, which was already a classic area of computer science. Researchers from Japan, France, and Finland have considered the behavior of tree-structured dictionaries implementing the Lempel-Ziv algorithm (Jacquet and Szpankowski, 1991; Kawabata and Yamamoto, 1991; Murashima et al., 1990; Yokoo, 1992). Welch's variation not only gained speed, but saved output bits by deferring the encoding of raw characters.[38] Further attempts to save output bits often entail using variable- rather than fixed-length pointers to dictionary strings while the memory is filling for savings of a few percent on shorter files, and almost none on longer files (Williams, 1991). The technique is employed in the first LZ patent[39] and in subsequent designs such as Stacker, PKzip, Gzip, and DoubleSpace.[40] A critical step shared by all Lempel-Ziv compressors entails finding the longest occurrence in the recent input that matches the current input. Given current advances in parallel algorithms and the growing accessiblility of parallel processors, research on parallel searching and matching is of the highest importance. Parallel execution will make compression invisible to the users and to application programs. In that sense, compression will become "free," as it is not now. Parallel algorithms project *sublinear* execution times for dictionary-based compressors. Naturally, the input must be preloaded or loaded in parallel in order to take advantage of sublinear-time execution. Systolic pipeline architectures can process 20 megabytes per second (Mbps) with VME interface and promise 100 Mbps with HiPPI logic.[41] No non-US claims were found when conducting this assessment. However, theoretical work in this important area does seem more foreign-based than domestic,

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37 Fiala and Green, 1988, op. cit.

Ziv and Lempel 1977, 1978, op. cit.

38 M. Cohn, "Ziv-Lempel Compressors with Deferred Innovation," *Image and Text Compression*, Ed. J. A. Storer, Boston: Kluwer Academic Publishers, 1992, 145–158.

39 Eastman et al., 1984, op. cit.

40 Gzip, PKZip, Stacker, DoubleSpace, op. cit.

41 J. H. Reif and J. A. Storer, "A Parallel Architecture for High-Speed Data Compression," *J. Par. Distrib. Comput/, 13*, (1991), 222–227.

J. A. Storer, "Massively Parallel Systolic Algorithms for Real-Time Dictionary-Based Text Compression," *Image and Data Compression*, Ed. J. A. Storer, Boston: Kluwer Academic Publishers, 1992.

R. Zito-Wolf, "Systolic Architecture for Sliding Window Data Compression," *Proc. IEEE VLSI Signal Processing Conf., San Diego*, 1990, 339–351.

although several workers, Apostolico, Naor, and Teng, work at US institutions at least part of the time (Apostolico et al., 1987; Naor, 1991; Crochemore and Rytter, 1991; Teng, 1987).

Rapid adaptation, within bounded memory, to changing input statistics is part of the general source-modeling problem, raising the question of how past inputs should be discounted or how dictionary entries should be discarded to make room for newcomers. For the ZL family of compressors, strategies such as discarding the least-recently used (LRU) strings or the least-frequently used (LFU) strings are well known and widely used.[42] Moreover, high-performance software based on these ideas is available; the latest, called Gzip, involved Jean-loup Gailly, a Frenchman who also contributed to PKzip (Lagana et al., 1989; Jiang and Jones, 1991; Yokoo, 1992; Horspool, 1991; Murashima et al., 1990; Willems, 1987; Tischer, 1987).[43]

d. Combined Codes

Lossless compression has long been recognized as a possible adjunct, or follow-up pass, to lossy compression. Recent work has shown an interest in combining two lossless algorithms to gain better performance. There are two rationales. First, an adaptive algorithm can be working far below par during its early stages, so a recoding can remedy some inefficiency. Second, the combination of two (possibly non-optimal) lossless codes might yield a code with performance unattainable by any single code with the same parameters. Freeman (1991) has analyzed combinations of codes with better performance than any single Huffman or Tunstall code for a given complexity. Okumura (1988) distributed C-language programs for combinations of LZ-I with true Huffman and arithmetic coding (in contrast to Packer, PKzip, and Gzip, which use fixed, not necessarily optimal variable-length codes for output values). This software (especially the arithmetic tandem) achieved excellent compression for its time, although it was not nearly as fast as UNIX Compress. PKzip "Implode" and Gzip now outperform it in speed and compression. Other combinations have been proposed (Honary et al., 1990; Jiang and Jones, 1991; Park et al., 1988, 1989), but are not readily available for trial.

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42 J. A. Storer, *Data Compression: Methods and Theory*, Rockville, MD: Computer Science Press, 1988.

43 Gzip and PKzip, op. cit.

5. Applications of Lossless Codes

This subsection assesses non-US research and development activities whose specific goal is the application of lossless compression to a variety of practical problems rather than the invention or improvements of algorithms *per se.* Work that improved (or whose object was to improve) known algorithms with no particular application in mind have been discussed under the type of compressor or the type of improvement. The work discussed in this section simply gives a view of the many uses to which lossless compressors are being put.

a. Document Retrieval, Lexicon, Concordances

Compression is a natural requisite for large bodies of text, and *lossless* compression is almost always essential. Even though a human reader may tolerate misspellings, especially during rapid scanning, changes in text would confound sorting and searching functions that are commonly applied, and changes contradict the very meaning of the term "archival."

Researchers in Finland have considered text compression via citation (Katajainen and Raita, 1987) and prediction, including the problem of efficiently storing and accessing contexts consisting of preceding characters. This is closely related to the problem of modeling discussed earlier (Jakobsson, 1985; Raita, 1987; Raita and Teuhola, 1989; Teuhola and Raita, 1991). Sparse-bit or character vectors are often constructed in information retrieval work to indicate the categories to which an item relates. Compressing such vectors is a classic problem in source coding, and has been addressed recently in Israel (Jakobsson, 1982; Fraenkel and Mor, 1983; Fraenkel and Klein, 1985, 1986). Fraenkel has done much of his recent work in collaboration with US workers (Fraenkel et al., 1983), notably, Bookstein's group at the University of Chicago.

Workers in Canada, Australia, and New Zealand have studied the compression of data bases and concordances, addressing problems of extracting at least partial information from the compressed form, in order to avoid decompressing for every access (Cormack, 1985; Witten et al., 1991a-b). Other recent work from Canada (Moffat, 1989; Promhouse and Bennett, 1991) picks up an on older British investigation (Clare et al., 1972) by proposing to deal with text as words or even phrases, rather than isolated simple characters, and by invoking semantic and syntactic information. This approach is potentially powerful and is gaining credibility with the advent of cheaper and faster memory (Moffat et al., 1993; Cameron, 1988). It dovetails nicely with the efforts of computational linguists. In Italy, Lagana and colleagues have reported on the compression of dictionary entries and keys (Lagana et al., 1989, 1991). Other researchers have considered not only text (Jeuring, 1992; Aalbersberg, 1991; Basu, 1991) and games (Althofer, 1991), but also trees (Katajainen and Makinen, 1990), which can be used to represent structured data such as programs or data bases.

b. Linguistics

Closely related to document and concordance work is the application of information theory, compression, and parsing rules to linguistics. This is not surprising, given that automata and formal languages had their origins in natural-language processing. Grammatical prediction was proposed by Promhouse and Bennett (1991); Crochemore, in France, was cited above for parallel string processing; he has also been prolific in applications of string algorithms to linguistics (Cameron, 1988; Crochemore, 1989a-b; Gross, 1989; Hansel, 1989; Simon, 1989).

c. Image Coding

Digitized images are usually compressed with lossy algorithms capable of impressive compression ratios, but there are applications in which perfect reconstruction is required, for which lossless compression must be used. The medical and law-enforcement professions have been slow to accept lossy compression. In some applications, like electron-beam lithography, and in some methodologies, like contour coding, images have been precoded in such a way that small changes in reconstruction can have large effects.

Theoretical work on image coding is rare: students of Lempel and Ziv have elaborated the proof (Lempel and Ziv, 1985) that LZ compression is asymptotically optimal in higher dimensionalities greater than two (Sheinwald et al., 1990). The space-filling Hilbert curve used in that proof can also be employed in practice, as can a related Peano curve for non-square images. Japanese researchers have found that these have small advantages over the conventional raster scan, and, not surprisingly, that they disrupt the performance of the Q-coder,[44] whose context gathering is predicated on raster scanning (Skarbek et al., 1989; Agui et al., 1991).

Scientists at Toshiba have discussed the compression of a data file controlling the width of a scanning electron-beam, of obvious utility in integrated-circuit fabrication (Abe et al., 1991). Rather than a fixed scan rule with run-length coding (Chen and Weng, 1991), several papers advocate contour coding, finding small advantages (Liu and Prasad, 1991; Kim and Park, 1989). Progressive coding, as stipulated by the JBIG standard[45] for binary images, can be extended to gray-scale images by treating each bit-plane separately (Mashimo et al., 1988), by a pyramid of Gabor functions (a transform code—Ebrahimi and Kunt, 1991), or by quadtrees (Wang and Shi, 1991). Swedish researchers have studied successively more complex and effective lossless encodings of primary color planes, using predictive and arithmetic codings (Einarsson and Roth, 1987).

In contrast to scanning, images can be covered with blocks (by analogy with lossy vector quantization); rectangles as well as squares will serve (Stern, 1991). A Dutch university group reported on the theoretical computational complexity of covering two-dimensional arrays with objects selected from various collections of covering patterns (Bodlaender et al., 1991). Scientists from Philips in The Netherlands and the United States have jointly described a chip set for high-speed lossless image compression, using transforms as well as the LZ algorithm (Shah et al., 1991); and Belgian researchers have proposed a variable-length code to follow Hadamard-transform coding (Delogne and Macq, 1991).

Much of the non-US literature on image compression concerns the use of standard lossless algorithms to supplement lossy compression, and of this literature, a large proportion is applied to medical imagery—X-ray, computer-aided tomography (CAT), and magnetic resonance imaging (MRI—Arazaki et al., 1991; Bozzoli et al., 1989, 1990; Chang, 1990; Delogne and Macq, 1991; Einarsson, 1991; Finamore, 1987; Furlan et al., 1991; Gudmundsson and Randen, 1991; Ho and Law, 1991; Lavarenne

44 J. L. Mitchell and W. B. Pennebaker, "Optimal Hardware and Software Arithmetic Coding Procedures for the Q-coder," *IBM J. Res. Develop.*, 32(1988),727-736.

45 H. Hampel, R. B. Arps, C. Chamzas, D. Dellert, D. L. Duttweiler, T. Endoh, W. Equitz, F. Ono, R. Pasco, I. Sebestyen, C. J. Starkey, S. J. Urban, Y. Yamazaki, and T. Yoshida, "Technical Features of the JBIG Standard for Progressive Bi-Level Image Compression," *Signal Processing: Image Communication*, Elsevier Science Publisher, BV, 4(1992), 103–111.

and Sorel, 1989; Lutkenhoner, 1989; MacDonald et al., 1991; Saito and Iseda, 1987; Wang et al., 1991).

Finally, papers from China, Taiwan, and South Korea describe the lossless compression of characters, presumably to be used in document transmission and in "font files" for printers (Chang and Tsai, 1991; Chung and Wu, 1991; Han, 1990; Tang, 1990).

d. Secure Communications

The heading "Secure Communications" is intended to encompass the use of lossless compression in the areas of (1) cryptology, making data secure from eavesdropping and alteration; and (2) error resistance, making data secure from channel noise and storage errors.

Lossless compression has been related to cryptography ever since Shannon showed how perfect lossless compression could lead to perfect secrecy.[46] Perfect compression is not attainable, only approachable, but even practical compression enhances any secrecy system. British and Swiss researchers compare straightforward compression (Boyd, 1991) with the classical technique of "homophonic substitution," whereby frequent symbols are variously represented in order to smooth the frequency distribution. Proceedings of the annual conference "Eurocrypt," published by Springer-Verlag, are a perennial source for this material (Gunther, 1988; Jendahl et al., 1990; Maurer, 1991). Compression can afford character concealment (Chang and Tsai, 1991), can conceal file-access privileges (Jan et al., 1990), and can detect tampering (Koch and Sommer, 1991).

A more significant question is how to key a compressor so that its output can be correctly decompressed only by an intended receiver. An Australian proposal uses the source-model of an arithmetic coder as the encryption key (Bergen and Hogan, 1992), and a British invention suggests keying a hashed dictionary coder (Waterworth, 1987). A more popular idea makes use of the observation that the output of a lossless compressor approximates a random sequence, and that this property can be

46 C. Shannon, "Communication Theory of Secrecy Systems," *Bell Systems Tech. J., 28*, (Oct 1949), 656–715.

used to implement running-key ciphers (Jansen and Boekee, 1990; Yakovlev, 1991), or an auto-key cipher (Chung and Wu, 1990). All of these papers propose the Lempel-Ziv algorithm for its inherent speed. Analogously with the aforementioned estimation of the redundancy of sources, linguistic and otherwise, it is a natural step to propose lossless compression to confirm the security of a running key (Mund, 1991) and of speech-scrambling codebooks (Feistauerova, 1988), but cryptographically these are old ideas.

Noise presents an intrinsic problem for compressed data. This can be appreciated intuitively by the observation that for minimum redundancy, every code sequence must correspond to a unique source sequence, so that any change in the code sequence may decode into a totally different source sequence. Since redundancy is the prerequisite for error control, a minimally redundant message is maximally vulnerable to error. It is therefore something of a paradox to require both compression and error resistance, but the real world demands just that. The compromises that are usually made stipulate only that errors will not propagate indefinitely—that their effects be bounded in scope. The simplest way to bound errors is to compress by blocks, so that no block is corrupted by errors in other blocks. However, compressors, especially adaptive coders, become efficient only for very large blocks. The non-US literature in this area discusses mostly straightforward solutions to the problems of resynchronization (Higgie and Williamson, 1990; Mirkovic and Stojanovic, 1987; Rahman and Misbahuddin, 1989), error detection, and correction (Agnew, 1990; Honary et al., 1990; Nakai and Kasahara, 1989; Yamazato et al., 1991; Zhang et al., 1991). There is no mention in the non-US literature of error *resilient* coding, proposed in the United States by Reif and Storer,[47] whereby block length does not need to be limited, but the probability of error-propagation can be bounded exponentially in the propagation distance. The practical meaning is that extensive errors can be made disproportionately unlikely.

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47 J. A. Storer, "Adaptive Lossless Data Compression Over a Noisy Channel," *Sequences II, Methods in Communication, Security, and Computer Science,* New York: Springer-Verlag, 1993.

J. H. Reif and J. A. Storer, "Error Resilient Optimal Data Compression," manuscript, 1992.

e. Numerical Data

It is clear from previous applications that the lossless compression of texts, including natural language and computer programs, is receiving much attention. However, only recently have researchers addressed the problem of compressing *numerical* data. Such data arise obviously in remote monitoring and in tables of function values; the data can also describe byte-maps of images and other higher-dimensional arrays. Numerical data tend to be organized and correlated in a fashion different from text (but not so different from a dictionary or directory) (Lagana et al., 1991). Most recent work on numerical data is Japanese (Miyakawa et al., 1991; Yakovlev, 1991; Yamomoto and Ochi, 1991; Yokoo, 1990).

D. FINAL THOUGHTS

It has been difficult not to begin every paragraph or so with the phrase "Besides work in the United States ....," and the reader should keep in mind that, in almost all categories, the list of citations would have been doubled or more by inclusion of US work. It has also been difficult to determine whether work performed by non-US researchers at US institutions reflects technology that stems from their native lands or is destined to return there with them. The reverse puzzle, US researchers working abroad, arises less frequently.

General observations regarding the non-US research assessed in this chapter are as follows. There is little significant work to report from the Pacific Rim countries (China, Taiwan, South Korea, and Singapore), the African continent, or the United Kingdom, India, and Pakistan, as compared to their sister Commonwealth nations, Australia, Canada, and New Zealand, which are conspicuously active. This is somewhat surprising, given British contributions to related fields like cryptology. The Russians do analysis almost to the exclusion of synthesis, in contrast to their contributions in error control coding. Considering populations, the Finns and Israelis are represented out of all proportion.

1. Topics by Region

Countries or groups were mentioned in the earlier discussion by topics. This section provides a list of countries/groups with their apparent interests and specialties.

- Australia, Canada, and New Zealand: modeling of statistics; arithmetic coding; and document systems.

- Belgium, France, Germany, The Netherlands, and Switzerland: estimation; automata, formal languages; linguistics; cryptology; and surveys.

- China, South Korea, Singapore, and Taiwan: characters; and simple cryptology.

- England, Ireland, India, Pakistan, and the African nations: surveys, LZ extensions; and B/W images.

- Finland, Russia, and Hungary: entropy/ergodic theory; estimation; and linguistics.

- Israel: entropy/ergodic theory; estimation, classification; algorithm enhancement; document systems.

- Italy: combinatorics.

- Japan: inference, learning; image data, scans; code design; algorithm enhancement; and numerical data.

- Sweden: images.

2. Omissions

At first glance, it is surprising how few investigations involving fabrication of VLSI chips or chip sets for lossless compression were found in this assessment of non-US publications. One might conjecture that most countries are largely dependent upon a few others for device technology, and thus have no incentive to explore chip organization and fabrication. It might also be that those laboratories, private or national, that *do* design chips are reluctant to publish their work. A third inference is that many lossless designs are embedded as components in larger, lossy systems and will be encountered in the corresponding surveys.

There are also surprisingly few discussions of the integration of lossless compression with encryption.[48] This also is an area where publication may be inhibited for reasons of national security or proprietary ownership. On the other hand, it follows from Shannon's fundamental work[49] that, in principle, the functions of compression and encryption can be executed in tandem with no loss in performance, so the incentive to integrate them may not exist, except that practical considerations suggest that there should be economies of hardware and of speed, not to mention the elimination of an extra interface.

As mentioned above, a more general theory of probabilistic error recovery seems appropriate for a field where nonadaptive prefix- or run-length codes are losing ground rapidly to adaptive schemes that thrive on long input lengths. There were no thorough analyses of the tradeoffs involved between compression, with its tendency to engender error propagation, and error-control, with its necessity to add redundancy.

E. PROJECTIONS FOR THE FUTURE

This assessment has not uncovered non-US research thrusts substantially different from US activities. Dominance of the field by US journals and conferences makes unnoticed advances by non-US researchers unlikely. It is not surprising that countries like the United States and Japan, with enormous installed computing bases and device-manufacturing capabilities, will be most active in implementation, while researchers in less well endowed countries (like Russia, or even the United Kingdom) will be drawn to theory and specific applications. The present flood of computer science and engineering graduate students to the United States from

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48 PKzip and Gzip, op. cit., include encryption, apparently using shift-register sequence running keys.

49 Shannon, 1949, op. cit.

China, India, Pakistan, and the Pacific Rim, make it likely that research will eventually prosper in those places (if the students actually return home).

In the near future, it is very likely that all aspects of compression will begin to take advantage of the increasing availability of massively parallel computation. Parallel engines are evolving and becoming more widely available in Japan and Western Europe as well as in the United States. Some archival compression tasks can be subdivided readily, as can array computations (such as those used extensively in image processing). The expected increases in speed will facilitate real-time video compression. The use of parallel searching and matching can certainly accelerate lossless compressors of all types. A number of clever compromises have tweaked more speed out of arithmetic coders without seriously diminishing compression, but, among L/Z compressors, aside from Welch's decade-old version, there has been incremental but not dramatic acceleration. Something like probabilistic hashing or associative retrieval might yield still more speed with negligible performance loss, but improvements in device speeds will likely be dominant.

A national or international standard for lossless text compression would stimulate the market for, and hence the development of, hardware realizations. Such devices could then become standard in disk drives, data-transfer buses and modems, permitting the universal exchange of data in less time, bandwidth, and space. For instance, with a standard interface, software could be distributed in read-only memory chips to be expanded and executed by the host machine. Software could be "rented" per use, just as entertainment is distributed now. Input-output-intensive computation could be done remotely, renting time on powerful or application-specific processors. Likewise, the adoption of standards for reversible compression would facilitate the inclusion of compression in commercial cryptographic protocols, yielding enhanced security along with the usual advantages of economy.

It is quite possible that inductive tools like neural nets, genetic algorithms, and fuzzy logic will see increasing use in solving the matching problems and in designing the contents of dictionary-based compressors. As it is, schemes such as those of Lempel and Ziv, Elias, and Ryabko employ heuristics to populate their dictionaries; none of them can be proven "ideal." Other schemes might have advantages in convergence rate or other criteria.

One final observation is that, with the dissolution of the Soviet Union, there should be opportunities to attract Russian mathematicians and computer scientists to US universities and laboratories, temporarily or permanently. US universities already have seen an influx of Russian students, but the professional flow is just beginning. For example, Boris Fitingof, once prolific in the Soviet Union, is now at Syracuse University (via Israel). Others will doubtless follow.

F. KEY NON-US RESEARCH PERSONNEL AND FACILITIES

US journals, most notably the *IEEE Transactions on Information Theory* and certain ACM[50] publications, are the premier publishing venues for high-quality work in lossless compression. The fledgling annual Data Compression Conference at Snowbird has, as yet, no counterpart abroad. Researchers in the United States are ideally situated to observe and participate in current work. Since the late 1970s, lossless data compression research has been booming in Israel, Canada, Australia, New Zealand, and Japan. The United States is again fortunate that some of the leading foreign workers are employed in the United States, part- or full-time. Those who are not working in the United States frequently collaborate with US peers.

The institutions responsible for publications vary from country to country. The Canada-Australia-New Zealand group is comprised almost entirely of university faculty, with Calgary University outstanding in Canada, Melbourne University in Australia, and Canterbury and Waikato Universities in New Zealand. Likewise, the Finnish contributors are mainly academics, from Helsinki, Turku University, and Tampere University. The Technion (Israel Institute of Technology) is the center of research in Israel, with contributions from Tel Aviv University, the Weizmann Institute, and IBM-Israel. In The Netherlands, both Eindhoven University and the Philips Laboratories are prominent. Others that bear mention are Universités de Rouen and Paris Norde in France, the Universities of Rome and Salerno in Italy. The situation in Japan is widely distributed among a dozen universities and the research laboratories of Fujitsu, Mitsubishi, Toshiba, and others.

50 Association for Computing Machinery

Table III.1 lists the key non-US research personnel active in lossless compression, their affiliations, and the areas relative to this assessment in which they have been working.

TABLE III.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—LOSSLESS COMPRESSION

Researcher	Affiliation	Area of Expertise
	Australia	
R. P. Brent	Australian National University, Canberra	
A. Moffat N. Sharman	Melbourne University, Victoria	Algorithms
R. Williams	Renaissance Software, Adelaide	Algorithms
	Canada	
J. G. Cleary R. M. Neal	Calgary University, Calgary, Alberta	Algorithms, theory, statistics
G. Gabor	Dalhousie University, Halifax, Nova Scotia	
G. Promhous M. Bennet	IBM–Toronto	Statistics
G. V. Cormack R. N. Horspool	Victoria University, Victoria, British Columbia	Algorithms
G. Freeman	Waterloo University, Waterloo, Ontario	Theory
	Finland	
E. Makinen	Tampere University, Tampere	Linguistics
J. Katajainen Timo Raita Jukka Teuhola	Turku University, Turku	Theory
J. Jakobsson	Vaasa University, Vaasa	Linguistics

TABLE III.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—
LOSSLESS COMPRESSION (cont'd.)

Researcher	Affiliation	Area of Expertise
	France	
M. Crochemore D. Perrin	Université du Paris-Nord, Paris	Linguistics
G. Hansel	Université de Rouen, Mont Saint-Aignan	Theory
	Germany	
D. Manstetten	Robert Bosch GmbH	
	Israel	
D. Sheinwald	IBM–Israel, Haifa	Theory
A. Lempel J. Ziv J. Weinberger	Technion, Israel Institute of Technology, Haifa	Theory, algorithms, statistics
A. S. Fraenkel S. T. Klein	Weizmann Institute, Rehovot	Theory, statistics
	Italy	
R. Capocelli (deceased)	University of Rome, Rome	Theory
	Japan	
H. Yokoo	Gunma University	Algorithms
K. Hirota S. Murashima H. Nakamura	Kagoshima University, Kagoshima	
S. Handa H. Tanaka Y. Ueda	Kobe University, Kobe	
H. Iwamoto K. Kobayashi	Research Labs of Fujitsu, Hitachi, Mitsubishi, Toshiba, et al.	
H. Imai Y. Miyakawa J. H. Park Y. Saitoh Y. Takashima	Yokohama National University, Yokohama	

TABLE III.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—
LOSSLESS COMPRESSION (cont'd.)

Researcher	Affiliation	Area of Expertise
	Netherlands	
T. Tjalkens F. M. J. Willems	Eindhoven University of Technology, Eindhoven	Theory, algorithms
	New Zealand	
I. H. Witten G. Cleary	Waikito University	Theory, statistics, algorithms
T. C. Bell	University of Canterbury, Canterbury	Theory, statistics, algorithms
	Russia	
R. Ye. Krichevskiy B. Ya. Ryabko V. K. Trofimov	Novosibirsk Telecommunications Institute (*Novosibirskiy elektrotekhnicheskiy institut svyazi*—NEIS), Novosibirsk	Theory, error correction
Yu. M. Shtar'kov	Problems of Information Transmission Institute (*Institut problem peredachi informatsii)*, Moscow	Theory, error correction
	Switzerland	
J. L. Massey	Swiss Federal Institute of Technology, Zürich	Theory, cryptography, error correction

CHAPTER III: LOSSLESS CODING
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CHAPTER IV
QUANTIZATION

A. SUMMARY

Development activity in quantization algorithms outside the United States emphasizes traditional scalar quantization combined with predictive coding, transform coding, or block truncation coding; it is comparable in quality and content to analogous work in the United States. Non-US developers in this field have devised a variety of minor improvements and hybrid techniques in recent years, but have made no more significant breakthroughs in terms of implementation complexity or distortion/rate tradeoffs than have their US colleagues. Activities in vector quantization techniques in the United States and abroad are also of comparable quality and scope. These include applications of Hopfield and Kohonen neural net ideas to the design and implementation of vector quantizers. In both scalar and vector quantization algorithm development, the primary non-US work is being done in Japan, South Korea, Germany, the United Kingdom, and Italy, with notable smaller efforts in Taiwan, Hong Kong, France, and Spain.

Western and Eastern Europe, Israel, and Japan have made significant contributions to the theoretical aspects of quantization research. However, recent developments outside the United States in the traditional asymptotic theories due to Shannon (information theory) and Bennett and Zador (high-rate quantization theory) have been primarily of mathematical interest, and have not suggested new fundamental insights or supported fundamental breakthroughs in code design. Research outside the United States has led to no theoretically motivated practical improvements comparable to lattice quantizer and trellis coded quantizer improvements recently demonstrated in the United States.

The area of segmentation coding has been considerably more active outside the United States. A variety of promising results have been obtained by Swiss and Canadian researchers in this area, which has received little attention in the Pacific Rim or in the United States. These codes are computationally intensive, but they yield the highest verifiable compression ratios reported in the literature.

Non-US research on fractal-based compression techniques has largely followed US work and, as in the United States, has been primarily promoted in the popular scientific press (with occasional tutorials in technical journals). The United Kingdom has been the primary site of non-US activity on fractals for image compression, although recently good work has been reported in Norway. Most papers referring to fractal coding are descended from M. Barnsley's "iterated function system," or IFS approach. An alternative technique called "fractal geometry codes" was introduced by researchers from IBM-Israel in 1986. These codes are far simpler than the IFS approach and are more in the spirit of Mandelbrot's original fractal geometry work in that they use the idea of a "traveling yardstick" to fit a piecewise-linear approximation to one-dimensional data. These codes have received little attention in the literature, but the simplicity of the technique and its good performance and theoretical performance bounds that compare well with simulation all suggest that the technique may be well suited to low-complexity applications.

B. INTRODUCTION

This chapter examines non-US research on the fundamental theoretical and algorithmic aspects of quantization (broadly defined). The theory emphasizes the achievable optimal performance in mathematical models of signal sources and the approximate or exact analyses of the effects of quantization error in communication and signal processing systems. The algorithmic side focuses on the development of quantization algorithms with a good balance of computational complexity, performance as measured by some fidelity criterion, and bit rate. This work has usually been done in the context of particular applications, such as speech or image compression, and several topics might equally well be considered in one of the subsequent chapters devoted to the specific applications. It is included here if the primary contribution of the work is to the quantization component of a compression system and its ideas have applications more general than those that motivated the developer.

The theory and practice of quantization used for analog-to-digital conversion, for data compression, and for combinations of these operations originated in the United States during the late 1940s and the 1950s. Since that time, theoretical foundations have matured to provide general performance bounds, tools for statistical and systems analysis for common algorithms (used in most current practical systems), and novel ideas for more sophisticated algorithms (used in present advanced applications and ambitious proposed future applications). Most recent non-US work has been performed in the Pacific Rim (especially Japan, South Korea, Taiwan, Hong Kong, and Australia) and Europe (especially Germany, the United Kingdom, France, Italy, Switzerland, and Russia). West European quantization theory researchers publish their work in the IEEE *Transactions* on Information Theory, on Communications, on Image Processing, and on Signal Processing, and in the corresponding *Proceedings* of the British IEE. At least until recently, East European work appeared primarily in the Russian-language journal, *Problems of Information Transmission (Problemy peredachi informatsii)*, and the Czech journal, *Cybernetics (Kybernetica)*. Non-US researchers who develop quantization algorithms and combinations of quantizers with other signal processing algorithms publish their work in *Electronics Letters*, the IEE and IEEE journals mentioned above, in the *Proceedings* of EUSIPCO (the European Signal Processing Conference), and in the Japanese journal *IEICE Transactions*. Many papers have appeared in IEEE and SPIE conferences around the world, which provide a means for non-US researchers to easily reach the English-speaking technical world.

C. DISCUSSION OF NON-US WORK

1. Quantization Theory

After two decades of considerable activity, Shannon source coding theory is relatively mature. The best work on Shannon source coding theory outside the United States is being done in Russia, Czechoslovakia, and Israel, but no major breakthroughs have been reported recently anywhere. M. S. Pinsker and his colleagues at the Problems of Information Transmission Institute, Russian Academy of Sciences, in Moscow, continue to solve new problems in rate-distortion theory, providing theoretical performance bounds for source coding systems under various constraints and using various probabilistic source models (Gorbunov and Pinsker, 1988, 1989). This group (which includes Dobrushin, the two Gelfands, Shtarkov, and Ryabko) has done some of the best mathematical information theory work in the world for many years, but it has had little impact on engineering practice in Russia (or elsewhere). A similar description holds for the Information Theory and Automation Institute, Czech Academy of Sciences, in Prague. For example, Sefl and Vajda (1987) have reported an extension of rate-distortion theory to include mathematical measures of distortion that match subjective distortion better than the ubiquitous squared error, but the promised details have not yet appeared. Such a theory would be useful for evaluating and perhaps designing compression systems matched to human perception. Given Vajda's excellent reputation and the existence of sophisticated speech distortion measures that have proved their usefulness in vector quantization design, the proposed extension of the theory is plausible. It is not clear, without seeing an actual manuscript, how far the development was taken, however.

The Mathematical Institute of the Hungarian Academy of Sciences in Budapest has a long history of contributions to Shannon theory, with a substantial minority of the work falling in source coding theory, the Shannon theory of data compression. Senior workers such as I. Csiszar, K. Marton, and J. Körner are among the best in the world, although their work tends to be highly abstract and has had little direct impact on practice. (Körner is now in Italy and is no longer active in source coding.) Recently, however, G. Lugosi and T. Linder have begun some outstanding work applying Vapnik-Cernovenkis theory to the study of rates of convergence of quantizer design algorithms and to related problems in nonparametric learning and estimation. This work has been done largely in cooperation with US researchers at the University of Illinois (Professor K. Zeger and Dr. A. Nobel) and Stanford University (Professor R. Olshen), but Lugosi and Linder will return to their home institution, and the work is certain to continue there. The work is important because it quantifies the rate at which clustering algorithms for code design converge, providing quantitative estimates of the amount of training data needed for reliable code design and, conversely, quantitive measures of how much one can trust a code designed on a given amount of data. They have also developed results providing an optimal vector size for a vector quantizer designed for a training set of a given size. Larger vectors usually mean better performance, but they mean fewer training vectors from a fixed size learning set, and hence the code will not be as robust. This work provides results on how to balance these effects. Little of this work has been published yet; it exists only as preprints, some of which have been submitted for publication. (A brief description of some aspects may be found in a paper by Linder et al., 1993.) It is the best work of its kind, however, and is likely to lead to further important theory and better clustering-based quantizer design algorithms.

Bennett's asymptotic quantization theory is commonly applied in non-US research to analyze the impact of noise on quantization systems, but there has been virtually no new work to extend or improve the theory itself outside the United States. There has been work to replace the approximations by an exact analysis of the statistical behavior of quantization noise in systems where the Bennett approximations are known to be untrustworthy, especially in quantizers with low bit rates and analog-to-digital converters that include feedback, such as the popular Delta-Sigma or noise-shaping oversampled analog-to-digital converters.[1] The nonlinear nature of this problem has made for slow progress, even using powerful mathematical tools from dynamical systems theory and random process theory. Some of this work has been performed by non-US researchers visiting the United States (for example, Feely[2] and Hein,[3] who have since returned to Ireland and Denmark, respectively). Koski (1993), a Finn working in Sweden, has used ideas from classical ergodic theory to derive several properties of the behavior of quantization noise in these systems. Although none of these results has had a serious impact on the development of new systems or on design methods, they do help explain behavior such as limit cycles and audible tones that is not predicted by the usual linear approximations (incorporated in the Bennett method).

Much of the early fundamental work on universal quantization and universal lossy compression was done by Jacob Ziv at the Technion in Israel. As the previously cited work, this work does not use the Bennett approximation, but it does provide a demonstration of conditions under which the Bennett approximations are consistent with the exact behavior. A descendent of this work is the recent remarkable work of Zamir and Feder (1992, 1993) at Tel Aviv University. They use the properties of dithered quantizers to quantify the tradeoffs between sampling rate and quantizer bit rate in certain analog-to-digital converters incorporating linear filtering, lattice quantizers, and dithering. Their work provides, for the first time, a quantification of the increase in sampling rate required to compensate for a decrease in quantizer bit rate. The corresponding practical problem is important, because in modern circuitry

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1 J. C. Candy and G. C. Temes, *Oversampling Delta-Sigma Data Converters: Theory, Design and Simulation*, IEEE Reprint Volume, IEEE Press, 1991.

2 O. Feely and L. Chua, "The Effect of Integrator Leak in Σ–Δ Modulation," *IEEE Trans. Circuits Systems, 38*, (1991), 1293–1305.

3 S. Hein and A. Zakhor, "Stability and Scaling of Double Loop Σ–Δ Modulators," *Proc. 1992 IEEE ISCAS*, 1992, 1312–1315.

it is often much more difficult to build a quantizer with high bit rate than it is to build a low-bit-rate quantizer that runs very fast. Theoretical results of this type provide a benchmark for comparison with practical systems and may eventually lead to new practical architectures.

2. Spline Fitting Codes

Little new theory on the theory of spline fitting codes has been developed in the United States or abroad, but a fair amount of European work applies spline fitting codes to certain one-dimensional medical waveform compression problems, especially to electrocardiogram (ECG) compression. Techniques range from fitting straight-line (piecewise-linear) approximations to achieve roughly 5:1 compression with acceptably small squared error by Huang et al. (1992) at the University of Sussex, United Kingdom, to 20:1 using high-order polynomial fits by Philips and De Jonghe (1992) at the University of Ghent, Belgium. The latter work permits the algorithm to vary the choice of the interval size and starting point to ensure a better spline fit, and hence this technique can be viewed as a one-dimensional version of the segmentation codes used for image compression. Here, however, the residual is not coded. No comparable work is done by US researchers, who typically use predictive coding techniques or simple piecewise-linear approximations for such applications.

An interesting comparison of the performance of polynomial fitting codes in terms of a variety of quality measurements, including the diagnostic accuracy when the decompressed data is used as an input to an automatic diagnostic algorithm, has been carried out by Bedini et al. (1991) at the University of Pisa, in Italy.

3. Transform and Predictive Coding

The theory and practice of the small number of internationally standard, commercially dominant compression system architectures and design techniques (which perform their quantization step on transformed data or on predictor residuals) are well understood. Nonetheless, research continues in the United States and elsewhere on variants of the well-established techniques that are claimed to have advantages in complexity or performance. The best of such work is sound and provides reasonable improvements over the standard techniques, but the improvements do not constitute breakthroughs. Rather, they demonstrate that the non-US research community is as active as that in the United States at building a toolbox of variations, improvements, and combinations that are potentially useful for custom applications.

Notable examples include the work of Yamane et al. (1987) at Okayama University, Japan, who argue that a discrete sine transform can provide quality comparable to that achieved with the dominant discrete cosine transform at half the complexity when operating on predictor residuals in a predictive coder; Koh et al. (1991) at Nanyang Technological University, Singapore, who show that algorithms involving nonorthonormal integer transforms can be more efficient than the standard algorithms, both in computational efficiency and in concentrating information in lower-order coefficients; and Chen and Weng (1991) at National Chiao Tung University, Taiwan, who replace a single linear predictor by a family of vector predictors and then select the best one in each current context. The latter technique is a form of universal coding where the best predictor is selected prior to quantization.

4. Block Truncation Coding

Many researchers have reported improvements to this basic scheme, for example, Kamel et al. (1991) at the University at Waterloo, Canada, and Lu et al. (1991), at Sussex University, United Kingdom, who rediscovered the original Lloyd[4] properties for optimal quantizers. Hybrids that incorporate block truncation coding (BTC) are often universal coders that use BTC to code edges in an image, and some other scheme like discrete cosine transform coding or vector quantization to code the rest. Other hybrids use discrete cosine transform coding or vector quantization to code the overhead (moment) information or the residual (Udpikar and Raina, 1987; Wu and Coll, 1991; Efrati et al., 1991). The schemes generally provide a good balance of simplicity and quality, but have not seriously challenged the JPEG standard.

Qiu et al. (1991), at Lancashire Polytechnic, United Kingdom, have developed an unusual hybrid that combines a Hopfield neural network classifier with BTC, where the neural net is used to classify the image to be compressed, and then a BTC fine

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4 S. P. Lloyd, "Least Squares Quantization in PCM," 1957, unpublished Bell Laboratories Technical Note. Portions presented at the Institute of Mathematical Statistics Meeting, Atlantic City, NJ, Sept 1957. Published in the Mar 1982 special issue on quantization of the *IEEE Trans. Inform. Theory*.

tuned to that class of image is used, providing performance improvement at the cost of a moderate increase in complexity.

5. Segmentation Coding

Segmentation coding, one of the most promising recent approaches to quantization and compression, is being pursued most vigorously outside the United States. The proposed schemes differ primarily in the way the contours are extracted, the constraints placed on the shapes of the regions, and the techniques used to code the residuals.

One of the oldest approaches is well represented in modern form in the work of Barba and Bertrand (1988) at LRII/IRESTE, Nantes, and INSA, Rennes, France.[5] They use traditional edge detection techniques combined with contour extraction algorithms to form the segmentation. They then use polynomial interpolation for the background of each region, and then finally apply an adaptive discrete cosine transform code to code the regions' contents. They attempt to optimize the choice of regions, polynomial interpolation, and the description of the textures. The performance is good but not startling, in that it attains excellent quality at slightly under 1 bit per pixel (bpp), but the algorithm is complicated and requires full frame processing before compression.

Denatale et al. (1991), at the University of Genova (Genoa), Italy, developed a scheme that selects corner pixels in a quadtree[6] fashion so that a simple bilinear interpolation between the pixels yields a good overall reproduction of the image in a mean-squared sense. Corner pixels are sent at full rate and the reproduction formed by interpolation between these pixels within each segment. The algorithm yields good visual quality and 28.7 dB at under .2 bpp for monochrome images. Denatale (1993) combines the bilinear interpolation with a discrete cosine transform coder for the residuals, resulting in higher quality at the cost of a higher bit rate. The general approach is conceptually simple and effectively yields both a very low-rate, low-

5 LRII is the Laboratoire de Robotique et d'Informatique Industrielle, a research laboratory associated with IRESTE, the Institut de Recherche et d'Enseignement Superieur aux Techniques de l'Electronique. This Laboratory has been replaced by Laboratoire d'Analyse et Traitement des Images (LATI).

6 H. Samet, "The Quadtree and Related Hierarchical Data Structures," *ACM Computing Surveys, 16,* (Jun 1984), 188–260.

resolution reproduction and a high-rate, high-quality reproduction, making it useful in multirate applications.

M. Kunt and his colleagues at the Ecole Polytechnique Fédérale de Lausanne (EPFL) in Switzerland are perhaps the best known and most prolific proponents of this approach to image and video compression. They pioneered the "split-and-merge" segmentation approach, in which an image is successively decomposed into regions in a manner that optimizes the reconstruction quality when polynomial interpolations between the region boundaries are used to represent the image (Kunt et al., 1987). They have also extended the method to image sequence coding by segmenting three-dimensional blocks (Willemin et al., 1991). Reed et al. (1991) have considered more sophisticated model-based segmentation techniques at EPFL. These methods are among the best available in the sense of optimizing quality for a given bit rate, and fair quality has been reported at compression ratios of 100:1 from 8 bpp original video. The enormous computational requirement of the approach makes it impractical for real-time implementation using current technology. As computational power increases and algorithms are streamlined, however, these techniques may become much more attractive for future very-low-bit-rate image compression systems. Research on segmentation codes has accelerated recently in the United States, especially since Reed moved from EPFL to the Image Processing Institute at the University of California at Davis.

Wu (1992), at the University of Western Ontario, Canada, has extended and optimized the split-and-merge idea by applying looser constraints on the shapes of the regions and optimizing the segmentation to improve the bit-rate/mean-squared error tradeoffs. Each region is simply represented by the centroid of its pixels, but good quality can be achieved because of the flexibility of the segmentation. Wu comes from a computer graphics background, a fact that has influenced the style of his algorithms, which improve on predecessor techniques in elegance, performance, and simplicity.

Arduini et al. (1990), at the University of Genova (Genoa), combine a split-and-merge segmentation algorithm using a polynomial fit with a Kohonen vector quantizer for the residuals to achieve a claimed compression of 60:1 and greater on ordinary still images. Their algorithm is quite sophisticated, incorporating prediction and finite-state vector quantizer techniques. Similar performance is reported by

Carlsson (1988) at the Royal Institute for Technology, Sweden, who uses a contour extraction segmentation, maximally smooth spline fitting interpolation, and a Bert-Adelson Laplace pyramid coding of the residuals. Both of these schemes are essentially variations on the EPFL techniques. They use more complicated residual coders and thereby achieve better quality, but the compression is somewhat less.

The segmentation codes can be summarized as attempting to achieve compression from 8 bpp to approximately .1 bpp for still-frame monochrome images (more compression for color or for video) while achieving good reproduction quality of 27-dB signal-to-noise ratio or more. Their computational complexity remains the primary shortcoming of these techniques.

6. Vector Quantization

a. Lattice and Trellis Quantizers

Surprisingly little activity on lattice and trellis quantizers is apparent outside the United States, in spite of their demonstrated relatively favorable balance between complexity and performance and their ability to control overload and granular noise. Penzhorn (1988) in South Africa has written a general survey of the topic, which suggests commencement of activity. (Penzhorn has proposed South Africa as a location for a future IEEE Workshop in Information Theory to the IEEE Information Theory Society.) Herbert (1991) at the University of Erlangen-Nürnberg, Germany, has developed a quantization scheme that has interesting supporting theory and good potential for application to speech coding. Following his mentor Brehm's approach of modeling speech as a spherically invariant random process, Herbert has developed a simple transform applicable to such processes that renders the statistical distribution more uniform, improving the performance of a succeeding simple lattice code transform. Application of many of the most popular lattices with his fast search algorithms for the encoding produces good performance, but it is not clear if the performance gain is worth the significant added complexity of the transform. Optimization of the subset of the lattice chosen for the codebook might work equally well at greatly reduced implementation complexity. Antonini et al. (1992) and Senoo and Girod (1992) use lattice codes in their work on wavelet transform image compression, a topic considered in more detail in Chapter V. The work by Senoo and Girod was originally done in the United States at the MIT Media Lab, but Senoo is now with the Audio and Video Research Center of Matsushita in Osaka, Japan, and Girod is with the Academy of Media Arts in Köln (Cologne), Germany, where both are continuing work on these ideas. More recently, Barlaud et al. (1993) showed that a lattice code with elliptical support was particularly well-suited for coding wavelet coefficients.

b. Clustering Algorithms

Most recent activity in development of clustering algorithms for application to data compression, inside and outside the United States, has concentrated on modern methods of clustering that are not part of the classical statistical literature (although they can be regarded as statistical signal processing algorithms). Most notable examples of this work employ neural nets and simulated annealing.

Many research groups in the United Kingdom, Italy, Greece, Sweden, and France have used Hopfield neural nets to design and implement vector quantizers (as have many groups in the United States). Much of this work is straightforward, but the following are examples of novel contributions that are unique to non-US research. Luttrell (1991) at the UK Defence Research Agency has developed a theory of self-supervised vector quantizer design and related it to the Lloyd clustering methods. His algorithms can be applied to the design of hierarchical vector quantizers and vector quantizers in networks. This theory provides useful insight into the probabilistic structure of good codes. Easton and Goodyear (1991) at Liverpool University, United Kingdom, have applied these ideas to the design of code-excited linear prediction (CELP) speech coders to demonstrate that a few iterations of a neural net could provide good residual codewords that yielded an overall quality indistinguishable from that achieved using exhaustive search techniques. This provides a reduction in algorithm complexity even if the neural net is implemented in software. Marsi et al. (1991) at Telettra SpA, Milan, Italy, pre-classify inputs according to a local activity index and then provide different neural nets for further classification within each category. Less active blocks can be coded by simple neural networks (fewer layers) using fewer bits. They also observe that a neural network can efficiently perform a transform to obtain only the few high-energy coefficients required for coding faster than a traditional fast transform algorithm can obtain a full transform (which contains unneeded coefficients). A similar observation was made by Yan (1991) at the University of Sidney, Australia, who explicitly developed a neural network to speed up transform coding by computing only those transform coefficients that are actually to be coded. (Conventional transform coders use a fast algorithm to compute all the coefficients, then code only those with significant energy.) Yan has developed a circuit implementation of the neural network part of his system. The idea of using a neural net to compute only those transform coefficients to be quantized is interesting, but it is not clear if there is any genuine advantage over traditional fast transform methods.

The work of Wu and Fallside at Cambridge University, United Kingdom, is among the best work world-wide on combination of Kohonen neural networks and vector quantization (VQ). By careful analysis and comparison of the Kohonen approach with the Lloyd approach, they are able to determine optimal update formulas for the Kohonen learning algorithm with essentially the same form as the Lloyd optimality conditions. This provides faster converging algorithms, as well as insight into the relative advantages and disadvantages of the two approaches. They have applied their ideas both to the design of vector quantizers and to the development of a novel form of neural network, the codebook-excited neural network (Wu and Fallside, 1991, 1992). Fallside is a well-known senior researcher in information theory and speech coding, as well as in neural networks.

Burel and Pottier (1991) at the Rennes Electronics Lab, Thomson-CSF, examining the topological structure of the Kohonen codes, have observed that Kohonen vector quantizer is robust against channel errors, because close binary channel codewords imply good reproduction. These ideas, which provide a simple approach to some important problems of joint source and channel coding, were anticipated by US researchers.[7] Similar work was also done independently at the University of Naples by G. Poggi (1993) and at EPFL by M. Kunt's group (unpublished).

A more recent project using the Kohonen clustering ideas to optimize VQ compression systems for noisy channels has been initiated by Per Hedelin (1992) and his students at Chalmers University of Technology in Göteborg, Sweden. The idea is

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7 J. McAuliff, L. Atlas, and C. Rivera, "A Comparison of the LBG Algorithm and Kohonen Neural Network Paradigm for Image Vector Quantization," *Proc. ICASSP, I-101–I-107* (1988), 2293–2296.

E. A. Riskin, L. E. Atlas, R. Ladner, and R.-Y. Wang, "Index Assignment for Progressive Transmission of Full Search Vector Quantization," submitted to *IEEE Trans. Sig. Process.*

to make the compression system more robust against channel errors, an idea similar to that described previously. His group has a long string of contributions to using VQ ideas in speech coding.

Koenig and Glesner (1991) at the Darmstadt University of Technology, Germany, claim to have achieved significant performance improvements for image compression (relative to the more conventional Lloyd design algorithm) using the Kohonen clustering algorithm. These results must be taken with a certain skepticism, as they exceed by several decibels those reported elsewhere using similar ideas. In particular, the previously described work of Wu and Fallside suggests that, for equivalent initial conditions, the overall performance achievable in the sense of distortion/bit-rate tradeoffs is the same. The advantage of a suitably optimized Kohonen clustering is convergence speed and the topological structure of the resulting codebook.

Rodrigues-Fonollosa et al. (1990) have developed a novel application of the Kohonen idea that applies Kohonen adaptive updating to multistage vector quantizers in which each stage quantizes the residual error produced by the preceding stages. In a two-stage quantizer, the quantized error for the first section is used in a least-mean-squared error update formula, so the decoder can track the adaptation of the first stage. This is an interesting idea and the authors claim excellent performance in CELP and multipulse speech coders.

A further neural net technique used to design VQs is the competitive learning algorithm of Rumulhart and Zipser.[8] Ta et al. (1991) at Sidney University designed VQs for image compression using the competitive learning algorithm with a frequency-sensitive distortion measure, that is, a distortion measure that included a weighting of the relative frequency of codeword use in order to prevent codewords from being underused. Fjällbrant et al. (1992) have an ongoing project studying the relationships between VQ and neural network models and combining the two methods to design both tree-structured and full-search VQs having reduced memory and complexity requirements. The VQ is implemented as a multilayer perceptron network, and a frequency-sensitive competitive learning algorithm is used for the

8 D. E. Rumulhart and D. Zipser, "Feature Discovery by Competitive Learning," *Cognitive Science*, *9*, (1985), 75–112.

design. This work is primarily of interest as it demonstrates an alternative neural net approach to VQ design, but neither group provides sufficient information in their papers to accurately judge the relative merits of this approach. One thing the frequency-sensitive idea accomplishes is that it produces a relatively uniform relative frequency of codewords. Although intuitively appealing, this is, in effect, a "maximum entropy" approach that is a bad idea in source coding: a skewed distribution with a lower entropy would permit further compression by subsequent lossy compression. This fact was realized by NTT's Veda and Nakano (1993), who modified the design to force the codewords to contribute equally to the average distortion rather than to have equal probability. Gersho[9] has shown that when the bit rate is large, optimal vector quantizers indeed have this equal distortion property, so that the Veda and Nakano variation on the competitive learning design algorithm yields nearly optimal quantizers. The algorithm compares quite favorably with other clustering techniques.

Simulated annealing has been used as a vector quantization clustering algorithm in the United States and abroad, because it can avoid the pitfalls of local minima. The approach has not been popular because convergence of the design can be extremely slow, but it is a useful research tool as it usually provides the best possible rate/distortion tradeoff for a given vector quantizer structure. Kodama et al. (1991) at the Kyoto Institute of Technology, Japan, are proponents of this approach. Lech and Hua (1992) at Melbourne University, Australia, have studied a variety of variations to the basic approach (as well as both competitive learning and Kohonen neural net clustering). Their work provides a useful survey of both areas and a nice summary of the relative advantages and disadvantages of the different approaches.

c. Fast Search and Tree-Structured Algorithms

Most work on fast search techniques for vector quantization concentrates on a variety of speedup tricks like partial distortion computation, ordered searches, creative use of the triangle inequality, and limiting the search to codewords with similar moments. These are mostly descendants of the techniques first developed in

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9 A. Gersho, "Asymptotically Optimal Block Quantization," *IEEE Trans. Inform. Theory*, *25*, (1979), 373–380.

the United States by Gersho and his students at UCSB.[10] Good examples of several variations on these techniques may be found in the work of Ngwa-Nidifor and Ellis (1991) at London City University, United Kingdom, of Hsieh and his colleagues at Chung Cheng Institute of Technology, Taiwan (Hsieh et al., 1991a-b; Chen et al., 1991), and Kamel and his coworkers at the University of Waterloo, Canada (Kamel and Guan, 1992). The United States led in the original development of many ideas, but the non-US work now shows more activity and has produced several minor improvements.

Lopez-Soler et al. (1990) at the University of Grenada, Spain, have developed a variety of "bottom-to-top" tree-structured search algorithms that are quite competitive with the products of similar work in the United States. The work at Grenada is expected to be enriched by Lopez-Soler's recent visit with Nariman Farvardin at the University of Maryland, another active researcher in this area.

Chang et al. (1992) at National Tsing Hua University, Taiwan, have developed novel modification of tree-structured vector quantizers in which they are able to adaptively update the codebook of the quantizer without transmitting the new codewords as side information. This appears to be the first development of a means of adapting a tree-structured code to local behavior without requiring a rate increase. In a system that combined their quantizer with a multipath search algorithm, they obtained a 2.5-dB improvement over traditional tree-structured vector quantizers of equal bit rates at the cost of a moderate increase in complexity. This work compares favorably with leading US multi-stage vector quantizer work at the University of Utah and Georgia Tech. This is one of only two projects found dealing with adaptive tree-structured VQ. An alternative approach (which is more *ad hoc*) was developed by Lavagetto and Zappatore (1992) at the University of Genova (Genoa), Italy. They developed an adaptive growing and pruning procedure for tree-structured-VQ based on a principal components analysis to do three-dimensional VQ of video sequences (two spatial and one temporal dimension). The work is both clever and promising, but few hard comparisons were provided with competing techniques.

10 D. Y. Cheng and A. Gersho, "A Fast Codebook Search Algorithm for Nearest-Neighbor Pattern Matching," *Proc. ICASSP*, (1986), 265–268.

D. Y. Cheng, A. Gersho, B. Ramamurthi, and Y. Shoham, "Fast Search Algorithms for Vector Quantization and Pattern Matching," *Proc. ICASSP, San Diego*, (1984), 911.1–911.4.

Ra and Kim (1991) at the Korea Advanced Institute of Science and Technology (KAIST) propose a clever scheme wherein they first confine the search to only those codewords having similar weights as measured by their sample mean. Many codewords are eliminated from the search by a simple application of the Cauchy-Schwartz inequality of real analysis. The authors claim a four-to-one speedup over the partial distortion method.

d. Transform Vector Quantization

Non-US research in which the scalar quantizers in transform coding are replaced by vector quantizers has a stronger theoretical component than the effort in the United States. In much of this non-US work, transformed signals are modeled as spherically invariant random processes (with multidimensional distributions that are constant for fixed-magnitude vectors). This robust model for the data has suggested development of efficient, low-complexity gain/shape vector quantizers that separately code magnitude and shape. Notable work of this type includes that of Saito et al. (1987, 1988) at Kanagawa University, Brehm and Herbert (1990) at the University of Erlangen, and Rosebrock and Besslich (1992) at the University of Bremen. Po and his colleagues at the City Polytechnic of Hong Kong have taken an alternative approach, using the energy compaction property of transform codes to approximate the average distortions in a small-dimensional subspace (Po and Chan, 1990, 1991a-b; Chan and Po, 1992; Chan et al., 1991; Po, 1993). Their vector quantizer can be implemented as either an ordinary vector quantizer with a transform domain distortion measure or as a transform code, the primary purpose of their approach being to reduce search complexity by reducing the dimensionality of the distortion computation. They do not, however, take adequate account of the added complexity required by the transform, which may outweigh the savings of the lower dimension. Their approach may be useful when applied to transform codes or in trying to speed code design, but it seems of little use as a complexity reducing tool for ordinary vector quantization. Its use for transform codes also seems limited, because, once in the transform domain, most systems already effectively use the same reduction by not using low-energy transform coefficients in distortion computations. In the earlier version of this work, they used a Hadamard transform as a simple means of detecting edges in images, allowing them to adapt their compression algorithm to local image behavior (Po and Chan, 1990; Chan et al., 1991b). These well-motivated efforts achieve good performance/complexity balance, but the Saito and Brehm group efforts are stronger in both soundness of theory and likelihood of eventual practical implementations. An alternative approach by D'Alessandro et al. (1993) is to use Fischer's pyramid VQ[11] to code the discrete cosine transform (DCT) coefficients, a good match since the quantizer is designed for a Laplacian distribution, which is a good approximation to DCT coefficient behavior. These researchers claim quality comparable to the CMTT/2 standard, while requiring only half the bit rate.

e. Classified and Adaptive Vector Quantization

Many traditional schemes can be improved by classifying the input signal's behavior and then applying a custom compression algorithm. Following its original application to VQ by Ramamurthi and Gersho,[12] this approach has been a popular research tool for image compression. Different researchers use different classification rules and different code design techniques (for each signal class). One especially novel classified VQ is that of Kubrick and Ellis (1990) at the City University, London, who apply the pairwise nearest neighbor clustering algorithm of Equitz to a classified learning set to form codebooks with built-in good bit allocation. The technique is simple and fast and the bit allocation algorithm is among the most intelligent that has been reported. Davignon (1990) at the Ecole Centrale des Arts et Manufacturière, France, classified edges by binary template matching to thresholded gray-scale images that had been rescaled to a standard dynamic range. Tu et al. (1988) at the Catholic University, Leuven, Belgium, used classification based on motion compensation performance to choose between standard motion compensated DCT coding and VQ for abrupt scene changes in video coding. Ngan et al. (1989) at the National University of Singapore reduced the side information necessary for communicating the chosen class (and hence the specific codebook) by predicting based on contiguous coded blocks and sending only a small amount of correction information. The scheme is an improvement in that the Hadamard transform edge detector works well for classification and the bit rate is reduced by the prediction, but the technique is perhaps better viewed as a finite-state VQ where the classification is based on the coded neighbors rather than on the block to be coded.

11 T. Fischer, "A Pyramid Vector Quantizer," *IEEE Trans. Inform. Theory*, *IT-32*, (1986), 568–583.

12 B. Ramamurthi and A. Gersho, "Classified Vector Quantization of Images," *IEEE Trans. Commun.*, *COM-34*, (1986), 1105–1115.

f. Predictive and Finite-State Vector Quantization

Predictive VQ and finite-state VQ use past-coded blocks to determine which code will be used for the current block, either by forming a prediction residual for coding or by selecting one of a finite number of available codebooks based on context. Itoh et al. (1989) at the University of Tokyo take advantage of the Laplacian behavior of prediction residuals to derive a novel predictor structure that depends on the previously coded pixels only near the borders of the current block. Lee and Chan (1992), at the City Polytechnic of Hong Kong, apply finite-state VQ ideas to quantizing in color space, the idea being to optimize color display on cheap low-rate color monitors. Both schemes are competitive in performance and complexity with the many predictive schemes developed in the United States.

g. Multiresolution Vector Quantization

The general topic of multiresolution vector quantization will be treated later in this report, where it can be discussed more naturally in conjunction with our examination of signal decompositions in Chapter V. However, two non-US contributions that stand out because of their extreme simplicity (they do not require any of the mathematical power typical of wavelet schemes) will be discussed here. Yamada and Tazaki (1991) at Ehime University have developed an algorithm of "top-down" pyramidal design (contrasting with the bottom-up design of the classical Burt and Adelson pyramid). Their hierarchical vector quantizer is of quadtree structure, with simple piecewise-constant interpolation by centroid. Their single codebook for residuals at all levels bears an intuitive resemblance to wavelet decomposition, but it is obtained by clustering. It would be interesting to see if these inherently digital "wavelets" formed by clustering bear any resemblance to those produced by the currently active theoretical research. Yates and Ivey (1991) use an even simpler mean pyramid scheme for coding frame-to-frame pixel differences in image sequences to construct a video coder that they claim is two orders of magnitude less complex than the international standard, while delivering comparable quality. Their pyramid scheme is not new, but they demonstrate that the extremely simple decomposition is amenable to real-time inexpensive video Codec implementation using Actel gate array devices.

h. Miscellaneous Vector Quantization

We collect here discussions of non-US contributions to vector quantization research and development that merit notice, but do not fit within any of the usual categories. Park and Lee (1992) at Seoul National University, South Korea, have developed design algorithms for vector quantizers based on projections of image data. Their work is intended to be primarily of theoretical interest, but because such projections correspond very directly to the line integrals that constitute the acquired data in computerized tomography (CT) imaging systems, the work may have important practical applications. Their work suggests the possibility of compressing the raw data in such systems, rather than first synthesizing an image and then compressing that. Conceivably, this could make storage and eventual reconstruction of these images more efficient and faster. No one else has taken this approach. Huguet and Torres (1990) at the Cataluna Polytechnic University, Spain, apply vector quantization to signals with both spatial and temporal components, with each vector having pixels from a corresponding block in several successive frames. This technique was pioneered several years ago by the MIT Media Lab, but Spanish participation in this research is evidence of growing activity to develop sophisticated compression techniques in Spain.

7. Fractal Codes

a. Iterated Function System Codes

A popular press account by Wright (1992) at UK Iterated Systems—claiming that fractal-based codes would be the *de facto* international image compression standard (displacing the JPEG standard) by 1993 because of their excellent quality at compression ratios of 70:1 and their low complexity—indicates that highly enthusiastic promotion of fractal-based codes is not unique to the United States. The claimed performance is upped to 100:1 by McKeon (1992), who is the Managing Director of Iterated Systems, Ltd., the UK subsidiary of Iterated Systems. As in similar US presentations, no details of the proprietary algorithm are given, making independent verification of the claims impossible. The compression ratio is, in fact, achievable by segmentation codes, so that the utility of fractal-based codes eventually may be judged by a comparison of their performance, robustness, and computational complexity with that of segmentation compression codes. Nonetheless, it is indisputable that fractal coding ideas have become popular in the United Kingdom. Barnsley, the inventor of iterated function system (IFS) codes (while at Georgia Tech and his company, Iterated Systems, Inc., in Atlanta, Georgia), is a graduate of Oxford, and he helped promote the approach during a visit to the United Kingdom. His conference presentation provided fundamentals background on the approach and promoted his company's hardware, but few details on actual coding algorithms (Barnsley, 1990).

Freedland and Durrani (1991) at Strathclyde University, United Kingdom, provide a good survey of the applications of iterated function systems to modeling, but their discussion of applications to compression records little more than wishful thinking. An examination by Beaumont (1991) at BT Laboratories, Martlesham, United Kingdom, of public domain IFS compression algorithms had far more modest findings, indicating that these algorithms are not competitive with transform coding in either performance or complexity. The mathematics and implementation of these compression algorithms have been well developed in recent years by Jaquin in the United States,[13] and by Ramstad and his colleagues in Norway (Øien et al., 1991, 1992a-b). These techniques are extremely expensive computationally on the encoding end, but Ramstad and his colleagues have found low-complexity decoders. They admit, however, that the decoders are more complicated than those of unconstrained vector quantization. At Tampere University in Finland, Raittinen and Koski (1993), studying IFS codes for binary images, report high compression ratios; and Thomas and Deravi (1993) modify Jaquin's method to permit irregularly shaped transformations and report improvements in compression. Recent work by D. Monro (1993) at the University of Bath, United Kingdom, proposes extensions of the Barnsley approach and describes low-complexity decoders. The quality of these coders is comparable to that of existing standard methods of far less complexity. At best, they appear to provide quality comparable to JPEG with a simpler decoder and far more complicated encoder. The dream that fractals will be as important for image compression as they have become for image generation remains elusive.

13 A. E. Jaquin, "A Novel Fractal Block-Coding Technique for Images," *Proc. ICASSP*, (1990), 2225–2228.

b. Fractal Geometry Codes

Zhang and Yan (1991) at the University of Sidney used interline prediction with residual error coding to obtain the best performance yet reported using this technique, a signal-to-noise ratio of 29 dB on the standard USC Lena image at 8/13 bits per pixel. The work improves on US work on the same subject.[14]

It is difficult to judge the importance of these results precisely, because the computational complexity of these algorithms relative to standard techniques is not quantified. They seem less demanding than the IFS codes, with computational demands of the same order of magnitude as traditional techniques. Reported performance, comparable to the best available international standard techniques and to vector quantization methods, makes this work quite interesting. On the other hand, it is notable that the originators of the technique at IBM in Israel have not pursued it further. All of the recent publications by Wallach and Karnin (see Section II.C.8) have been on different subjects (lossless coding and neural networks).

D. PROJECTIONS FOR THE FUTURE

It seems unlikely that there will be major breakthroughs (in the United States or abroad) in the theory or practice of quantization in the next five years comparable to those seen during the 1980s, which included refinement of transform codes into JPEG, development of a variety of vector quantization algorithms, achievement of high compression ratios using segmentation codes, development of powerful and promising subband and wavelet codes, development of simple lattice and trellis quantizers that are effective against overload quantization noise, and intelligent use of signal processing techniques (such as prediction and classification) to improve performance in any quantization algorithm. Non-US developers, like those in the United States, are likely to continue to make minor improvements in a variety of algorithms, and to find that various combinations of the general approaches are particularly suitable for particular applications. In the United States and abroad, those who study the nonlinear theory that describes the behavior of quantizer error

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14 K. Yang, L. Wu, and M. Mills, "Fractal Based Image Coding Scheme Using Peano Scan," *ISCAS '88*, 1988, 2301–2304.

B. Goel and S. Kwatra, "A Data Compression Algorithm for Color Images Based on Run-Length Coding and Fractal Geometry," *Proc. ICASSP*, (1988), 1253–1256.

and quantifies the bit-rate/distortion tradeoffs in quantization systems are likely to continue slow development of that theory and to use it to suggest new architectures, as those who studied Shannon and Bennett theories once did.

E. KEY NON-US RESEARCH PERSONNEL AND FACILITIES

Table IV.1 lists the key non-US research personnel active in quantization research, their affiliations, and the quantization-related areas in which they have been working.

TABLE IV.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—QUANTIZATION

Researcher	Affiliation	Area of Expertise
	Australia	
Y. Hua M. Lech	Melbourne University, Melbourne	Simulated anneal and neural networks for VQ
H. Yan N. Zhang	University of Sidney, Sidney	Fractal geometry codes, Hopfield neural networks
	Canada	
X. Wu	University of Western Ontario, Ontario	Segmentation coding
G. Freeman J. Mark M. Kamel C. T. Sun J. Hanson Z. Wang	University of Waterloo, Waterloo, Ontario	Tree and trellis coding, speech coding Block truncation coding; Kohonen VQ
	Czechoslovakia	
O. Sefl I. Vajda	Czechoslovak Academy of Sciences	Shannon theory

TABLE IV.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—
QUANTIZATION (cont'd.)

Researcher	Affiliation	Area of Expertise
	France	
D. Barba J.-F. Bertrand	LRII/IRESTE, Nantes	Segmentation coding
G. Burel I. Pottier	Laboratoires Electroniques de Rennes, Thomson-CSF, Cesson-Sevigne	VQ using Kohonen
M. Antonini M. Barlaud	University of Nice Sophia-Antipolis, Nice	Wavelet VQ
	Germany	
H. Brehm M. Herbert	Erlangen-Nürnberg University, Nürnberg	Lattice VQ
B. Girod	Köln Institute of Media Arts, Köln	Wavelet VQ
	Hong Kong	
C.-K. Chan W.-F. Lee L.-M. Po	City Polytechnic of Hong Kong, Kowloon	Low-complexity and finite-state VQ
	Hungary	
I. Csiszar T. Linder G. Lugosi K. Marton	Mathematical Institute of the Hungarian Academy of Sciences, Budapest	Shannon theory; learning theory
	Ireland	
O. Feely	Trinity College, Dublin	Nonlinear dynamical approach to quantization theory
	Israel	
J. Ziv (and students)	Technion, Israel Institute of Technology	Shannon theory; quantization theory; universal coding theory
M. Feder R. Zamir	Tel-Aviv University, Tel-Aviv	Quantization theory

TABLE IV.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—QUANTIZATION (cont'd.)

Researcher	Affiliation	Area of Expertise
	Italy	
F. Arduini F. G. B. Denatale G. S. Desoli D. D. Giusto F. Lavagetto S. Zappatore	University of Genova, Genova (Genoa)	Segmentation coding Adaptive tree-structured VQ for video coding
G. Poggi	University of Naples, Naples	VQ, Kohonen VQ, address-VQ
P. D'Alessandro	Contraves Italiana, SPA, Rome	Pyramid VQ for videocoding
	Japan	
S. Tazaki Y. Yamada	Ehime University, Matsuyama	Multiresolution and successive approximation VQ
K. Aizawa H. Harashima T. Saito H. Takeo	Kanagawa University, Yokohama	Adaptive DCT coding with gain/shape VQ
M. Kasahara H. Kodama K. Wakasugi	Kyoto Institute of Technology, Kyoto	Simulated annealing VQ
N. Veda R. Nakano	NTT Communications Science Labs, Kyoto	Competitive learning neural net VQ
H. Hamada Y. Morikawa N. Yamane	Okayama University, Okayama	Sine transform coding
S. Furui T. Saito S. Itoh I. Naitoh T. Utsunomiya	University of Tokyo, Tokyo	VQ applied to speech compression and recognition Adaptive DCT using VQ Adaptive VQ

TABLE IV.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—
QUANTIZATION (cont'd.)

Researcher	Affiliation	Area of Expertise
	Norway	
S. Lepøsy G. E. Øien T. A. Ramstad	Trondheim	Attractor codes
	Russia	
R. L. Dobrushin M. S. Pinsker	Information Transmission Problems Institute, USSR/Russian Academy of Sciences, Moscow	Shannon theory
	Singapore	
S. Koh	Nanyan Technological University, Singapore	Transform codes
	South Korea	
J. K. Kim S. W. Ra	Korea Advanced Institute of Science & Technology (KAIST), Seoul	Fast search
C. W. Lee H. B. Park	Seoul National University, Seoul	VQ of projection data
	Spain	
J. M. Lopez-Soler A. Peinado-Herreros V. Sanchez-Calle J. C. Segura-Luna A. J. Rubio-Ayuso	Granada University, Granada	Tree-structured VQ
	Sweden	
P. Hedelin	Chalmers University, Chalmers	Channel optimized VQ, Kohonen clustering for VQ
S. Amirijoo T. Fjällibrant F. Mekuria	Linköping University, Linköping	
	Switzerland	
M. Kunt	Ec[illegible] Polytechnique Federale de Lausanne (EPFL), Lausanne	Image and image sequence coding

TABLE IV.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—QUANTIZATION (cont'd.)

Researcher	Affiliation	Area of Expertise
	Taiwan	
J.-C. Chang C.-H. Hsieh P.-C. Lu	Chung Cheng Institute of Technology	Fast search techniques
L.-H. Chen S.-F. Weng	National Chiao Tung University, Hsinchu	Transform coding
S.-H. Chen W. M. Hsieh		Fast VQ
R.-F. Chang W.-T. Chen J. S.-Wang	National Tsing Hua University, Hsinchu	Adaptive tree-structured vctor quantization
	United Kingdom	
D. M. Monro	Bath University, Bath	IFS or fractal codes for image compression
F. Fallside L. Wu	Cambridge University, Cambridge	Kohonen neural nets and VQ
T. Ellis A. Kubrick J. Ngwa-Nidifor	City University, London	Classified VQ, fast search
S. P. Lutrell	Defence Research Agency, Malvern	Neural nets and VQ
M. G. Easton C. C. Goodyear	Liverpool University, Liverpool	Hopfield nets for speech compression
P. N. H. Davies M. P. Gough W. W. Lu	Sussex University, Brighton, Sussex	Block truncation coding
L. Thomas F. Deravi	University of Wales, Swansea	Fractal image compression

CHAPTER IV: QUANTIZATION
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CHAPTER V
SIGNAL DECOMPOSITIONS

A. SUMMARY

Research on signal decompositions useful for data compression is active worldwide. The focus is on finding better transforms, subband and wavelet expansions, reducing implementation complexity, and building complete compression systems taking advantage of inherent features of the decomposition. Good non-US work is done in Europe (France, Germany, England, Scandinavia, Belgium, Switzerland), the Far East (Japan, South Korea, Taiwan), Australia, and Brazil.

Some key ideas in subband coding and wavelets have come out of Europe. France, in particular, has been at the forefront of the work on wavelets, initially from mostly a theoretical point of view, with a lesser view on applications. This theoretical maturity has also led to interesting practical investigations, for example, demonstrating that wavelet filters are potentially the best filters for subband coding.

Research groups are active in filter design for subband coding in Europe (France, Belgium, Norway, Switzerland) and in the Far East (Japan, Taiwan). They are concerned with all aspects of filter design for compression, including objective and perceptual criteria as well as complexity.

Compression of speech, music, images, and video based on various signal decompositions is active worldwide, with interesting contributions from Europe (France, Germany, Norway, The Netherlands, Switzerland) and the Far East (Japan, South Korea). Often, large-scale projects with government funding lead to investigations in which a large number of researchers collaborate, with mixed results. A successful example is digital audio broadcast based on MUSICAM (masking pattern adapted universal subband integrated coding and multiplexing) in Europe. MUSICAM is a standard for CD-quality digital audio compression using subband coding. It has been developed in Europe, and products are likely to come from Europe and the Far East.

Overall, non-US work on decompositions for compression compares with the US work in the following sense:

- Some groups (for example, in France) have contributed theoretical results comparable to US work. Often, collaborative efforts on this type of theoretical work cross international borders.

- Work on application of decompositions to actual compression is very active worldwide, in both quantity and quality. While the United States might have a slight edge in terms of originality and quality, when it comes to more practical schemes, work in Europe and the Far East at least matches the US effort.

B. INTRODUCTION

This chapter discusses non-US research on signal decomposition for compression.

Signal decompositions used in data compression can be seen as a preprocessing stage that is followed usually by quantization (scalar or vector based) and lossless compression. These signal decompositions are usually applied on sampled signals that are often already quantized in amplitude (but with fine quantization, as compared to the follow-up coarse quantization for compression). The decompositions essentially come in three forms:

- **Block transforms.** The sequence of samples is divided into blocks of N non-overlapping samples, and each block is transformed independently of its neighbors by a size N transform that is usually orthogonal. The expansion operator is thus a block diagonal matrix.

- **Overlapping block transforms or subband and wavelet coding.** This can be seen as a sliding block transform, but is usually described equivalently by a filtering with a bank of N filters, followed by subsampling by N; thus, it is often called a *subband decomposition*. Because the filter impulse response is longer than N, samples from neighboring blocks influence the current "transform" coefficients. The expansion operator is thus block Toeplitz, and can be orthogonal or not. Often, cascading of such filter bank structures is used, and a popular, constant relative bandwidth case is referred to as a *wavelet transform*.

- **Overcomplete decompositions.** This is a decomposition in terms of a redundant set of basic functions, leading to an "oversampled" representation in signal processing terms. While this might seem counterproductive since ultimately compression is desired, quantization can now be coarser (for example, zero crossing representation only).

For overviews of transform and subband coding, see Jayant and Noll,[1] Jain,[2] Woods,[3] and Gersho and Gray.[4]

The signal decompositions frequently have an added feature that can be seen as a successive approximation property: certain coefficients contain more of the energy from the original signal and contribute more to the reconstruction quality than others, and, thus, a good approximate reconstruction can be obtained from these coefficients alone. This is sometimes called a *multiresolution feature,* since the approximation corresponds to a lowpass and subsampled version of the original signal. Such a multiresolution feature is useful in a number of applications, ranging from browsing coarse resolution images in a data base to compatible coding where more than one standard format can be obtained from a source coder. For a review of multiresolution techniques in image and video compression, see Vetterli and Uz.[5] While signal decompositions in themselves are well understood, and the mathematics used rely on standard linear algebra, random processes, and linear system theory, the study of decompositions for compression has been very active recently. This is because of both the general interest in data compression and the fact that finding good decompositions for medium to high compression of real signals is a non-trivial problem. The key problems in finding good decompositions for compression are the following:

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1 N. S. Jayant and P. Noll, *Digital Coding of Waveforms,* Englewood Cliffs, NJ: Prentice-Hall, 1984.

2 A. K. Jain, *Fundamentals of Digital Image Processing,* Englewood Cliffs, NJ: Prentice-Hall, 1989.

3 J. W. Woods, *Subband Image Coding,* Boston: Kluwer Academic Publishers, 1991.

4 A. Gersho and R. M. Gray, *Vector Quantization and Signal Compression,* Boston: Kluwer Academic Publishers, 1992.

5 M. Vetterli and K. M. Uz, "Multiresolution Coding Techniques for Digital Video: A Review," *Special Issue on Multidimensional Processing of Video Signals, Multidimensional Systems and Signal Processing,* 3(1992), 161–187.

- **Finding "good" basis functions.** In the high-rate (or low-compression) case, the solution is known as the Karhunen-Loeve transform, but for other cases, conflicting requirements and the interaction with quantization leave open questions.

- **How to construct the basis functions.** Once the criterion to choose the basis function is set, the construction is relatively well understood.

- **Complexity of the implementation.** While the complexity of expansion computation is fairly well understood, especially in the case of floating point implementation, there are open questions related to finite precision realizations.

- **Systems issues.** When there are additional requirements on the decomposition, such as multiresolution or successive approximation, some open problems remain to be solved.

- **Mathematical questions related to signal decompositions.** Interesting mathematical developments, like the theory of wavelets, are related to certain basis constructions. The importance of these results to compression is unclear and currently under investigation.

Most signal expansions used in compression are related to the Karhunen-Loeve transform (KLT). Given a vector process and its correlation matrix, the KLT is the decorrelating transform based on the eigenvectors of the correlation matrix. In the limit of fine quantization, the KLT is also tne best choice of block transform for compression.[6] The KLT has several problems, however. First, it is signal dependent, and thus should be recalculated if the signal characteristics change. Second, it does not have a fast implementation and uses an order N operations per sample. Then, in most compression applications, the size N vector process on which the KLT is calculated is obtained by blocking a signal, but the KLT ignores the remaining correlation across boundaries. Finally, while the KLT is optimal in the fine quantization case,

6 Gersho and Gray, 1991, op. cit.

there is no such result for coarse quantization, and at high compression rates, perceptional criteria dominate.

The first two problems have been addressed by the discrete cosine transform (DCT),[7] which is a good approximation of the KLT of first-order Markov processes with high correlation (a rough but adequate approximation to images). Because of its good performance/complexity trade-off, the DCT has established itself as the preeminent signal decomposition method for medium compression of natural images (JPEG), and is used as part of video coding standards as well (H261, MPEG). As far as block transforms are concerned, the DCT has a leading role, and little research is directed toward finding alternatives for it. However, block transforms tend to be plagued by the "blocking effect," that is, a discontinuity in the reconstructed signal at block boundaries. This is particularly noticeable in audio coding and higher-compression image coding.

The problems associated with the DCT cited earlier have led to the development of overlapping block transforms or subband coding (Croisier et al., 1976; Vetterli, 1984; Princen and Bradley, 1986).[8] Subband coding addresses, at least partly, some of the shortcomings of the DCT. Because of the overlap between blocks and the possibility of cascading several decompositions if needed, correlation across blocks can be better taken into account. Also, the basis functions can be better tuned to perceptional criteria. For example, the blocking effect can be removed, and constant relative bandwidth analysis (obtained by cascading a two-band decomposition on the lowpass channel) allows a better and more natural trade-off between time and frequency analysis than does the DCT.

Once known what basis for decomposition is desired, the problem of designing it is fairly well understood. That is, once the criteria to optimize are set, searching for

7 D. R. Rao and P. Yip, *Discrete Cosine Transform*, San Diego: Academic Press, 1990.

8 P. Cassereau, "A New Class of Optimal Unitary Transforms for Image Processing," SM Thesis, Dept. EECS, MIT, May 1985.

H. S. Malvar and D. H. Staelin, "The LOT: Transform Coding Without Blocking Effects," *IEEE Trans. ASSP, 37*, 4(1989), 553–559.

J. W. Woods and S. D. O'Neil, "Sub-Band Coding of Images," *IEEE Trans. ASSP, 34*, 5(1986), 1278–1288.

J. W. Woods, Ed., *Subband Image Coding*, Boston: Kluwer Academic Press, 1991.

the solution within a class of solutions (for example, unitary transforms, orthogonal filter banks) is a relatively standard optimization problem.[9]

When a compression algorithm is used within a bigger system, considerations other than just pure compression come into play:

- compatibility with other standards;
- possibility of successive approximation and browsing;
- robustness to channel errors, or joint source-channel coding.

Such system questions are less well posed than pure source coding questions, but often a signal decomposition that has several independent channels, where additional channels improve quality, is desired. Typical examples are pyramid, subband, and wavelet coding.[10]

Most recent non-US work has been done in Europe (France, Germany, England, Scandinavia, Belgium, Switzerland), the Pacific Rim (Japan, South Korea, Taiwan, Australia), and Brazil. Researchers in Europe and the Far East publish in the various *IEEE*[11] *Transactions* (on Image Processing, Signal Processing, and Communications), the corresponding *IEE*[12] *Transactions*, and the European *Signal Processing* and *Image Communication* journals. Japanese researchers often publish in the *IEICE*[13] *Transactions. Proceedings* of IEEE Conferences (for example, ICASSP,[14] ISIT[15]) and of EUSIPCO[16] are also good research reference sources.

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9 For design of two- and M-channel filter banks based on complete orthogonal structure, see, P. P. Vaidyanathan and P.-Q. Hoang, "Lattice Structures for Optimal Design and Robust Implementation of Two-Band Perfect Reconstruction QMF Banks," *IEEE Trans. ASSP, 36*, 1(1988), 81–94; and P. P. Vaidyanathan, T. Q. Nguyen, Z. Doganata and T. Saramäki, "Improved Technique for Design of Perfect Reconstruction FIR QMF Banks with Lossless Polyphase Matrices," *IEEE Trans. ASSP, 37*, 7(1989), 1042–1056.

10 M. Vetterli and K. M. Uz, "Multiresolution Coding Techniques for Digital Video: A Review," *Special Issue on Multidimensional Processing of Video Signals, Multidimensional Systems and Signal Processing*, 3(1992), 161–187.

11 Institute of Electrical and Electronics Engineers

12 Institution of Electrical Engineers (United Kingdom)

13 Institute of Electronics, Information, and Communication Engineers

14 International Conference on Acoustics, Speech, and Signal Processing

15 (IEEE) International Symposium on Information Theory

16 European Signal Processing Conference

C. DISCUSSION OF NON-US WORK

1. Good Basis Functions

a. Transforms

Gilge et al. (1989), at the University of Aachen, Germany, present an interesting development on block transforms. In region-based image coding, arbitrary shaped regions have to be compressed, and standard transforms are thus unsuited. The authors develop orthogonal transforms for such regions by starting with a basis and using an orthogonalization procedure. While questions remain as to what would be a "best" transform, this is a step in the right direction.

b. Subband Coding

Research on better subband coding decompositions is active in the United States and abroad. Subband coding originated in Europe with work on medium compression of speech at IBM–La Gaude in France (Croisier et al., 1976). Initial results on perfect reconstruction filter banks (leading to subband coding with no degradation in the absence of lossy compression of the subbands) soon followed in the United States[17] and abroad (Vetterli, 1986).

The design of filter banks adapted to signal statistics (or KLT-like subband coding) has been addressed more recently. Two non-US contributions to this subject are quite interesting. Vandendorpe (1991, 1992), at Université Catholique de Louvain, Belgium, has designed perfect reconstruction filters matched to signal statistics by minimizing the product of subband variances, which leads to the best coding gain. (Similar work is being done in the United States.)[18]

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17 F. Mintzer, "Filters for Distortion-Free Two-Band Multirate Filter Banks," *IEEE Trans. ASSP, 33,* 6(1985), 626–630.

M. J. T. Smith and T. P. Barnwell, "Exact Reconstruction for Tree-Structured Subband Coders," *IEEE Trans. ASSP,* 34, 6(1986), 434–441.

P. P. Vaidyanathan, "Theory and Design of *M*-Channel Maximally Decimated Quadrature Mirror Filters with Arbitrary *M*, Having Perfect Reconstruction Property," *IEEE Trans. ASSP, 35,* 4(1987), 476–492.

18 P. P. Vaidyanathan, *Multirate Systems and Filter Banks,* Englewood Cliffs, NJ: Prentice-Hall, 1993.

Even more original work by Delsarte, Macq, and Slock (1991, 1992) at Philips Laboratories, Brussels, Belgium, carries out adapted subband decomposition by optimizing a filter-bank lattice for each image to be coded, using an iterative algorithm to find the best filter. However, the Philips Laboratory in Brussels has since been closed and the personnel dispersed (Delsarte and Macq to Université Catholique de Louvain, Belgium, and Slock to EURECOM, Sophia-Antipolis, Nice, France).

The above work was focused on two-channel filter banks (lowpass/highpass splitting). On multichannel filter banks, Malvar at the University of Brasilia, Brazil, has produced a continuing stream of interesting papers, starting with the original work on lapped orthogonal transforms lots (which are filter banks subsampled by N and with filters of length 2N) performed in the United States,[19] continuing with modulated lapped orthogonal transforms (LOTs) and LOTs optimized for certain classes of signals, general LOTs (arbitrary filter lengths) and LOTs related to wavelets (Malvar, 1990). A very complete and thorough reference on LOTs is the recent book by Malvar (1992). DeQueiroz and Malvar (1992) show the suboptimality of hierarchical transforms like the wavelet transform.

In Australia, Princen (now with Telecom Research Labs in Melbourne) has produced one of the original LOT papers (Princen and Bradley, 1986).

Filter design for subband coding has always tried to meet several sometimes contradictory requirements. Traditionally, filters were designed so as to be good approximations to lowpass/highpass filters (in the two-band case), or N-th-band filters (multichannel case). But, orthogonality and linear phase (symmetry/antisymmetry) of the filters, which are exclusive in the two-band finite-length filter case, were required as well. While orthogonality (or near orthogonality) is very useful (for example, for bit allocation in compression), the case for linear phase, even for image coding, is not clear. Antonini et al. (1990, 1992—Barlaud's group, Université de Nice, France) argue strongly for it, but the results are not reproducible. These researchers have investigated various subband-/wavelet-based image coding schemes, includ-

19 Malvar and Staelin, 1989, op. cit.

ing lattice vector quantization (VQ) of wavelet decomposed images, and produced some interesting results (Antonini et al., 1992).

An added constraint, called *regularity*, arose with relation to wavelets. A filter is regular if its iterated version tends to a smooth function.[20] This requirement leads to the design of maximally flat filters.[21] The importance of regularity for compression has been much debated, and the first firm results by Rioul (1993a) at CNET,[22] Paris, indicate a slight advantage in a simple coding environment where first-order entropy of subband decomposed images followed by quantization was measured. Rioul just defended his PhD thesis (Rioul, 1993b) at the ENST,[23] Paris, and it is a very fine piece of work spanning theory (regularity estimation of discrete filters), algorithms (fast wavelet transforms and filter design techniques), and applications (influence of regularity on the performance of image coding).

One problem of interest in real-time coding of speech is to design filters with minimum delay, and this problem is addressed in a paper by Takebe et al. (1991) at Kanazawa University, Japan.

While most of the work has focused on finite-length filters, some interesting work on infinite impulse response (IIR) filters, pointing out the efficiency of the scheme and the suitability for image coding (where causality is not a problem) has been done at the Norwegian Institute of Technology, Trondheim, Norway, by Ramstad et al. (Ramstad, 1988; Husøy and Ramstad, 1990). Related work was done in the United States by Smith et al.[24] IIR filters related to subband coding were studied in the context of wavelets by Argenti et al. (1992), and also in the United States.[25]

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20 For a review, see, O. Rioul and M. Vetterli, "Wavelets and Signal Processing," *IEEE Signal Processing Magazine*, *8*, 4(Oct 1991, 14–38.

21 I. Daubechies, "Orthonormal Bases of Compactly Supported Wavelets," *Commun. Pure Appl. Math.*, *XLI*, (1988), 909–996.

22 Centre National d'Etudes de Télélecommunications

23 Ecole Nationale Supérieure de Télécommunications

24 M. J. T. Smith, "IIR Analysis/Synthesis Systems," in J. W. Woods, *Subband Image Coding*, Boston: Kluwer Academic Publishers, 1991, 101–142.

25 C. Herley and M. Vetterli, "Wavelets Generated by IIR Filter Banks," *Proc. IEEE, ICASSP-92*, (1992), 601–604.

Probably the biggest success of subband coding has been the CD-quality audio coding standard MUSICAM (Dehery et al., 1991; Veldhuis et al., 1989) studied by various groups in Europe (CCETT[26] in France, Philips in The Netherlands, and the IRT[27] in Germany). The key has been a design of filter banks matched to the hearing system, that is, that can take advantage of phenomena called masking.[28] While similar research has been done in the United States or in collaboration between the United States and foreign countries, MUSICAM as a system and its application, for example, to digital audio broadcast (DAB) are mostly European efforts.

Masking in subband image coding has been studied in the United States[29] and abroad. In particular, design of filters for subband decompositions with a strong emphasis on the perceptual aspects relevant in image coding has been done in Kunt's group in Lausanne, Switzerland. This group has also studied Gabor-like expansions and their generalization where wavelet-like scale changes are used (Ebrahimi, 1992; Ebrahimi and Kunt, 1991).

Westerink (formerly at the Delft University of Technology, The Netherlands) and Kronander (Linköping University, Sweden) have written good PhD theses analyzing and comparing various subband coding methods for images, with emphasis on bit allocation and quantization in the former case, and filter choices and extension to video in the latter (Westerink, 1989; Westerink et al., 1988; Kronander, 1989).

Chen and Lee (1992) at National Taiwan University give a new design algorithm for filter design in filter banks that seems to give good results at low complexity. However, the filters are not perfect reconstruction because they require linear phase and modulation.

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26 Centre Commun d'Etudes de Telélédiffusion et Télécommunications

27 Institut für Rundfunktechnik, München/Munich

28 J. D. Johnston, "Transform Coding of Audio Signals Using Perceptual Noise Criteria," *IEEE J. Sel. Areas Commun.*, *6*, 2(1988), 314–323.

James D. Johnston and Karlheinz Brandenburg, "Wideband Coding Perceptual Considerations for Speech and Music," *Advances in Speech Signal Processing*, Eds. Sadaoki Furui and M. Mohan Sondhi, New York: Marcel Dekker, Inc., 1992, 109–140.

29 R. J. Safranek and J. D. Johnston, "A Perceptually Tuned Subband Image Coder with Image Dependent Quantization and Post-Quantization Data Compression," *Proc. ICASSP-89*, Glasgow, May 1989, 1945–1948.

Katto and Yasuda (1991) at Tokyo University, Japan, analyze subband coding systems for images with non-orthogonal filters, a case of importance in practice, since linear phase filters cannot be orthogonal. Then, they derive interesting coding gains and bit allocation procedures, and give a unified coding gain, which works also for non-orthogonal transforms.

Ohta et al. (1991, 1992) at NEC, Japan, write a nice review on transform, subband, and wavelet coding, followed by comparisons of scanning orders of coefficients. While SNR performance at a given bit rate is comparable between DCT and wavelet coding, they argue that usually, the latter performs "dramatically better," but not very convincingly.

One of the places where choice of basis functions and quantization do interplay is in the scanning of the transform coefficients and the use of run-length coding and end-of-block (EOB) flagging. For example, the DCT coefficients in JPEG are zig-zag scanned (after quantization), the resulting sequence is run-length coded, and an EOB is used when there are only zeros left. Similar schemes have been developed for wavelet transforms, in particular. Lewis and Knowles (1991, 1992) at Imperial College, London, propose zero-trees, which are the equivalents of EOBs for wavelet coded images. This idea has since been used by others[30] and is a nice contribution.

c. Overcomplete Expansions

The most popular overcomplete expansion is certainly the Burt and Adelson pyramid[31] that has been used for multiresolution coding of images. The oversampling is small, and, thus, it is usually ignored. Park and Lee (1991) at the National University in Seoul, Korea, optimize the filters in the pyramid so as to minimize the difference signal energy. They obtain interesting filter designs and better compression results than most other pyramid image coders. This might also be related to a more sophisticated compression of the difference signal, where they used DCT plus classified vector quantization. Highly overcomplete expansions, such as non-decimated wavelet transforms,[32] where, for example, only local extrema or zero

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30 J. Shapiro, "An Embedded Hierarchical Wavelet Coder," *ICASSP-92*, (1992), 657–660.

31 P. J. Burt and E. H. Adelson, "The Laplacian Pyramid As a Compact Image Code," *IEEE Trans. Com.*, *31*, 4(1983), 532–540.

32 S. Mallat, "Zero-Crossings of a Wavelet Transform," *IEEE Trans. IT*, *37*, 4(1991), 1019–1033.

crossings are kept in the transform domain, have not been considered in the research literature outside the United States.

d. Adaptive Bases

An interesting idea first proposed by Coifman et al. of Yale University[33] is to adapt the basis functions to the signal to be coded, using an appropriate criterion. The Yale work has been done in collaboration with researchers from France (Y. Meyer, CEREMADE[34], Université Paris IX), and thus the idea has spread in Europe, where it was picked up by a few groups; however, thus far it has not lead to results beyond that known in the United States.

2. Construction of Bases

Unitary operators like orthogonal filter banks have parametrizations that go back to classical circuit theory of the 1960s, when there was a strong European school (Belevitch, 1968), which has continued with, for example, Fettweiss (1971) at the Ruhr University, Bochum, Germany. Thus, some good work on structures for filter banks, typically using wave digital filters, is done, for instance, in Germany and Poland (Bleja and Domanski, 1990; Domanski, 1991). Work on filter banks with coefficients having limited precision, a subject that has been studied a lot for single filters, but not much for filter banks, has also been performed (Ebrahimi, 1992). Excellent work on wavelet filter design, where regularity and frequency selectivity can be traded-off one against the other, has been done by Rioul (1993a) at CNET, Paris. His work on regularity testing of discrete-time filters is also to be noted for its quality and clarity (Rioul, 1992a-b).

33 R. R. Coifman, Y. Meyer, S. Quake, and M. V. Wickerhauser "Signal Processing and Compression with Wavelet Packets," Dept. of Math, Yale Univ., Preprint, 1990.

34 Centre de Recherches de Mathématiques de la Décision

3. Complexity of Implementation

a. Transform Computation

There has been an ongoing stream of papers on the fast computation of the DCT, in one or more dimensions, both in the United States and abroad. Despite this large volume, contribution is generally limited. Notable exceptions are contributions by Duhamel at CNET, Paris—work on relating the DCT to convolution (Duhamel and H'Mida, 1987) and on multidimensional transforms (Duhamel and Guillemot, 1990)—and Arai et al. (1988) in Japan—work on a new factorization of DCT matrix that permits scaling to be removed, thus lowering the computational complexity by 0(N), which is non-negligible for the small transform sizes common in image compression.

b. Filter Bank Computations

Limited progress in filter bank computations has been made recently. Historically, the major advance was Bellanger's algorithm for modulated filter banks (Bellanger and Daguet, 1974). Work on LOTs and their fast algorithms has been done by Malvar (1992) at the University of Brasilia, Brazil, and others like Mau (1991) at CCETT, France, who presents a good overview of the trade-offs involved. Optimizing the fast transform involved in modulated filter banks has been done by Duhamel et al. (1991).

c. Tree Structures and Wavelet Transforms

An excellent overview of the complexity of the wavelet transform, and state-of-the-art algorithms was written by Rioul and Duhamel (1992). The famous "a trous" algorithm, developed by the Marseilles group (Grossman et al., 1992; Rioul and Duhamel, 1992) is a relatively standard result of multirate signal processing.

d. Finite Word Length Implementation

Two sources of error appear in implementations: namely, coefficient quantization and computation quantization. The former is well understood, and several structures exist that will preserve certain properties (like orthogonality) even if the coefficients are quantized.[35] Some of this work, such as the wave digital filters of Fettweiss (1971), comes from Europe. Such wave digital filters are used in filter bank work conducted in Europe (especially in Prof. Fettweiss' group in Germany) aimed at implementations (Domanski, 1991).

Computation quantization is usually modeled using simple additive white-noise sources. This might not be sufficient for high-precision computation and perfect inversion (like for lossless coding). The problem is of sufficient interest that the IEEE Circuit and Systems Society established a committee to study the quantization and inversion of the DCT, mainly because of the multiple encoding/decoding problem. The solution is usually brute-force (just use enough bits). Bruekers and van den Enden (1992) at Philips Laboratories, The Netherlands, show how to control quantization after multiplications using so-called "perfect inversion networks." As this is a major source of errors in fixed-point implementations, this is an interesting result. However, the approximation properties of such structures have yet to be understood.

Another approach taken for the implementation of the DCT is the Integer Cosine Transform (ICT) developed by Cham and colleagues in Hong Kong. The ICT is constructed by requiring its basis to have the same shape as the DCT basis, and by requiring the basis vectors to be orthogonal (Cham, 1989; Cham and Chan, 1991). Except for these two conditions, the construction is generally *ad hoc* and often involves computer searches. This lack of theory is the greatest shortcoming of the ICT. Not only does it prevent the extension of the technique to transforms of arbitrary size (so far, ICTs have been developed for sizes 8 and 16 only), it also makes development of fast algorithms difficult. Nonetheless, ICTs have been implemented with some success (Choi et al., 1988).

4. Systems Issues

Work in Europe has focused on compatible coding of digital television, where full resolution is high-definition television (HDTV), and subresolution is regular television. Typically, filter bank decompositions are used, and lead to interesting

35 See, P. P. Vaidyanathan, *Multirate Systems and Filter Banks*, Englewood Cliffs, NJ: Prentice-Hall, 1993.

results; see, for example, the work of Mau (1991) at CCETT, France; Biemond et al. (Bosveld et al., 1992) in Delft, The Netherlands; and Pécot et al. (1990a-b; Tourtier et al., 1992) at Thomson, in France. The suboptimality of such schemes, which have been studied in a theoretical context,[36] is not quantified in this practical work.

Robust coding for digital broadcast, which was studied in the United States,[37] has been further investigated in Europe, where it is pursued with some interest (de Bot, 1992; Amor et al., 1992).

5. Mathematical Questions Related to Signal Decompositions Used in Compression

Some interesting connections between subband coding and lapped orthogonal transforms and expansions of continuous-time functions have appeared recently.

A major development was the theory of wavelets, pioneered in France in the early 1980s by a group in Marseilles centered around Morlet and Grossman (Goupillaud et al., 1984), who started the field, and a group in Paris centered around Meyer (1990), who formalized and advanced the subject, and established connections with signal processing (Cohen, 1990).

Pioneering work was also performed in the United States, often in connection with the French groups, by Mallat on multiresolution analysis,[38] by Daubechies on construction of wavelets based on discrete-time filter banks,[39] and by Coifman and his group on wavelet packets (related to subband coding trees) and local cosine

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36 W. H. R. Equitz, "Successive Refinement of Information," PhD Thesis, Stanford Univ., 1989.

37 W. F. Schreiber, "Considerations in the Design of HDTV Systems for Terrestrial Broadcasting," *Electronic Imaging '90, Boston, 30 Oct 1990.*

K. Ramchandran, A. Ortega, K. M. Uz, and M. Vetterli, "Multiresolution Broadcast for Digital HDTV Using Joint Source-Channel Coding," to appear in *IEEE JSAC*, 1993.

38 S. Mallat, "A Theory for Multiresolution Signal Decomposition: The Wavelet Representation," *IEEE Trans. PAMI, 11*, 7(1989), 674–693.

S. Mallat, "Multiresolution Approximations and Wavelet Orthonormal Bases in $L^2(R)$," *Trans. AMS, 315*, 1(Sept 1989), 69–87.

39 Daubechies, 1988, op. cit.

I. Daubechies, *Ten Lectures on Wavelets*, SIAM, 1992.

transforms (related to LOTs).[40] Coifman's efforts have been conducted in collaboration with Meyer and his group in Paris. Excellent work along these lines is done in Europe, particularly in France. For example, Rioul (1992a-b) at CNET, Paris, has studied the behavior of iterated filter banks in detail. Work on wavelets with rational dilation factors has been done by Auscher (1991) at Université Paris IX.

For a good overview of the work going on in Europe, see the *Proceedings of the Second Marseilles Conference on Wavelets* (Combes et al., 1989). A follow-up conference was held in 1992 in Toulouse (France), but the *Proceedings* have not yet appeared.

This subject is quite popular in the United States,[41] where further work on the connection with filter banks has been done.

Wavelets have also become popular in Japan (Ohta et al., 1992), but with less theoretical work than in Europe or the United States.

6. Subband Coding of Video

Three-dimensional subband coding[42] was proposed as a simple source coding technique for video. It has limited compression performance because of the absence of good motion handling. Alternatives for subband-like video compression have been considered by various researchers in the United States[43] and abroad (Westerink et al., 1990).

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40 R. R. Coifman, Y. Meyer, S. Quake, and M. V. Wickerhauser, "Signal Processing and Compression with Wavelet Packets," Dept. of Math, Yale Univ., Preprint, 1990.

41 For books and edited volumes on the subject, see, C. K. Chui, *An Introduction to Wavelets*, Academic Press, 1992.

C. K. Chui, Ed., *Wavelets and Applications*, Academic Press, 1992.

M. B. Ruskai et al., Eds., *Wavelets and Their Applications*, Boston: Jones and Bartlett, 1992.

42 G. Karlsson and M. Vetterli, "Three Dimensional Subband Coding of Video," *Proc. IEEE ICASSP*, Apr 1988, 1100–1103.

43 J. W. Woods and T. Naveen, "Subband Encoding of Video Sequences," *Proc. SPIE Conf. Visual Communications & Image Processing*, Nov 1989, 724–732.

H. Gharavi, "Subband Coding of Video Signals," *Subband Image Coding*, Ed. J. W. Woods, Norwell, Mass: Kluwer Academic Publishers, 1990.

TABLE V.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—SIGNAL DECOMPOSITIONS

Researcher	Affiliation	Area of Expertise
	Belgium	
Benoit Macq L. Vandendorpe	Université Catholique de Louvain, Louvain	Subband image coding
Phillipe Delsarte	Phillips Research Laboratory, Brussels (closed in 1991)	
	France	
Y. Meyer	CEREMADE, Université Paris IX, Paris	Wavelet theory
J. Mau	CCETT, Rennes	Subband coding of television
P. Duhamel O. Rioul	CNET, Paris	Fast algorithms Wavelets
A. Grossman J. Morlet P. Tchamitchian B. Torresani	Université de Marseilles, Marseilles	Wavelet theory
P. J. Tourtier	Thomson, Rennes	Subband coding of television
M. Barlaud	Université de Nice, Nice	Wavelet/subband image coding
	Germany	
H. Mussman	Hanover University, Hanover	Subband image coding
R. Schäfer	Heinrich Hertz Institute, Berlin	Image and video coding and transmission
	Japan	
T. Nishitani	NEC Research Laboratories, Tokyo	Wavelet coding of images and video
Naohisa Ohta H. Watanabe	NTT Research Laboratories, Tokyo	Hierarchial coding, LOTs
YasuhikoYasuda	Tokyo University, Tokyo	Subband image coding

Three-dimensional pyramid coding[44] using motion interpolation gives good performance, but is fairly complex. A simple alternative that has low complexity is non-motion-compensated video pyramids. In the United Kingdom, Yates and Ivey (1991) report good results at two orders of magnitude of complexity reduction over standard coders (H.261, MPEG), but the results seem preliminary as far as video quality is concerned.

An alternative is to replace the DCT in a standard motion-compensated video coder by subband coding or wavelet coding. Lewis and Knowles (1991) at Imperial College, London, report good results (better than H.261) using a two-dimensional wavelet transform and simple frame difference (instead of motion-compensated frame difference). Their claim that quality is better than H.261 (but without motion compensation) is hard to believe.

D. PROJECTIONS FOR THE FUTURE

No major changes in signal decomposition research are expected in the near future. Good theoretical work will continue, for instance, in France. Good applications, especially related to complete systems, will be done, for example, in Germany and The Netherlands. The Far East, while producing probably less theoretical work, will lead in implementations. The United States will remain in a good position as far as new ideas and theory are concerned, as well as high-end applications.

E. KEY NON-US RESEARCH PERSONNEL AND FACILITIES

Table V.I lists the key non-US research personnel active in signal decomposition research, their affiliations, and the signal-decomposition-related areas in which they have been working.

44 K. M. Uz, M. Vetterli, and D. LeGall, "Interpolative Multiresolution Coding of Advanced Television with Compatible Subchannels," *IEEE Trans. CAS for Video Technol.*, *1*, 1(Mar 1991), 86–99.

TABLE V.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—
SIGNAL DECOMPOSITIONS (cont'd.)

Researcher	Affiliation	Area of Expertise
	Norway	
T. A. Ramstad	Norwegian Institute of Technology, Trondheim	Subband coding, filter banks
	South Korea	
Sang Uk Lee Seop Hyeong Park	National University, Seoul	Pyramid coding of images
	Sweden	
T. Kronander	Linköping University, Linköping	Subband image coding
	Switzerland	
Murat Kunt	Ecole Polytechnique Fédérale de Lausanne (EPFL), Lausanne	Image and image sequence coding
	Taiwan	
J. H. Lee S. C. Pei	National Taiwan University, Taipei	Filter design
	The Netherlands	
J. Biemond	Delft University of Technology, Delft	Subband image coding
Various researchers	Philips Labs, Eindhoven	Subband coding, systems work
	United Kingdom	
G. Knowles A. S. Lewis	Imperial College, London	Wavelet image and video compression

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CHAPTER V: SIGNAL DECOMPOSITIONS
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Antonini, M., M. Barlaud, P. Mathieu, and I. Daubechies, "Image Coding Using Vector Quantization in the Wavelet Transform Domain," *Proc. IEEE ICASSP, Albuquerque, NM*, Apr 1990, 2297–2300.

Antonini, Marc, Michel Barlaud, Pierre Mathieu, and Ingrid Daubechies, "Image Coding Using Wavelet Transform," *IEEE Trans. Image Process.*, *1*, 2(Apr 1992), 205–220.

Arai, Y., T. Agni, and M. Nakajima, "A Fast DCT-SQ Scheme for Images," *Trans. IEICE, E-71*, 11(1988).

Argenti, F., G. Benelli, and A. Sciorpes, "IIR Implementation of Wavelet Decomposition for Digital Signal Analysis," *Electron. Lett.*, *28*, 5(1992), 513–514.

Auscher, P., "Wavelet Bases for $L^2(R)$ with Rational Dilation Factor," *Wavelets and Their Applications*, Eds. G. Beylkin et al., Boston: Jones and Bartlett, 1991.

Bellanger, M. G., and J. L. Daguet, "TDM-FDM Transmultiplexer: Digital Polyphase and FFT," *IEEE Trans. Commun.*, *22*, 9(1974), 1199–1204.

Belevitch, V., *Classical Network Synthesis*, San Francisco: Holden Day, 1968.

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CHAPTER VI
SPEECH AND AUDIO CODING

A. SUMMARY

Outside the United States, there is substantial ongoing engineering research and development in speech coding, much of it motivated by the move to digital transmission for cellular mobile systems and other wireless applications. Very good work in the development and testing of speech coding algorithms and implementations is taking place in geographically dispersed locations, particularly in Western Europe, Japan, and Canada. Relatively little of this work is of a fundamental and innovative nature compared to the dominance of innovative algorithmic approaches originating in the United States. Much of the non-US work consists of a large variety of complexity reduction methods and minor algorithm modifications, with slight quality improvements over the more basic versions of the algorithms previously developed in the United States. A few very innovative and imaginative coding algorithms or components of coding algorithms have appeared in the non-US literature in recent years, but none have had a major impact on the current state of the art in speech coding. In fact, there is a major gap between academic and industrial research in most countries. The primary non-US work in speech coding is being done in Japan, France, the United Kingdom, Italy, Germany, Canada, and Norway. Smaller efforts can be found in The Netherlands, Spain, Sweden, Denmark, Belgium, South Korea, Australia, Singapore, Ireland, Hong Kong, Brazil, Australia, China, Taiwan, Portugal, Turkey, Greece, India, and elsewhere.

In contrast, most of the important work in wideband audio coding has come largely from outside the United States, primarily from Western Europe, particularly Germany, The Netherlands, and France. Standardization of speech (and audio) coding algorithms is the driving force for a large fraction of the activity in speech and audio coding.

B. INTRODUCTION

This chapter focuses on research and development (R&D) activities for digital compression of speech and audio signals. The word "coding" in this area is usually used as being synonymous with compression and does not refer to the use of error-correcting codes. Motivating much of the work in speech and audio coding are various international and national standardization efforts. Standards are essential for compatibility of terminals in voice and audio communication systems.

Speech coding algorithms can be divided into two main categories: *waveform coders* and *vocoders*. They differ according to the way in which the speech signal is represented. In waveform coders, the data transmitted from encoder to decoder provide a representation of the original speech waveform so that the reproduced signal approximates the original waveform and, consequently, provides an approximate recreation of the original sounds. In vocoders (historically a contraction of "voice coders"), no attempt is made to represent the waveform but, instead, parameters that characterize individual sound segments are specified and transmitted to the decoder, which then reconstructs a new waveform that will have a similar sound. Generally, these parameters either (1) characterize the short-term spectral properties of a sound that are important to the human auditory system, or (2) they specify the parameters of a model of the human vocal tract so that the decoder can synthesize the speech sound by modeling the human speech production mechanism for the sound. The decoder synthesizes a speech sound from these parameters. Vocoders operate at much lower bit rates than waveform coders, but the reproduced speech quality, while intelligible, usually suffers from a loss of naturalness and often some of the unique characteristics of an individual speaker are lost.

Much of the work in speech coding is dominated by a handful of different algorithmic approaches, and most of the developments in recent years have focused on modifications and enhancements of these generic methods. Most notable and most popular for speech coding is *code-excited linear prediction* (CELP). Other methods in commercial use today that continue to receive attention include *adaptive predictive coding* (APC), *multipulse linear predictive coding* (MP-LPC), *regular pulse excitation linear predictive coding* (RPE-LPC), *adaptive differential pulse code modulation* (ADPCM), *linear predictive coding* (LPC) vocoders, *multiband excitation* (MBE) *coding*, and *sinusoidal transform coding* (STC). All of the above-mentioned algorithms except the last three, LPC, MBE, and STC, are waveform coding algorithms. MP-LPC, RPE-LPC, and CELP are sometimes viewed as hybrid algorithms having some features of vocoders, but they basically belong to the class of waveform coders.

Most coders with other names are closely related to those listed here. Of diminishing interest are RPE-LPC, MP-LPC, and ADPCM although all three have become standardized for specific application areas. Of limited but increasing interest is MBE, which has overtaken CELP in one standards competition. Organizations involved in speech coding standardization include the International Consultative Committee on Telephone and Telegraphy (CCITT), (North American) Telephone Industry Association (TIA), Japanese Digital Cellular (JDC) organization, Groupe Speciale Mobile (GSM), International Marine Satellite organization (INMARSAT), and the Airlines Electronic Engineering Committee (AEEC) administered by Aeronautical Radio, Inc. (ARINC). The CCITT has recently changed its name to Telecommunications Standards Sector (TSS).

Most of the work today in speech coding is based on a very small number of basic algorithmic approaches, and, in virtually each case, the fundamental approach originated in the United States. Nevertheless, many good quality enhancements and modifications have been emerging in Western Europe, Japan, and Canada. Sporadic isolated bits of activity may be found in many other countries, but generally these efforts are isolated university research projects as opposed to seriously funded R&D motivated by national or international standardization. A few countries, such as Brazil, actively participate in international standards, although there appears to be little or no research activity in these countries. The major non-US participants in CCITT and other international standardization activities are Japan, Canada, the United Kingdom, France, Italy, Germany, Norway, and, to some extent, The Netherlands. Generally, these efforts are undertaken in industrial or government R&D laboratories rather than universities. Countries where almost all speech coding activities are in industry, with very little university-based research, include Japan, France, and Italy. In some countries, such as Hong Kong, Taiwan, Turkey, India, Singapore, and Greece, virtually all work in speech coding is done at universities.

The ease of real-time implementation of speech coding algorithms with single-chip digital signal processor (DSP) chips has led to widespread implementations of speech algorithms in the laboratory, as well as an expansion of applications to communication and voice storage systems. The most enormous potential market for speech coding is in the emerging personal communication systems (PCS) area, where volumes of hundreds of millions are expected in the United States alone, with a comparable number in Western Europe and Japan in the next decade, as all telephones become wireless. With so many applications already emerging or expected to emerge in the next few years, it is not surprising that speech coding has become such an active field of research.

Wideband audio coding for high-fidelity reproduction of voice and music has emerged as an important activity. Most audio coding algorithms today are based on transform or subband coding approaches and involve sophisticated perceptual masking models for dynamically allocating bits to different frequency bands to yield a reproduced quality nearly indistinguishable from that of compact-disc (CD) audio quality. Applications of audio coding lie largely with the broadcasting industry, motion picture industry, and consumer audio and multimedia products. A key international standard developed by the Motion Picture Experts Group (MPEG) includes an audio coding algorithm. Much of the leading work in audio coding has been done outside the United States, in Germany, The Netherlands, and France.

C. DISCUSSION OF NON-US WORK

1. Code-Excited Linear Prediction

The dominant coding method studied and implemented today worldwide is *code-exited linear prediction* (CELP) coding also known as *stochastic coding* or *vector excitation coding* (VXC). The invention of CELP is generally attributed to Atal and Schroeder.[1] A similar conceptual approach to CELP was also introduced independently by Copperi and Sereno (1984), and various other antecedents to the key CELP approach, analysis-by-synthesis coding with vector quantization, have been studied by US researchers since the late 1970s. Initially viewed as an algorithm of extraordinary complexity, CELP served only as an existence proof (with the help of supercomputers) that it is possible to get very high speech quality at bit rates far below what was previously considered feasible. In 1986, soon after its introduction, reduced complexity methods for implementation of the basic CELP algorithm were

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1 B. S. Atal and M. Schroeder, "Stochastic Coding of Speech Signals at Very Low Bit Rates," *Proc. Int'l. Conf. Commun.*, May 1984, 1610–1613.

M. R. Schroeder and B. S. Atal, "Code-Excited Linear Prediction (CELP) High Quality Speech at Very Low Bit Rates," *Proc. Int'l. Conf. Acoust. Speech. Signal Process.*, Mar 1985, 937–940.

One widely used technique for reducing search complexity is the use of sparse excitation codebooks where most of the code vector elements have value zero. The basic idea, introduced in the United States,[6] led to many variations and improvements. In particular, Xydeas et al. (1988) proposed the use of ternary codebooks where the randomly located nonzero entries are forced to be +1 or -1. Salami (1989), in the United Kingdom, proposed the use of fixed regularly spaced positions for the nonzero entries so that a short binary word can directly specify the nonzero polarities, eliminating the need for a stored excitation codebook. This technique, called BCELP (for binary CELP), reduces complexity and sensitivity to channel errors while reportedly maintaining good quality. An improved version of BCELP was later reported in Belgium by Boite et al. (1990) to have slightly better quality while further reducing complexity.

Kondoz and Evans have introduced a novel method for excitation coding called CELP-BB *(CELP with base-band coding)*, where the LPC residual after low pass filtering and downsampling is CELP-coded using only long-term (pitch) synthesis (1988; Kondoz et al., 1990; Atungsiri et al., 1990). The resulting algorithm achieves a major reduction in complexity while reportedly giving comparable quality to ordinary CELP coding. Bit-rates ranging from 9.6 down to 2.4 kb/s were tested.

Moulsley and Elliott (1991a-b), in the United Kingdom, introduced a simple orthogonalization technique to partition the short-term filtered excitation codebook into four subcodebooks, each having entries that are orthogonal to those of the other three codebooks. The perceptually weighted input speech is similarly partitioned into four orthogonal components, and four separate but reduced size searches are performed.

Kipper et al. (1990) reported a *multigrid* CELP technique that combines features of CELP, MP-LPC, and RPE-LPC using a set of codebooks, each containing shifted versions of regular nonzero pulse positions with a particular spacing. They claim superior speech quality is obtained, compared to the use of stochastic excitations. Akamine and Miseki (1990), in Japan, introduced an *adaptive-density-pulse* (ADP) excitation method, in which the excitation vectors are pulse trains with nonzero

6 Davidson and Gersho, 1986, op. cit.

introduced by Trancosa and Atal,[2] by Davidson and Gersho,[3] and by Hernandez-Gomez et al. (1986), and it became recognized that CELP was a viable method that could be implemented in real-time. Since 1986, the number of papers on CELP and its applications seems to have grown exponentially. Numerous variations, extensions, improvements, complexity reductions, and implementations have been reported. The most significant developments came from the United States with the development and adoption of Federal Standard 1016, a CELP algorithm at 4.8 kb/s for secure voice transmission with various refinements by Campbell et al.[4] and the development of *vector-sum-excited linear prediction* (VSELP) by Gerson and Jasiuk,[5] which has been adopted as a standard for the North American digital cellular telephone at 8 kb/s (not including error correction) and at 6.3 kb/s for the Japanese Digital Cellular standard. Currently, the GSM is establishing a standard for half-rate digital cellular systems in Europe, and the two remaining candidates for the speech coding component are CELP algorithms.

Most of the many advances to CELP coding are oriented toward reducing complexity, increasing robustness to channel errors, or improving quality. Much of the effort is oriented toward improving the excitation signal while controlling or reducing the excitation search complexity. Some efforts have been made to improve the modeling of the short-term LPC synthesis filter or the quantization of the LPC parameters.

Akamine and Miseki (1989; Miseki and Akamine, 1991), in Japan, and Flanagan et al. (1991), in Ireland, each introduced a different technique to implement a pole-zero synthesis filter that can handle nasal phonemes more effectively and improve the modeling of the high-frequency range. Both studies reported some useful improvements in speech quality.

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2 I. M. Trancoso and B. S. Atal, "Efficient Procedures for Finding the Optimum Innovation in Stochastic Coders," *Proc. Int'l. Conf. Acoust. Speech. Signal Process., Tokyo*, 1986, 2379–2382.

3 G. Davidson and A. Gersho, "Complexity Reduction Methods for Vector Excitation Coding," *Proc. I Int'l. Conf. Acoust. Speech. Signal. Process., Tokyo*, 1986, 3055–2058.

4 J. P. Campbell, Jr., T. E. Tremain, and V. C. Welch, "The DoD 4.8 KBPS Standard (Proposed Feder Standard 1016)," *Advances in Speech Coding*, Eds. B. S. Atal, V. Cuperman, and A. Gersho, Kluwer Academic Publishers, 1991, 121–133.

5 I. A. Gerson and M. A. Jasiuk, "Vector Sum Excited Linear Prediction (VSELP)," *Advances in Speech Coding*, Eds. B. S. Atal, V. Cuperman, and A. Gersho, Kluwer Academic Publishers, 1991, 69-79.

pulses at regular intervals, and different pulse densities can be used for different subframes. They claim enhanced speech quality with reduced complexity.

LeGuyader et al. (1988) use binary valued code vectors of unit magnitude so that binary words directly map into excitations without codebook storage, but the excitation alone requires 8-kb/s coding so that its use is limited to high-rate applications. The idea is similar to, but less versatile than, BB-CELP described earlier.

Another approach to improving CELP performance is to jointly optimize the stochastic and adaptive codebook searches and associated gain values. This was studied in Germany by Muller (1989), who showed that for the same speech quality, the use of joint optimization allows the excitation codebook size to be reduced so that up to 600 b/s can be saved.

Lee and Un (1989), in South Korea, proposed a multistage version of the self-excited coder (from Rose and Barnwell),[7] where no excitation codebook is used, but a new excitation segment is obtained from delayed replicas of the past excitation (after an initialization).

The use of lattices in multiple dimensions for excitation codebooks has been extensively studied by J. P. Adoul, in Canada, and others. This is another way to avoid the need for storing a codebook, since lattices are readily generated and mappings between lattice points (code vectors) and binary words are known. The authoritative work of Conway and Sloane in the United States has provided an encyclopedic source of information on lattices that serves as the basis for its application to CELP. See, in particular, the work by Lamblin et al. (1989) and DiFrancesco (1992).

Codebook design algorithms for CELP were systematically evaluated by LeBlanc and Mahmoud (1990) in Canada.

These are just examples of a vast amount of literature on CELP speech coding. It is generally difficult to assess the quality of individual contributions in speech

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7 Richard C. Rose and Thomas P. Barnwell, III, "The Self-Excited Vocoder—An Alternative Approach to Toll Quality at 4.8 kb/s," *Proc. IEEE Int'l. Conf. Acoust., Speech, Sig. Process., 1, Tokyo, Apr 1986*, 453–456.

coding, since reported signal-to-noise ratio (SNR) improvements do not necessarily indicate perceptual quality improvements, and self-reported quality assessments of researchers is at best unreliable. Additional examples of contributions to CELP coding from France, Italy, and Japan are research reported by Le Guyader et al. (1989); Moreau and Dymarski (1989); Dymarski et al. (1990); Galand et al. (1990, 1992); Delacovo and Sereno (1990); Andreotti et al. (1991); Copperi (1991); Akamine and Miseki (1990b); Tanaka et al. (1990); Taniguchi et al. (1991a–b); Johnson and Taniguchi (1990, 1991); Mano and Moriya (1991); and Moriya (1992).

2. Multipulse and Regular Pulse Excitation LPC Coding

Prior to the widespread success of CELP, *multipulse LPC* (MP-LPC) was viewed as the most promising algorithm for speech coding in the range of 8 to 16 kb/s. *Regular pulse excitation LPC* coding, or RPE-LPC, was introduced by Kroon et al. (1986) in The Netherlands. RPE-LPC uses regularly spaced pulse patterns for the excitation, with the position of the first pulse and the pulse amplitudes determined in the encoding process. Although inspired by MP-LPC, in some sense it is closer in spirit to CELP. A modified version of RPE-LPC with long-term prediction was subsequently selected as the basis of a speech coding standard by the European mobile radio community, Groupe Speciale Mobile (GSM) and was largely developed by groups in Germany and France. All of these earlier algorithms are gradually fading away as candidates for future products or standards, since the superior performance capability of CELP for bit rates ranging from 4.8 to 16 kb/s has generally become recognized.

3. Frequency Domain Coding

Frequency domain coding refers to a family of coding techniques in which emphasis is given to representing and reproducing the spectral character of the speech signal rather than the time domain waveform. Subband and transform coding methods fall into this category and were extensively studied a decade ago. They continue to be of major importance for image, video, and wideband audio coding, but they are generally not regarded today as competitive techniques for speech coding. Some isolated studies of subband and transform coding of speech still continue, but generally these methods are viewed as inferior and less promising than CELP coding. One CCITT standard for wideband (7 kHz) speech at 64 kb/s uses a two-band subband coder (Taka et al., 1986; Mermelstein, 1988).

Another class of frequency domain coding methods has recently emerged; these methods are viewed as a viable alternative to CELP, particularly for rates of 4 kb/s and below. The general focus of these coders is to characterize the evolving short-term spectra of speech by extracting and quantizing parameters that specify the spectra, giving particular attention to the pitch harmonics present in voiced speech. Three approaches have been studied: harmonic coding, sinusoidal transform coding, and multiband excitation coding. Perhaps, the first conceptual introduction of this approach was by Hedelin (1981). Later, Almeida and Tribolet developed harmonic coding of speech[8] and, in subsequent papers, reported very high quality at 6 to 8 kb/s. As with CELP, the original work was done in the United States, although Tribolet is Portuguese and subsequently returned to Portugal. More recently, Marques, Almeida, and Tribolet studied critical issues needed to achieve high quality with harmonic coding at lower rates, and they presented a 4.8-kb/s version of the algorithm (Marques et al., 1990).

A frequency domain coding technique related to harmonic coding, called *sinusoidal transform coding* (STC) was developed and extensively refined by McCauley and Quatieri.[9] More recently, another approach to frequency domain coding with sinusoidal synthesis of the speech called *multiband excitation coding* (MBE) was developed by Jae Lim and his students at MIT.[10] An *improved* version, called IMBE, was subsequently adopted by INMARSAT as a standard for satellite voice communications, competing favorably over CELP and other algorithms. Both STC and MBE identify spectral peaks in each successive frame of speech and encode and transmit the amplitude (and, in some cases, the phase) of these peaks. The receiver synthesizes speech with very similar time-varying spectra by controlling the magnitude and phase of a set of sine waves.

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8 L. B. Almeida and J. M. Tribolet, "Harmonic Coding: A Low Bit-Rate Good-Quality Speech Coding Technique," *Proc. Int'l. Conf. Acoust., Speech, Sig. Process., Paris*, 1982, 1664–1667.

9 R. J. McAulay and T. F. Quatieri, "Speech Analysis/Synthesis Based on a Sinusoidal Representation," *IEEE Trans. Acoust., Speech, Sig. Process.*, *34*, (1986), 744–754.

10 D. W. Griffin and J. S. Lim, "Multiband Excitation Vocoder," *IEEE Trans. ASSP*, *36*, 8(1988,), 1223–1235.

Although MBE originated in the United States, further developments of the approach have come from Canada and the United Kingdom. Specifically, Yeldener et al. (1991, 1992a–b), at the University of Surrey in the United Kingdom, have shown that the bit rate may be substantially reduced by replacing the spectral modeling with an LPC modeling technique. They report significantly superior quality at 2.4 kb/s compared to the (relatively ancient) US government standard LPC vocoder algorithm, LPC-10. A different approach was taken by Hassanein et al. (1992), in Canada, where the MBE analysis is followed by a postprocessor that selects three fixed-bandwidth windows and sends spectral information only for these regions. They report comparable quality with full-band MBE for noise-free speech. An interesting comparison between CELP and sinusoidal coding by Trancoso et al. (1990), in Portugal, suggest that these techniques are complementary and future work might lead to some merging of these two approaches.

Recently, Sony and Mitsubishi have developed speech coders based on MBE as candidates for the Japanese Digital Cellular (JDC) half-rate standard. Sony's coder is described by Nishiguchi et al. (1993). The total amount of work on sinusoidal coders has been very little compared to the work on CELP, but indications are that this is a promising approach and will lead to increased studies. In fact, there is some indication that in the next few years MBE will either overtake CELP for 2.4-kb/s coding, or some hybrid coding scheme or an entirely new approach will replace both. These coders still retain some vocoder type imperfections in reproduced speech but generally give cleaner, crisper reproduction than is available with CELP coders at comparable rates (2.4 to 4.8 kb/s).

4. Low-Delay Speech Coding

For many applications, the time delay introduced by speech coding into the communications link is a critical factor in overall system performance. While ADPCM introduces negligible delay, most contemporary coding algorithms, such as CELP, must buffer a large chunk of input speech samples prior to further signal processing. This buffering delay, plus delay due to computation, plus the delay in the decoder in unpacking and interpreting the data bits can lead to one-way end-to-end coder delays of 60 to 100 ms, and occasionally even higher values. In 1987, the CCITT established a maximum delay requirement of 5 ms for a 16-kb/s standard algorithm, with a desired objective of only 2 ms. This culminated in the adoption of the LD-CELP algorithm as CCITT standard G.728 in 1992. Currently, the CCITT is working on the adoption of a moderately low-delay algorithm at 8 kb/s, with a total coder delay of 32 ms (Hayashi, 1992).

Important contributions to low-delay speech coding have originated from Canada and Japan, although the 16-kb/s LD-CELP came from AT&T Bell Labs in the United States.[11] Other contributions to low-delay coding have come from the United Kingdom and Italy.

Several important ideas for modifying the CELP algorithm to achieve very low coding delay with high quality for both 16 and 8 kb/s rates are due to the work of Prof. V. Cuperman and students at Simon Fraser University, Canada, in a sequence of papers from 1988 to 1992. The key idea of backward adaptation of the LPC synthesis filter was reported by Watts and Cuperman (1988). Backward adaptation of the pitch predictor was introduced by Pettigrew and Cuperman (1989). The use of lattice predictors for backward adaptation of the LPC synthesis filter was studied by Peng and Cuperman (1991). A high-quality low-delay tree coder was reported by Iyengar and Kabal (1988) in Canada. Prof. P. Kabal of McGill University is also an active and highly regarded Canadian researcher in speech coding. Finally, another very active speech coding researcher, J. P. Adoul at the University of Sherbrooke in Canada, and his colleagues have recently developed a low-delay 8-kb/s coder, which, in cooperation with the Centre Nationale d'Etudes des Telecommunications (CNET) in France, was submitted to the CCITT as a candidate for the 8-kb/s standardization (Lefebvre et al., 1993).

A technique for combining forward and backward pitch prediction for low-delay coding at 8 kb/s was introduced by Kataoka and Moriyaat (1991), at NTT in Japan. Moriya spent a year as a visiting researcher at AT&T Bell Labs (roughly in 1988-89) at the time when the LD-CELP coder was being developed there by J. H. Chen.[12] More recently, a very high-quality low-delay coder was reported by Kataoka et al. (1993), and this is a candidate for the CCITT 8-kb/s standardization program. Their method makes use of conjugate vector quantization (VQ) introduced earlier by

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11 J.-H. Chen, R. V. Cox, Y.-C. Lin, N. Jayant, and M. J. Melchner, "A Low-Delay CELP Coder for the CCITT 16-kb/s Speech Coding Standard," *IEEE J. Sel. Areas Commun.*, *10*, (Jun 1992), 830–849.

12 Ibid.

Moriya (1992). T. Moriya and colleagues at NTT have made several good-quality contributions to speech coding. Another study of 8-kb/s low-delay CELP motivated by the CCITT standardization is due to Soheili et al. (1993) in the United Kingdom.

5. Variable-Rate Speech Coding

Generally, speech coding algorithms produce a constant bit-rate data stream as is usually required for digital transmission systems. However, the short-term entropy in the speech signal varies widely with time so that variable-rate coding is more natural and efficient. A much lower average bit rate for a given reproduced speech quality is achievable if the rate can be allowed to vary with time. Typically, variable-rate coders switch from one rate to another at intervals as frequent as 20 ms. The rate may be controlled *internally* by the statistical character of the incoming speech signal and/or *externally* by the current traffic level in a multi-user communications network.[13]

Applications motivating the study of variable-rate speech coding are:

- Speech storage for voice mail systems,
- Packetized voice systems for fast packet transmission and switching communication systems,
- Digital speech interpolation (DSI) for digital circuit multiplication equipment (DCME), and
- Code division multiple access (CDMA) systems for satellite and terrestrial mobile radio transmission.

Most of the literature in this area has been based on older coding methods such as ADPCM (Japan—Yatsuzuka, 1982; Nakada and Sato, 1990) or subband coding (Canada—Wu and Mark, 1990; Norway—Lundheim and Ramstad, 1986). Recently, a few interesting studies on variable-rate coding based on CELP have been reported

13 For a recent review of variable rate coding, see A. Gersho and E. Paksoy, "Variable Rate Speech Coding for Cellular Networks," in *Speech and Audio Coding for Wireless and Network Applications*, Eds. Bishnu S. Atal, Vladimir Cuperman, and Allen Gersho, Kluwer Academic Publishers, 1993.

from France, the United Kingdom, and Italy (DiFrancesco, 1990; DiFrancesco et al., 1990; Vaseghi, 1990; Wong, 1992; DeIacovo and Sereno, 1991).

An important component in variable-rate speech coding is *voice activity detection* (VAD), which is needed to distinguish active speech segments from pauses when the speaker is silent but only background acoustical noise is present. In the topic of VAD algorithms, non-US work appears to lead the field. An effective VAD algorithm is critical for achieving low average rate without degrading speech quality in variable-rate coders. An example of a good VAD scheme for asynchronous transfer mode (ATM) digital transmission is the work of Nakado and Sato (1990) in Japan. The design of a VAD algorithm is particularly challenging for mobile or portable telephones due to vehicle noise or other environmental noise. The most notable contribution in this category is from Freeman et al. (1989) from British Telecom in the United Kingdom. Their VAD technique has been adopted as a part of the European GSM digital mobile telephony standard (Southcott et al., 1989; Braun et al., 1990). We are not aware of any significant work done within the United States although it is likely that good quality proprietary algorithms have been developed in US industry for DCME applications.

6. Very-Low-Rate Speech Coding

Important work on speech coding for 300 b/s and lower has been performed at NTT in Japan by Shiraki and Honda (1988; Honda and Shiraki, 1988). The area of vocoders operating at rates below 1000 kb/s is a much more specialized domain of importance in defense applications but, until recently, of little or no commercial interest. Most of the pioneering work in this area has been done in the United States at BBN, Lincoln Labs, Signal Technology, Inc. (STI), and the National Security Agency (NSA). The Japanese work is probably motivated by an emerging interest in satellite paging systems with voice messages.

7. LPC Parameter Coding

In CELP and many other speech coders, the linear prediction parameters are used in modeling the signal and are quantized and transmitted every 20 or 30 ms. These parameters consume a large fraction of the total bit-rate for low-rate coders. Hence, considerable efforts have been invested in finding efficient ways to represent these parameters, most of them involving the use of vector quantization. Of the various US efforts in this area, the benchmark work that is often used as a reference for comparing other results is by Paliwal and Atal.[14]

Most of the work in this area is based on the quantization of *line spectral pairs* (LSPs), also known as *line spectral frequencies* (LSFs) and originally introduced by Sugamura and Itakura (1981) in Japan more than a decade ago. Professor Itakura was also the pioneering inventor of LPC coding, developed while he was at NTT in 1967. Some high-quality work on attaining "transparent" perceptual quality coding of LPC parameters at low rates using multistage VQ while controlling complexity was reported by V. Cuperman and colleagues in Canada (Bhattacharya et al., 1992). Another multistage VQ method for quantization LPC reflection coefficients was proposed by Chan and Law (1992) in Hong Kong; and a multistage VQ technique including a partially adaptive codebook was introduced by Tanaka and Taniguchi (1993) in Japan. A new computationally efficient algorithm for finding LSP parameters was reported by LeGuyader and colleagues in France (Saoudi et al., 1992). The use of thinned lattice filters for efficient LPC modeling and quantization was introduced by Chan (1992) in Hong Kong. A novel set of LPC parameters that shows promise of having some advantage over the LSP representation was introduced recently by Bistritz and Peller (1993) in Israel. An LSP coding method that adapts to the long-term history of the speech spectral parameters was introduced by Xydeas and So (1993) in the United Kingdom. Finally, some interesting methods for LSP quantization that take into account the effect of channel errors can be found in the works of Hagen and Hedelin (1993) in Sweden and Secker and Perkis (1992) in Australia.

8. Pitch and Voicing Determination

A vital building block in low-bit-rate speech coding algorithms is the algorithm for determining the pitch or fundamental frequency in voiced speech and tracking its time evolution. There has been a large volume of research on pitch detection. In Germany, Hess (1983) published a very long and comprehensive book with an extensive bibliography; and a subsequent book chapter by Hess (1992) on pitch and

14 K. K. Paliwal and B. S. Atal, "Efficient Vector Quantization of LPC Parameters at 24 Bits/Frame," *Proc. IEEE Int'l. Conf. Acoust., Speech, Sig. Process., Toronto, Canada, May 1991*, 661–664.

voicing determination provides an updated review of research techniques in pitch detection. Recently, a novel, unusual, and apparently effective technique was introduced by Dologlou and Carayannis (1989, 1991) in Greece and further studied and enhanced by Hult (1991) in Sweden. Unlike most pitch analysis algorithms, this method iteratively applies a sequence of lowpass filtering operations on the speech wave form, resulting in the extraction of a sinusoid at the fundamental frequency.

Closely related to pitch is the topic of voicing. Although not needed for the generic CELP algorithm, the determination of whether or not a speech segment is voiced is a critical feature of virtually all vocoders. It is also needed for some modified CELP algorithms. A new voicing classification algorithm that is adaptive to changing speaker characteristics was recently reported by Moulsley and Holmes (1989) in the United Kingdom.

9. Audio Coding

Audio coding usually refers to the compression of high-fidelity audio signals, that is, with 15- or 20-kHz bandwidth for consumer hi-fi, professional audio including motion picture and HDTV audio, and various multimedia systems. Sometimes the term audio coding is also used to refer to wideband speech coding, the compression of 7-kHz bandwidth speech audio for video teleconferencing and for future integrated subscriber digital network (ISDN) voice communication where higher-quality speech is feasible and desirable.

Digital coding of audio probably began in the early 1970s. Initial efforts simply used uniform or nonuniform (for example, logarithmic) quantization of audio samples for digital transmission and storage. The BBC (British Broadcasting Corporation) developed an audio compression scheme called NICAM (nearly instantaneous companding audio multiplex) for digital audio transmission. NICAM uses a block adaptive-gain amplitude scale where one of five scale factors is specified for every block of 32 samples, represented with 10 bits per sample. Including overhead bits, the NICAM standard carries a stereo audio signal of 15-kHz bandwidth at a rate of 728 kb/s. For an overview of NICAM and its application to digital transmission, see, for example, Rumsey (1990).

Virtually all the current work in hi-fi audio coding relies on either subband or transform coding for the signal decomposition, scalar quantization and entropy coding, and on perceptual masking models for determining adaptive bit allocations across the spectral domain signal components. Much of the work in hi-fi audio coding has been done in Western Europe, particularly Germany, France, and The Netherlands. Some important work based on transform coding was also done in the United States by Johnston (1988) at AT&T Bell Labs.[15]

The Fraunhofer Institute in Germany, in conjunction with the University of Erlangen, has also been involved in audio coding under a government grant; the notable work is by K. Brandenburg and his colleagues (1990), who used transform coding for the signal decomposition. The AT&T work was subsequently combined with the work of Brandenburg et al. (1991) in Germany to form a coding algorithm known as ASPEC (adaptive spectral perceptual entropy coding of high-quality music signals). Brandenburg was also a visiting researcher at AT&T, where he collaborated with Johnston.

The first major event in audio compression was the development of MUSICAM (masking pattern adapted universal subband integrated coding and multiplexing), which was adopted in Europe for use in digital audio broadcast (DAB/Dehery et al., 1991). A closely related subband coding method is MASCAM (masking pattern adapted subband coding and multiplexing) developed at IRT (Theile et al., 1988). Most of the current international interest in audio compression algorithms is centered around the recent ISO/MPEG audio standardization activities. The dominant contributions to this standard come from Philips in The Netherlands, IRT in Germany, and CCETT in France. For an outline of the MPEG audio algorithm, see Brandenburg and Stoll (1992).

The MPEG audio algorithm has three layers of coding, of increasing complexity and quality allowing different versions to be suited for different applications. Layers 1 and 2 are based on MUSICAM, with a simplified version used for the first layer. A filter bank of 32 equal-size bands is used. Layer 1 has the lowest complexity and worst quality; it does simple perceptual weighting for bit allocation and is less adap-

15 J. D. Johnston, "Transform Coding of Audio Using Perceptual Noise Criteria," *IEEE J. Select. Areas Commun.*, *6*, (1988), 314–323.

tive to transitory material. Layer 2 uses a better perceptual model, is more flexible in sending gains for blocks of samples in one band or shared for two or more adjacent bands. Block companding is used in each subband to quantize blocks of 12 samples each. Quantization resolution is determined by masking model. Layer 2 differs from Layer 1 only in the joint quantization and coding of each triplet of scaling factors from three consecutive companding blocks in each subband. Layer 3 has the highest complexity and best quality, and consists of a combination of the ASPEC transform coding algorithm and the MUSICAM filterbank; an overlap-DCT transform is performed in each subband of a subset of lower subbands. Each layer of the standard also includes a technique for joint coding of two stereo channels (Herre, 1992).

The two new consumer hi-fi audio products that aspire to replacing the compact disc both use audio coding. They are the DCC (digital compact cassette) from Philips and the minidisc from Sony. Other industrial partners are involved in these. Both use compression based on perceptual masking methods. The DCC scheme is called *precision adaptive subband coding*, or PASC, the minidisc system is called *adaptive transform acoustic coding for minidisc*, or ATRAC. PASC is quite close to the ISO MPEG Layer 2. The ATRAC coder is the result of work by Tsutsui et al. (1992) from Sony in Japan; and PASC is by Lockhoff (1992), at Philips in The Netherlands.

A list of key European researchers in audio coding and their affiliations is given in Table VI.1 in Section VI.E.

Japan has also been active in audio coding, particularly with regard to consumer audio products; but some interesting research ideas have also originated in Japan. Sony, NEC, and Matsushita, in particular, have ongoing efforts in audio compression. Specifically, an adaptive block size transform coding approach was reported by Iwadare et al. (1993) at NEC. This is especially well suited to perform temporal masking, as well as the usual frequency-domain masking. Several Japanese companies are actively involved in making commercial IC chips to implement the MPEG algorithm.

Good examples of recent work in wideband speech coding are by Laflamme et al. (1991) and Salami et al. (1992) in Canada, and Fuldseth et al. (1992) in Norway. In each case, the algorithm is based on CELP.

10. Speech and Audio Quality Assessment

One of the most difficult aspects of speech coding is the assessment of speech quality produced by an algorithm. In the earlier days of pure waveform coding, signal-to-noise ratio (SNR) served as an adequate measure of quality and was easily used as a guide to algorithm design and parameter optimization. However, with the advent of closed-loop or analysis-by-synthesis coding methods (such as MP-LPC and CELP), a perceptually weighted mean-square error criterion guides the selection of excitation vectors, and, as a result, SNR has become a grossly inadequate measure of speech quality. For vocoders, where waveform matching is entirely absent, SNR has always been irrelevant, and subject tests have been the only way to assess quality. With the advent of widespread standardization efforts for speech coding in recent years, costly and time consuming subjective testing has been taking place on a large scale. Consequently, there is an expanding interest in finding objective measures of perceptual quality that will correspond closely to subjective testing results. An early US effort on objective measures is summarized in the book by Quackenbush et al.[16]

Leading the study of objective measures is the extensive work of Kitawaki and his colleagues at NTT in Japan (Itoh and Kitawaki, 1989; Kitawaki, 1991; and Irii et al., 1991). They have used an artificial voice generation process for calibrating quality measures and have done regression analyses to correlate objective measures with subjective testing results for a multilingual data base. They find that *cepstral distance* is a particularly effective spectral domain measure of quality. A pattern recognition approach to correlating objective and subjective results has been reported by Crowe (1992) at British Telecom in the United Kingdom. Another approach to quality measures, called *attribute matching,* was introduced by Halka and Heute (1992) in Germany, who introduced a new objective measure and a random process model of speech for quality evaluation. Continued attention to objective quality measures is encouraged by the existence of a CCITT study group seeking a standardized objective quality measure for future evaluations of speech quality. Recently some interesting and very promising studies of perceptual masking models for use in speech coding were reported by Sen et al. (1993} from Australia. This work shows that those

16 S. R. Quackenbush, T. P. Barnwell, III, and M. A. Clements, *Objective Measures of Speech Quality,* Englewood Cliffs, NJ: Prentice Hall, 1988.

spectral regions of speech that lie below an auditory threshold function of frequency can be completely filtered out with negligible perceptual effect. The result indicates that a new auditory measure of quality may significantly enhance the excitation search mechanism of CELP algorithms, allowing much reduced codebook sizes (and bit rates) or enhanced quality for a given rate.

Objective measures of perceptual quality for wideband audio signals, based on models of the human auditory system, have also been studied by Paillard et al. (1992) in Canada and by Brandenburg and his colleagues (Herre et al., 1992). The work of Herre et al. is particularly interesting because they have actually implemented hardware that measures perceptual quality in real-time.

11. Applications

While there is a notable amount of ongoing fundamental research in what is called "speech science," most of the technologically significant research in speech coding is application-oriented and tends to occur in industry or government research labs rather than at universities.

Specific application areas for speech coding are;

- Severe bandwidth constraints of digital cellular telephones for wireless applications, as well as security and reliability, motivate digital rather than analog transmission;
- General telecommunication systems and DSI systems for multichannel applications;
- DCME equipment for submarine fiber cable and satellite;
- PCS personal communication networks, also wireless;
- Satellite-based mobile (high-altitude satellites) low-altitude orbiting satellite-based mobile telephony (for example, Iridium); VSAT terminals particularly for underdeveloped countries;
- Rural wireless telephony systems for underdeveloped countries;

- Audio for videophones and video teleconferencing;
- Voice mail/messaging speech for store and forward communications;
- Voice message paging systems;
- Voice storage for telephone answering machines;
- Phone logging recorders;
- Secure voice terminals, for example, STU III.

D. PROJECTIONS FOR THE FUTURE

Where the action is in speech coding is at steadily decreasing bit rates. For example, when the ADPCM standard of 32 kb/s was adopted in 1983, research on such high-rate coding more or less died. With adoption of the new 16-kb/s CCITT algorithm, there is relatively little remaining research interest at this bit rate (other than implementation issues related to the g.728 standard. This standard may be adopted for PCS applications. The GSM RPE LPC standard was based on source rate of 13 kb/s, and the (North American) Telephone Industry Association (TIA) standard with VSELP at an 8-kb/s source rate. The Japanese Digital Cellular (JDC) standard is based on the VSELP algorithm at a source rate of 6.7 kb/s. Most of the mobile applications currently focus on "half-rate" second generation of digital speech compression, where the rate is half of the above full-rate values.

Following the imminent adoption of "half-rate" standards for speech coding in Europe and Japan, research efforts in the next five years are likely to concentrate on coding algorithms for 2.4 kb/s, with some continued activity at 4.8 and 8 kb/s for specialized applications. Increased focus on vocoder approaches is expected for 2.4 kb/s, because the performance of code-excited linear prediction (CELP) and other waveform coding methods algorithms is inherently limited at this rate. Research is likely to become increasingly dispersed as more Asian countries (South Korea, Hong Kong, China, Malaysia) expand their university-based research in speech coding. We see little indication of increased emphasis on basic research worldwide, although some significant and highly innovative work could possibly come out of academic institutions.

E. KEY NON-US RESEARCH PERSONNEL AND FACILITIES

Table VI.1 lists the key non-US researchers in audio coding and their affiliations. Audio coding is a narrow area; all those listed in the table have similar expertise. Table VI.2 identifies the "active" non-US researchers in speech coding. The descriptor "active" rather than "key" is used, because the non-US activity in speech coding is characterized by a large number of relatively minor participants (the leadership lies in the United States), so that there is no small subset of these researchers who really stand out as leaders—the exceptions being the Canadians, Adoul, Cuperman, and Kabal, who might be considered as "key" non-US researchers. No specific areas of expertise are given for this list of speech coding researchers because most of them have worked in the same dominant areas of CELP speech coding.

TABLE VI.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—
AUDIO CODING

Researcher	Affiliation
	France
Yves François Dehery Pierre Urcun	Centre Commun d'Etudes des Telediffusion et Telecommunications (CCETT), Cesson Sevigne
	Germany
Karlheinz Brandenburg	Erlangen University, Erlangen; Fraunhofer Gesellschaft, Institute für Integrierte Schaltungen, Erlangen
H. Gerhauser	Fraunhofer Gesellschaft, Institute für Integrierte Schaltungen, Erlangen, Germany
Martin Link Gerhard Stoll	Institut für Rundfunktechnik (IRT), München/Munich
	The Netherlands
Leon van der Keerkhoff Gerard C. P. Lokhoff Raymond N. J. Veldhuis	Philips Laboratories, Eindhoven

TABLE VI.2
ACTIVE NON-US RESEARCH PERSONNEL AND FACILITIES—SPEECH CODING

Researcher	Affiliation
	Australia
K. K. Paliwal	University of Brisbane, Brisbane
W. H. Holmes	University of New South Wales, New South Wales
Andrew Perkis	Wollongong University, Wollongong, New South Wales
	Canada
Paul Mermelstein	Bell-Northern Research, Ltd., Montreal, Quebec
W. P. LeBlanc S. A. Mahmoud	Carleton University, Ottawa, Ontario
Karen Bryden Hisham Hassanein	Communications Research Centre, Ottawa, Ontario
Vladimir Cuperman	Simon Fraser University, Burnaby, British Columbia
P. Kabal	McGill University, Montreal, Quebec
C. Laflamme Philippe Mabilleau R. H. Salami	Sherbrooke University, Sherbrooke, Quebec
	Denmark
John Aa. Sorensen	Technical University of Denmark, Lyngby
Henrik B. Hansen Henrik Nielsen	Telecommunications Research Laboratory, Horsholm
	France
R. J. Di Francesco C. Lamblin A. Le Guyader	Centre National d'Etudes des Télécommunications (CNET)/ France Telecom, Lannion
M. Delprat	MATRA Communication, Bois d'Arcy Cedex
N. Moreau	Telecom Paris

TABLE VI.2
ACTIVE NON-US RESEARCH PERSONNEL AND FACILITIES—SPEECH CODING (cont'd.)

Researcher	Affiliation
	Germany
Karl Hellwig Peter Vary	Philips Kommunikations Industrie AG, Nürnberg
P. Noll	Technical University, Berlin
J. W. Hess	University of Bonn, Bonn
H. Reininger D. Wolf	Applied Physics Institute, University of Frankfurt (Institut für Angewandte Physik der Universität Frankfurt am Main), Frankfurt
J. E. Natvig	Affiliation unknown
	Italy
Rosario Drogo de Iacovo Roberto Montagna Daniele Sereno P. Usai	Centro Studi e Laboratori Telecomunicazioni SpA (CSELT), Torino
F. G. Andreotti	Italtel SIT, Milan
Maurizio Copperi	SIP Headquarters, Torino
	Japan
Y. Ohta Y. Tanaka Tomohiko Taniguchi	Fujitsu Laboratories, Ltd., Kawasaki
Y. Yatsuzuka	KDD Co., Ltd., Tokyo
S. Saito	Kogakuin University
F. Itakura	Nagoya University, Nagoya
T. Araseki T. Miyano K. Ozawa	Nippon Electric Co. (NEC)
Masaaki. Honda A. Kataoka Yoshinao Shiraki	Basic Research Laboratories, Nippon Telephone and Telegraph Corporation (NTT), Tokyo

TABLE VI.2
ACTIVE NON-US RESEARCH PERSONNEL AND FACILITIES—
SPEECH CODING (cont'd.)

Researcher	Affiliation
	Japan (cont'd.)
Kazunori Mano Takehiro Moriya	NTT Human Interface Laboratories, Musashino/Tokyo
Kenzo Itoh Nobuhiko Kitawaki	NTT Telecommunication Networks Laboratories, Musashino/Tokyo
Masahiro Taka	NTT Transmission Systems Laboratories, Kanagawa
Tomokazu Morio	Sharp
M. Nishiguchi	Sony
Masami Akamine Kimio Miseki	Toshiba Research & Development Center, Kawasaki
	Norway
K. K. Paliwal Andrew Perkis Tor A. Ramstad	Norwegian Institute of Technology, Trondheim
E. Harborg	SINTEF DELAB, Trondheim
	Poland
P. Dymarski	Technical University of Warsaw, Warsaw
	Portugal
Luis B. Almeida Isabel M. Trancoso José M. Tribolet	INESC/Instituto Superior Tecnico (IST), Lisbon
	Spain
L. A. Hernandez-Gomez	ETSI Telecomunicación–UPM, Madrid

TABLE VI.2
ACTIVE NON-US RESEARCH PERSONNEL AND FACILITIES—
SPEECH CODING (cont'd.)

Researcher	Affiliation
	Sweden
Anders Bergstrom Per Hedelin	Chalmers University of Technology, Göteborg (Gothenburg)
B. Hedberg H. Hermansson P. Lundqvist T. B. Minde U. Wahlberg	Ericsson Radio Systems AB, Stockholm
	United Kingdom
I. Boyd	British Telecom
S. V. Vaseghi	East Anglia University, Norwich
M. Easton	Liverpool University, Liverpool
C. S. Xydeas	Manchester University, Manchester
P. W. Elliot T. J. Moulsley	Philips Research Laboratories, Surrey
Barry G. Evans Ahmet M. Kondoz S. Yeldener	University of Surrey, Guildford

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CHAPTER VI: SPEECH AND AUDIO COMPRESSION
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CHAPTER VII
IMAGE AND VIDEO COMPRESSION

A. SUMMARY

Research in image and video compression abroad has tended, for the most part, to be of comparable quality to that in the United States, but lagging in key areas such as transform coding, frequency decompositions, motion compensation, and vector quantization. Two notable exceptions are boundary- and segmentation-based coding and model-based coding, both of which originated and are largely championed abroad. Model-based coding especially represents an important long-term research direction.

Non-US industrial concerns have shown their ability, however, to generate good applied research aimed at commercializing compression technology, particularly in the area of consumer electronics. There is little indication that any US lead in basic compression research will manifest itself in significant reversals in US losses in this industry.

Little research is in progress, in the United States or abroad, on compression algorithms specifically tailored towards binary images such as facsimile or images produced by halftoning gray-scale images. Halftoning itself is a form of compression, but the literature reflects little research. Error diffusion and dithering techniques remain the dominant approach to halftoning, and generally accepted standards for binary image compression, both lossy and lossless, are not being seriously challenged by new methods.

With the growth of picture archiving and communication systems (PACS) and hospital information systems (HIS), a great deal of research and development are aimed at developing algorithms, validation procedures, communication and storage systems, and standards for medical images. Many believe that the future will see nearly all-digital radiology departments, with the one possible exception being analog acquired X-rays. So far, most compression schemes considered have been lossless, but there is a growing effort to develop lossy schemes such as transform codes, especially for archiving, recall, and educational purposes. It is unlikely that lossy images will be used for primary diagnosis for several years, but the use of computer-aided diagnosis techniques combined with compression may someday change this. The majority of work on medical image compression and on PACS, in general, is being done in Japan and in Western Europe, especially in The Netherlands, Belgium, and Denmark. The driving force there is to make maximal use of existing imaging centers and ease the sharing of images among such centers and the hospitals, clinics, and individual practitioners that they serve.

Geographically, the largest concentration of research is in Japan, and Western Europe, with many university and industrial laboratories participating. South Korea and Taiwan also show a surprising amount of research activity for their size. Pockets of important research activity can also be found in other countries, such as Australia, Singapore, and Canada.

B. INTRODUCTION

Among important natural and synthetic signal sources, images and video are some of the most prodigious consumers of communication and storage resources. The adage, "a picture is worth a thousand words," has an interesting parallel in digital signal processing: a sampled speech waveform requires around 100 kbps to transmit without compression, while an uncompressed video signal requires around 100 Mbps, a thousand times as much.

Given the strong advantages of digital transmission and storage of images and video in terms of robustness and interoperability with other data sources and processors, it seems inevitable that images and video will eventually be stored and transmitted, for the most part, in digital form. However, even with current increases in storage capacity of memory, magnetic, and optical media, it would probably take at least another decade before systems without compression would be widespread. Although compression of images and video has clear economical benefits, the compression machinery itself must also be affordable and fast enough to execute compression and decompression algorithms on large images in reasonable time and on video in real time. While digital compression of speech has been practical and widespread for some time, the required level of performance and cost for images and video have only become practical in the last few years, with continuing improvements in microelectronic processors.

Compression of images and video is now past the threshold of introduction and is being deployed in large-scale industrial and consumer applications. The importance of mastering this technology in order to be competitive in product value has been recognized by most technology development institutions, both domestically and internationally. While it is fair to say that US-based laboratories lead the way in development of the technology, it is also clear that significant research and commercialization of image and video compression are occurring in Europe and the Pacific Rim.

C. TECHNIQUES

Research into fundamental compression techniques for image and video coding has been underway for more than two decades. While some important techniques have emerged as keys to large-scale applications, many are still actively researched for niche applications or for future applications where requirements in terms of cost, performance, and efficiency vary. This section focuses on research into compression techniques for image and video compression that is underway outside the United States.

1. Discrete Cosine Transform

Image and video compression systems often use a "transform" to convert each small block of an image (for example, 8x8 pixels) into an alternative mathematical domain where energy and perceptual significance are concentrated in a relatively small set of coefficients. This allows the compression system to concentrate its use of channel or storage bits on these coefficients while coarsely representing or ignoring other, less important coefficients. Among many possible transforms, the discrete cosine transform (DCT) is by far the most popular due to a combination of good performance on typical image and video source material and widespread and efficient implementations in software and special-purpose silicon. The DCT is a key component of the major image and video compression standards described in a later section.

Research abroad in DCT concentrates on efficient implementations and special architectures for silicon. For example, Wu and He (1992) at Southeast University, in China, found a small reduction in the number of multiplies required in fast DCT. Bai and Yang (1992) at National Cheng Kung University, in Taiwan, found a way to reduce computations when zig-zag scanning of coefficients accompanies a transform, as occurs in most state-of-the-art digital video compression systems. Such reductions could potentially provide cost advantages in hardware and software implementations of the DCT, but recent research abroad has not provided improvements large enough to cause dramatic changes.

Jain et al. (1992), in Germany, and Demassieux and Jutland (1991), in France, went one step further by describing actual, fabricated DCT chips. These designs are internal to the two major European consumer electronics companies, Philips and Thomson, and demonstrate a parity in implementing this key image compression component with US semiconductor manufacturers.

Koh et al. (1991) at Nanyang Technological University, Singapore, pursued an intriguing investigation of a non-orthonormal transform that provides better compaction that the DCT. While interesting theoretically, such alternative transforms will have to show greater improvements in compaction in order to displace the DCT in any practical applications.

Realization of more efficient implementations may give a particular manufacturer a cost advantage in a standards-based product. However, given the significant amount of existing research already available on efficient implementation, future work is not likely to produce major breakthroughs. It will, however, serve to make individual institutions current on the technology and hence enable them to develop DCT components internally rather than purchasing them from US-based suppliers.

DCT techniques also receive attention in combination with other compression approaches, especially vector quantization (detailed in Section VII.C.7).

2. Motion Compensation

Motion compensation is a technique for increasing video compression efficiency by predicting frame-to-frame motion of small regions of an image as small offsets relative to the previous image. Only those areas of a particular image that were not well predicted will require significant use of channel or storage bits for their representation. The process of evaluating each potential offset for a particular block is typically the most computationally intensive portion of a video compressor.

One area of research in motion compensation consists of finding efficient implementations of typical contemporary algorithms. As with the DCT, this is likely to enable non-US manufacturers to develop their own implementations. However, there is more room for optimization and refinement of motion compensation procedures than for DCT, and some such research may result in a non-trivial competitive advantage for a particular manufacturer. Rampa et al. (1989), in Italy, demonstrated an intriguing architecture for pel-recursive motion estimation chips (unfortunately, pel-recursive motion estimation is much less widely used than blockwise motion estimation, so this work is not likely to be widely exploited). Researchers at Thomson in France, on the other hand, have developed an efficient semi-systolic implementation of blockwise motion estimation in commercial form; while Pirsch, at the University of Hannover in Germany, has detailed the tradeoffs in several classes of architectures for silicon-based blockwise motion estimation. As with the DCT, this work suggests that non-US manufacturers are likely to be as capable as US silicon companies in moving these functions onto dedicated silicon.

New types of motion compensation are also being researched with the goal of providing more accurate motion tracking (which can be useful if the motion vectors are also used to guide interpolation of missing frames) and use of only transmitted information for motion prediction (avoiding the need to send motion vectors as side information). Jang and Kim (1991), at the Korea Advanced Institute of Science and Technology (KAIST), provided some surprising results indicating that backward prediction of motion, in which the decoder calculates motion vectors based on received data, provides performance improvements in low-bit-rate video compression. This approach has been considered too unstable due to bad motion vector choices based on the quantized information, but Jang and Kim showed that providing a choice of forward and backward prediction overcomes the stability problem. Xie et al. (1991), at the University of Leuven in Belgium, developed variable block size motion estimation combined with an algorithm that uses a hybrid of recursive estimation and exhaustive search. The use of variable block size motion estimation appeared as a subject of serious consideration in the United States and abroad at roughly the same time; another important non-US example of this type of motion estimation is the work by Chan et al. (1990) at Telecom–Australia and Imperial College, London.

One of the more recent advances in motion compensation is the application of bidirectionally interpolated frames. This approach, in which some frames combine motion compensation from past frames with motion compensation relative to a (previously transmitted) future frame, was incorporated into the MPEG standard (see below). Hobson and Carmen (1991), at Britain's National Transcommunications Ltd., show their grasp of the advantages of this approach.

Continued progress in motion compensation will likely be important to improvement of existing systems and development of new ones, and hence bears tracking. Such progress generally relies on significant increases in computational power, and will consequently be paced by processor and silicon performance improvements. Because of this, research facilities with the best computer resources will be more likely to lead in such developments rather than follow.

3. Differential Pulse Code Modulation

Differential pulse code modulation (DPCM) is one of the oldest fundamental compression techniques. Rather than transmitting the value of each individual pixel, its value is predicted from one or more nearby (already transmitted) pixels, and only the difference between predicted and actual values needs be transmitted. The technique provides only moderate compression but is simple to implement. It is often one of the first techniques used in high-throughput compression applications (until processing power increases enough to allow economical application of higher-efficiency algorithms). DPCM was used in early video and high-definition television (HDTV) systems.

DPCM techniques continue to receive considerable research interest abroad, likely due to their simplicity and hence ease of implementation in initial forays into compression-based products. For instance, Kang et al. (1992), at KAIST, tried using a median predictor in place of the linear predictor normally used in DPCM. There have been more novel and exhaustive analyses, such as the use of adaptive filtering techniques in the work by Tziritas and Pesquet (1992) at CNRS in France, but, overall, theoretical and algorithm work in DPCM tends to be uninteresting.

An application where multi-resolution coding is more likely to have a significant effect is progressive image transmission. In a standard (or sequential) image transmission, the image essentially fills in from top to bottom as it is received. It may be necessary to have received the majority of the bits for the image before a viewer can decide whether it is the desired image. A progressive transmission technique provides for the decoder to generate a coarse representation of the image using only a small fraction (for example, less than 10 percent) of the bits used to represent the entire image. This allows for more efficient browsing through image data bases. Multi-resolution representations typically can provide a low-resolution version of the image as the initial decoded representation. Vandendorpe (1992), at the Université Catholique de Louvain in Belgium, described research in this area that is a part of broader European digital television and HDTV work. While no US HDTV proponent is currently advocating the use of a progressive or hierarchical decomposition, there is a chance that such an approach will be adopted later in European digital HDTV standardization.

7. Vector Quantization

Vector quantization (VQ) is a compression technique in which a group of pixels is represented by one entry from a codebook consisting of possible reproductions for the group. The power of vector quantization comes from the fact that, since pixels tend to be related to their neighbors, a codebook required to contain good representatives for likely blocks may be much smaller than a codebook required to contain exact representatives for every possible block. Vector quantization has a strong theoretical appeal, since as vector quantizer block size increases, the performance of a vector quantizer can approach the ultimate performance possible for a source with constant statistical properties.

Vector quantization tends to have high-compression complexity due to the necessity of determining which of the possible representatives is the best one, although various types of structured search can mitigate this problem. Decoding is very simple, however, consisting of a table look-up, so that VQ has been embedded in "software only" decompression algorithms in the United States (for example, with Apple Computer's QuickTime software). Vector quantization is often embedded in more complicated systems, though, in which some form of prediction, frequency-domain decomposition, or adaptation is used to improve performance at the expense of increased complexity.

The fundamentals of vector quantization are now well understood, so that most research in the United States and abroad is focused on hybrids of vector quantization with other techniques. In the areas of image and video coding, Wu and Coll (1991), at Carleton University in Canada, have explored a "grand hybrid" of block truncation coding, vector quantization, and DCT. Unfortunately, the work is more notable for its scope than improvements in compression efficiency.

Basic vector quantization uses a fixed codebook, but image and especially video sources have changing characteristics. Hence, adaptation of VQ in competitive practical systems will likely be either as a small part of a large hybrid in which adaptation to changes is provided by other parts of the system, or through adaptation of the VQ structure itself. Researchers in Italy (Braccini et al., 1992), Japan (Watanabe and Suzuki, 1991; Wang and Ozawa, 1991), and Taiwan (Chen et al., 1992; Chang et al., 1992) have recognized this and are attempting to incorporate various adaptation techniques into practical image and video VQ algorithms. A focus of this work is very-low-bit-rate video compression, in which a US company, PictureTel, has had commercial success with a VQ-based algorithm.

An alternative application of vector quantization is in very low complexity suited to software compression. Researchers at IBM-France and Université de Nice have explored a combination of hierarchical motion compensation and low-complexity VQ that may be useful in this type of application (Raimondo et al., 1991). These efforts have not resulted in breakthroughs, but do represent an important research direction.

Sang Uk Lee and colleagues at Seoul National University in South Korea seem to be making a cottage industry of research into combinations of DCT and VQ (their most recent work is discussed by Kim and Lee, 1992). To date, this has been a fairly obvious combination to try, but has not been particularly fruitful.

8. Boundary- and Segmentation-Based Coding

Boundary- and segmentation-based coding is an area of image and video compression that was initiated and has largely been studied in non-US institutions. The fundamental notion is to find the predominant edges in an image and code their locations and trajectories, and then provide simple interpolative coding for the regions bounded by these edges. The approach is motivated by the apparent importance of edges in human vision.

This approach can provide useful reconstructions at very low bit rates. However, the reconstructions have a strong "cartoon-like" character, due presumably to the importance of also characterizing texture and other second-order features of the regions between boundaries. If required to provide the near-transparent quality generally required of image and video compression, segmentation-based coding is no longer competitive with other approaches (such as DCT-based coding). It also has the disadvantage of fairly significant computational requirements (edge detection, edge tracing, region growing).

Murat Kunt and colleagues at EPFL in Switzerland were among the pioneers of boundary- and segmentation-based coding, and are actively pursuing extensions to sequence coding (Willemin et al., 1991). A similar thread has been picked up by Biggar and Constantinides (1988) at Imperial College in the United Kingdom. Huang (1992), at National Tsing-Hua University in Taiwan, investigated contour coding of image sequences by attempting to extract descriptions and motion information for three-dimensional objects.

Although segmentation-based coding has not moved into the forefront of practically applied techniques, it bears watching, as the fact that it originates from a fundamentally different viewpoint than most "waveform"-based image and video compression techniques provides the possibility of a surprising development.

Fundamental research in this area is surveyed in Chapter IV.

9. Model-Based Compression

Model-based compression is a second major area of video compression that originated and has been pursued largely abroad. It is one that bears special observation, as it has good potential in the long term as an important technique in some very low bit-rate applications.

Typically, model-based compression is applied to the "head-and-shoulders" type image typical of one-on-one video telephony. The position and motion of the talking head are mapped onto a wireframe model of a human head. A relatively small set of parameters is extracted from this mapping and sent to the receiver, which uses them to position its own wireframe model. The technique is most effective when the features of the person speaking are already resident at the decoder.

The technique draws strongly on pattern recognition (to identify the current state of the head at the transmitter) and computer graphics (to animate the head at the receiver) and has more than a passing similarity to the "morphing" now found in special effects sequences in popular movies, in which one face or object smoothly changes to another while maintaining continuity of expression or motion.

Much of this work originated in Japan and is still being actively pursued by Morikawa and Harashima (1991) at the University of Tokyo. Among industrial concerns, Matsushita has taken an interest in the technique (Nobori et al., 1992). Researchers in the United Kingdom have aggressively picked up on the techniques, however, both at the University of Essex, where Kokuer and Clark (1992) have explored the feature extraction stage of model-based coding, and at British Telecom, where W. J. Welsh (1991) has written an excellent treatise on applications of the technique to improve coding at 64 kbps or provide for coding at much lower bit rates. Reinders et al. (1991), at Delft University in The Netherlands, are also generating some impressive results of their own wireframe modeling techniques.

Research is still in the early stages due to the difficulty of the problem and the massive amount of computational power required at both ends. However, important practical applications could emerge in the five- to 15-year time frame, especially as continuing increases in computational power for computer graphics are available from the mainstream computer industry.

10. VLSI and Hardware for Compression

An important area of applied research is the development of architectures and components for high-speed and economical application of image and video compression. This area has seen significant activity overseas as foreign companies work to commercialize technologies regardless of whether they were originated inside or outside their own countries.

One important architectural approach to video compression is to develop a special-purpose programmable processor that includes on-board acceleration of key operations such as motion compensation and DCT. In the United States, such chips have been developed by C-Cube Microsystems and Integrated Information Technology. Development of this type of processor is also very active in Japan: Mitsubishi has developed a 24-bit video-specific DSP (Murakami et al., 1989), and NEC has developed a 16-bit video-specific DSP (Tamitani et al., 1987). A second architectural approach is to develop more highly integrated silicon that is specific to a single-image compression algorithm. For instance, Bolton et al. (1991), at SGS Thomson in the United Kingdom, have developed a single-chip implementation of the JPEG algorithm.

Of the research groups describing implementations of video compression systems, a particularly important organization is National Transcommunications Ltd. in the United Kingdom. This company produces impressive hardware video compression systems such as a 12-Mbps system for terrestrial broadcast (Underwood et al., 1991). It will likely be a strong competitor in the field of broadcast and HDTV video compression equipment.

D. STANDARDS

In the period 1988–1994, international standards will have been defined to cover the key applications of image compression, facsimile, videoconferencing, entertainment video from storage devices, and broadcast entertainment video. The existence of these standards will tend to accelerate the deployment of the technology and lower entry barriers to both domestic and foreign manufacturers. In most cases, this will tend to both grow the market faster and increase foreign competition for US firms relative to what would have occurred in the absence of standards in domestic markets. Non-US markets should be more accessible than they would have been otherwise.

A significant amount of work on efficient implementations of these standards by foreign firms has been published.

1. JPEG

The Joint Photographic Experts Group (JPEG) is a joint committee of the International Standards Organization (ISO) and the International Consultative Committee on Telephone and Telegraphy (CCITT). JPEG has developed a DCT-coding-based image compression standard for still images. This standard will likely see widespread deployment in computer-based image transmission and in electronic cameras. It is also a practical algorithm for use in video compression systems that require frame-by-frame editing (since such systems cannot use prediction between frames).

SGS Thomson in the United Kingdom has developed a single-chip JPEG system (Bolton et al., 1991), as has Sanyo in Japan (Ogawa et al., 1992). Nakagawa et al. (1992), at Toshiba in Japan, have developed a four-chip architecture that adjusts JPEG quantization based on a first scan to detect activity in an image. This type of control is also important in video compression. Suwa et al. (1991), at Sharp in Japan, have described an IBM PC-compatible JPEG compression and decompression board that includes an ISDN interface for still-image compression of images up to 1024x1024 pixels, while Tasaki (1992), at Ricoh, described a system that uses JPEG in a very important application, color facsimile.

2. Facsimile and JBIG

Modified Huffman and Modified Relative Element Address Designate (READ) compression schemes were recommended in the CCITT Group 3 standard. Modified Modified READ was originally developed for Group 4 facsimile, but may be included in revisions of the Group 3 standard.[4] These standards are discussed below.

a. Modified Huffman Coding

This technique uses the principles of Huffman coding, but the encoding is fixed on the basis of what is assumed will be representative of facsimiles in general. Since two-tone facsimile scans generally have long sequences of either black or white, the scheme performs run-length coding followed by Huffman coding of the run-lengths. The run-length coding scheme counts the number of pixels of the same tone occurring in a row in the input stream and records this number instead of the individual pixels themselves. (See Chapter III for a discussion of Huffman coding.)

b. Modified READ Coding

This coding scheme is optional for Group 3 facsimiles and is based on the Modified Huffman coding described previously. This scheme is considered a two-dimensional code, since it uses the previous line as a reference for encoding the current line (the line under consideration at the moment). To achieve this comparison, three different codes are used to relate the position of black run-lengths in the current line to the position of black runs in the reference line. A Vertical Code is used if the shift of the black run is within 3 pixels of the original. A Pass Code indicates that a black run is missing in the current line. A Horizontal Code indicates a black run where none existed in the reference line. Because this coding is very sensitive to transmission errors, a new reference line is encoded using just the Modified Huffman Code every few lines, generally every other line. The Modified READ code is supposed to achieve a 20-percent improvement over the plain Modified Huffman Code.

c. Modified Modified READ Coding

This code is essentially based on the Modified READ Code, with the main difference being that there is only one reference line (the first line). This is an accept-

4 "Terminal Equipment and Protocols For Telematic Services," *CCITT Volume VII—Fascicle VII.3*, Oct 1984.

able approach where digital transmission is used because of the immunity from errors. This scheme was originally designed to be used in Group 4 facsimile equipment, but has also been incorporated into some Group 3 equipment.

d. Mixed Mode Techniques

Mixed Mode compression techniques are allowed under Group 4 recommendations. This approach separates a page into "symbols" and "graphics." The symbols would be letters, numbers, and punctuation that, for example, would be encoded using ASCII character codes. The graphics part, which might include signatures, pictures, and graphic designs, would be encoded using the Modified Modified READ Code.

e. V.42bis Technique

This compression scheme is based on British Telecom's variation of the well-known Lempel-Ziv algorithm (see Chapter III) and is covered under CCITT Study Group XVII: "Data Transmission over the Telephone Network." The technique also includes a transparent mode of operation when the input pixels are not compressible.

f. JBIG

The Joint Bi-Level Image Experts Group (JBIG) has developed a proposed standard for the compression of binary images (although the more general JPEG could also be used for such an application). The international flavor of this standard is exemplified by the tutorial paper by Hampel et al. (1992), the authors being from Germany, Greece, Japan, and the United States. This paper compares the emerging JBIG standard with the existing standards, Groups 3 and 4 as well as JPEG. A summary of the findings is not only instructive, but shows the importance placed on recognized standards.

The JBIG coding standard, like the Group 3 and 4 facsimile standards, defines a lossless compression for binary or two-tone images. One advantage over Group 3 and 4 is superior compression, especially on two-tone images producing approximations to gray scales. On such images, compression improvements of a factor of 10 are common.

A second advantage of the JBIG standard is that it allows for differing resolutions (multi-resolution) so that it is adaptable to the required application. Some equipment does not need or cannot use detailed resolution beyond a certain degree. For such devices, it is a waste of time and money to transmit excessive detail that must be discarded anyway.

It is possible to effectively use the JBIG coding standard for encoding gray scales and color images as well as two-tone images. The simple strategy of treating bit-planes as independent two-tone images for JBIG coding yields compressions at least comparable to and sometimes better than the JPEG standard in its lossless mode. The excellent compression and great flexibility of JBIG coding make it attractive in a wide variety of environments. The publication of this coding as an open standard is expected within a year or so.

3. H.261

The CCITT has developed a set of standards to provide for videoconferencing from 64 kbps to 2.048 Mbps. These algorithms combine motion compensation and DCT in a low-delay implementation. The video compression algorithm component of these standards is H.261. A number of telecommunications equipment manufacturers are developing systems to address the rapidly growing market for videoconferencing equipment.

Uwaba et al. (1991), at Graphics Communication Technologies (GCT) in Japan, have designed a special-purpose chip for the zig-zag scanning, quantization, and buffer control functions of H.261. Tajiri et al. (1991), at NTT in Japan, described an entire H.261 compressor/decompressor (Codec) on a single board (280 mm x 280 mm). Philips Kommunikations Industrie in Germany has developed a working videophone based on H.261 and ISDN (Achhamer et al., 1991). This unit represents a prototype, as it includes a monitor, ISDN phone, and separate (and fairly large) video compressor/decompressor. Nakabayashi et al. (1992), at Sharp in Japan, also have described a complete "personal" video conference system based on H.261 and incorporating a wireless pen-based user interface. These published accounts represent only a fraction of the commercial activity in H.261-based video compression and videophone systems. Currently, the companies with the largest share of the global videoconferencing industry are small US companies (PictureTel, Compression Labs, Video Telecom), but there is clear evidence of H.261 product development by many European, Japanese, South Korean, and Taiwanese organizations.

4. MPEG

The Motion Pictures Experts Group (MPEG) is a committee sponsored by ISO. Its original charter was to study video compression for storage on compact disk (at a total bit rate of around 1.5 Mbps). The result of this work was a motion compensated, DCT-based standard, MPEG 1, now technically complete. MPEG 1 differs from H.261 in that it takes advantage of longer compression and decompression delays than would be acceptable in two-way communication. Since then, the committee has taken on the second task of defining a standard suitable for broadcast transmission of study quality footage (at rates of 4 to 9 Mbps), known as MPEG 2. This standard will be technically complete by mid-1993. A third effort, MPEG 3, was originally oriented towards high-definition television, but it was decided that the MPEG 2 algorithm extended naturally to HDTV as well. Finally, a new effort, MPEG 4, has just been started to pursue very-low-bit-rate video telephony for mobile applications.

One major attempt to develop a product that exploits MPEG standard video is the consumer electronics system CD-I developed by Philips in The Netherlands. Sijstermans and van der Meer (1991) described the more algorithmically complicated encoding task implemented on a parallel computer. Konoshima et al. (1990) at Fujitsu concentrated on the design for an MPEG decoder.

Important MPEG products will include silicon for decoders (SGS Thomson in France has an effort; Philips of The Netherlands has a joint design effort with Motorola), decoder systems and products (such as Philips' CD-I), and encoders and encoder workstations (such as the broadcast-format MPEG encoder developed by National Transcommunications Ltd. in the United Kingdom). US companies will be involved in and face competition from foreign companies in each of these areas.

E. APPLICATIONS

While significant research is oriented toward fundamental compression techniques and the implementation of these individual techniques, and to implementation of standards, other research focuses on particular applications. Such research tends to concentrate on near-term commercial opportunities and hence can be expected to impact US technological competitiveness within 10 years.

This section describes some of the key research efforts targeted at particular applications.

1. Digital Video Cassette Recorder

The video cassette recorder (VCR) is one of the most successful consumer electronics products of the last two decades. Current video cassette recorders are based on analog recording technology (VHS, developed by Japan's JVC, although the basics of video tape recording were developed in the United States by Ampex). Analog recordings tend to degrade with time, and quality suffers from multiple transfers. Digital recording on magnetic tape has continued to increase in efficiency, driven by the needs of the computer industry. Nevertheless, it is very expensive to record uncompressed digital video onto magnetic media (although such machines as D-2 and D-1 tape recorders are available for professional use at costs generally in excess of $40,000). As a consequence, several consumer electronics manufacturers are very actively pursuing compression for digital consumer VCRs. A particular feature of this application (like video telephony) is the need for cheap compression as well as decompression if the consumer is to be able to record as well as play back material.

De With et al. (1992), at Philips Research Labs in The Netherlands, have provided an exhaustive comparison of video compression techniques for application in consumer digital video recording. Hong et al. (1992), at Samsung in South Korea, have explored modification of DCT-based images to support VCR "trick plays," as do Yamamitsu et al. (1991) at Japan's Matsushita.

2. Digital Still Camera

The filmless camera has been a goal of many manufacturers, as it offers the potential of removing the costly and time consuming handling and processing of chemical-based film from photography. While the current state of electronic imaging sensors (CCDs) does not approach the full capability of film, it is good enough to consider filmless cameras for consumer and some industrial applications. Beyond the sensor, a key technology in such cameras is the ability to store a large number of electronic images in the camera. Since the camera typically needs to be small and light, compression of the digital image can be of great benefit.

Research into compression for such applications is evident in Japan at Chinon (a DPCM-based technique—Saigusa et al., 1991), Sanyo (JPEG based—Ogawa et al., 1992), and Toshiba (JPEG based—Nakagawa et al., 1992).

3. Facsimile

In the context of this section, facsimile is always a binary (two-tone, black and white) image consisting of text, line drawings, or halftone images. In examining publications on facsimile compression within the last five years, it is apparent that not much research is being done in this area. While accepting the probability that some publications have not been found, fewer than two dozen US references and only about a dozen references from abroad were identified in searches of the technical literature. It is concluded from this that most of the facsimile compression study has been left to those groups most likely to define international standards. It also appears that there is no sharp division of study between US and non-US groups. Similar topics are studied by both. International conferences on image processing promote a common international approach and interchange of concepts. Some research is not concerned with new aspects of compression algorithms, but rather with better ways to implement them.

Boehn et al. (1990) have introduced an innovative approach to compression of documents made up of text, graphics, and halftone segments, but its high complexity makes it unlikely to become a widely used facsimile compression method. They compare an adaptive predictive algorithm and a Classified Pel-Pattern (CLAP) method with the two-dimensional Group 4 standard. They find that the

CLAP algorithm achieves the best data compression, but at the cost of high complexity. Measured compression ratios for documents scanned with a resolution of 300 pixels per inch vary between 1.3 and 2.5 for halftone pictures and are 50 for line art. For text and graphic documents, Group 4 is only 5 to 10 percent worse, with significantly less complexity; however, Group 4 is unsatisfactory for halftones.

a. Block Coding

Some work has been done on one- and two-dimensional block coding for facsimile, including promising work by Maeder (1988). By dividing an image into small two-dimensional patterns of adjacent pixels, data compression algorithms can exploit the local structure that will have more correlation than when the whole picture is considered. This study used square blocks of pixels. One approach using run-length encoding was achieved by examining the image for the occurrence of each possible pattern. These blocks were then encoded using a Huffman technique. Contour encoding is produced by specifying the contour starting position followed by a succession of pattern pointers and direction indicators for each block pattern found along the edge. The pattern pointers may be Huffman encoded. A Medial Axis Transform (MAT) technique showed the greatest promise, but it could be very slow if not done efficiently.

The results show that local block pattern methods can achieve compression rates as good as or better than corresponding conventional methods. The benefits are obtained chiefly from the two-dimensional nature of the images. Except for contour coding, these methods are less sensitive to image content than conventional ones, as they can cater to both regular and irregular structure in images with no adaptation.

One-dimensional block coding was studied by Cheung (1989, 1990). Code design was based on first-order Markov models estimated from probabilities of occurrence. This approach is highly useful for images with large blank areas, but not so valuable for busy pictures.

Tan and Turner (1988) introduced a new method of compression, called Sequential Block Interleave Coding, which features a fixed-rate output and exhibits a strong immunity to transmission errors. The method encodes one-dimensional block-pairs of binary image data without the need for codebooks. This approach has difficulties if more than 50 percent of the image is black. Because of the fixed-length codeword output and the encoding scheme, maximum possible compression is only 2:1. The approach is unique and seems unrelated to more recognized techniques.

b. Halftoning

Linnainmaa (1988) describes cost effective methods for high-quality representations using only black and white based on the well-known error diffusion and Hilbert curve transversal. To be computationally efficient, Linnainmaa asserts that halftoning methods must not require any global information of the gray values in the image, nor must it require any tedious analysis of the region around each pixel, and once the final value of a target pixel is found, it must not be altered afterwards. The main problem in the simple diffusion method is that the image is traversed very monotonically, row by row. Hilbert curves allow transversing the image in an apparent "controlled disorder." While this method has been suggested previously, two novel effective modifications are described. One modification is to break the image into 16x16 blocks and use the same Hilbert curve for each block. In this way, the Hilbert curve path can be retrieved efficiently by table lookup. A smoothed version of error diffusion is realized by using a threshold level that alternates between the quarter and three-quarter level of the total range of gray values, but at some loss to picture resolution.

Hongu et al. (1992) at NEC proposed a number of innovative steps to improve transmission of facsimile involving text and gray-scale images. Some details of the study are missing, perhaps to protect proprietary interests. The basic steps are:

(1) The image is preprocessed to enhance high-frequency effects by using a two-dimensional transversal filter.

(2) The periodic error diffusion method is applied to the image. This consists of overlaying the periodic characteristic of ordered dithering on the error diffusion method.

(3) The output of the periodic error diffusion technique is processed for data compression by an adaptive prediction method that splits the image into

text and images and predicts the value of pixels. The difference between prediction and actual values is encoded.

(4) Encoding is achieved by the Modified Huffman approach.

This method has a high expression ability for both gray scale and resolution, and moiré patterns (shimmering, interference patterns) are not likely to occur for screened photographs. In addition, this method realizes a high coding efficiency, because the halftone data using this method have a periodic characteristic.

Skarbek et al. (1989) presented results of a comparative study between compression efficiency of raster and Hilbert scan for several lossless image coding techniques applied to dithered binary images. However, this approach does not work well since the images are dithered.

Costa and Sandler (1991) described an application of the binary Hough transform to the compression of a line drawing and a 16-level gray image. The approach consists of using the Hough transform for detection of straight-line segments only, since it is possible to represent any digitized curve this way. This greatly simplifies the computational complexity. The compression was about 4:1, but the reconstructed images lacked quality. The weakness seems to be the edge detection mechanism.

c. Line Drawing

Liu (1991) addressed the compression of line drawings, pictorial representation in which information is conveyed by locations and shapes of thin lines only. Typical examples are sketches, engineering drawings, maps, charts, and handwriting. This paper presents a theoretical model that permits an evaluation of the relative performance of vector chain coding (VCC) and run-length coding (RLC). The superiority of various forms of vector chain coding is demonstrated theoretically and reinforced by experiment for cursive writing, flow diagrams, and Chinese characters. The critical point of line drawing complexity is derived at which VCC becomes less efficient than RLC. Practical line drawings generally have a complexity far less than this critical value. VCC is affected by the number of possible vectors from a point, the minimum number being eight, and the maximum number for the study, 24. Compression is essentially twice as good for 24 vectors as compared to eight vectors.

Compared to run-length compression, VCC is about three times better for the 24-vector case. Considering that these images are hand-drawn, the compression ratios, which vary from 13:1 to 20:1, are impressive.

d. Implementation

Since many users will adopt a recognized international standard, some researchers are concerned with improved implementations rather than the compression technique itself. In fact, many good compression techniques would be useless without efficient implementations. Lyashevich (1991) stresses high-speed processing (not transmission), while Buck (1991) addresses the very practical problem of errors introduced into transmission of images. Some compression algorithms are very vulnerable to errors in transmission.

Buck (1991) reviews the facsimile compression standards, Group 3 and 4. In particular, he discusses an implementation by Oak Technology that can process six facsimile pages (A4), digitized at 300 pixels/inch, in one second. (Transmission rate, however, is determined by the Group 3 or 4 compression algorithm and the communications medium transmission rate.)

Because documents are exchanged over communications channels primarily by the use of Group 3 terminals over public switched telephone networks, a way to improve facsimile quality is to find a means for improving transmission error (noise) immunity. Lyashevich (1991) considered methods of reducing the damage to facsimile that would be caused by transmission errors. (For joint source and channel coding, see also Chapter VIII.) He found that nonredundant message transmissions in modern facsimile communications systems require the use of special methods to protect reception from errors. Systems with decision feedback allow the highest facsimile quality to be achieved, but result in the greatest decrease in transmission speed which becomes unacceptable for an error rate greater than 10^{-4}. Noise-immune statistical codes, particularly Modified Huffman code with line segmentation, cause significantly less loss in transmission speed; and, in combination with nonlinear signal filtering, provide for good quality reception with an error rate up to 0.5×10^{-2}, but require complication of Group 3 terminals and the exchange protocol. Logical filters, which perform nonlinear filtering, do not influence the speed nor method of transmission and do provide satisfactory quality up to 10^{-3} error rate. A logical filter operates only with other output signals of the decoder, and its inclusion as a part of the receiver therefore does not influence the functional capability of various Group 3 terminals.

4. Medical Image Compression

The volume of digital medical images has been growing steadily and is expected to continue to do so. A modest-sized hospital now generates something like 1 terabyte of image data per year. This has led to an enormous increase in research and development of systems capable of storing and transmitting medical imagery. The literature on picture archiving and communication systems (PACS) and hospital information systems (HIS) is now vast. Much of the research and development on PACS have taken place in Japan and Western Europe (especially in The Netherlands) because medical facilities in these nations have not been as willing and eager as those in the United States to purchase many expensive medical imaging devices. Instead they have opted for imaging centers, and they have developed compression and communication protocols for storing the images and communicating them among hospitals and clinics. Many small PACS have been installed, or are in the process of being installed, in Japan, Denmark, and The Netherlands. The emphasis is on images that are already digital, although several systems incorporate digitized X-rays.

Mattheus and van Goor (1992a) of the European Standardization Committee CEN TC 251/4 and the Commission of the European Communities, DG XIII-F/AIM, in Brussels, report on a European program called Advanced Informatics of Medicine (AIM) that is developing standards for medical information systems including medical images. The primary driving force is the need for compatible digital formats and the need to have standard connections of hospital systems to digital networks such as ISDN. No specific image compression techniques have yet been proposed, although compression standards for electrocardiogram (ECG) encoding are being developed. In spite of the strong case made in this paper, a more sobering appraisal of the success of PACS as an aid to the daily work of radiologists is made by Binkhuysen at the Academic Medical Center of Amsterdam (Mattheus et al., 1992b). His paper provides an excellent review of the many potential advantages of PACS for managing medical images, increasing their accessibility, and assisting the radiologists' tasks. These advantages remain in the future, however, for a variety of reasons, including insufficient quality of available digital image display equipment, slowness of the computing and display equipment, the inability to obtain a good case overview (analogous to having an entire study of many images together on a lightbox), and the lack of standards for image representation, storage, and transmission. There is wide agreement on the importance of PACS as an archive because of the minimization of lost images, the speed of retrieval of old images in comparison to hard copy, the easy accessibility of old images, and the ability to store additional case information along with the images.

Most of the research on compression techniques for PACS has taken place in Japan and Western Europe, with very few articles from US institutions. Much of the compression has been accomplished using variations of lossless compression algorithms, as described in Chapter III. Although controversial for diagnosis, lossy compression algorithms are of interest for a variety of reasons, including recall, archiving, education, and fast on-line access with a lossless version available in storage for legal reasons. Virtually all of the lossy compression of medical images has been accomplished by algorithms resembling JPEG, DCT-based transform coders combined with lossless coding.

Representative of work on compression techniques for medical images is the research by Gotoh et al. (1991), at Fujitsu, who employed a hybrid compression algorithm that detects edges and uses BTC if there are edges, and DCT coding if there are not. Using an adaptive bit allocation method, the researchers claim compression of 10:1. Nakagawa et al. (1991), also at Fujitsu, reported high-speed implementation of a similar compression system. Irie and Miyasaka (1992) at Hokkaido Hospital reported on the performance of a PACS incorporating a lossy DCT transform coder. It is interesting to note that these researchers point out that most unhappiness on the part of radiologists was due to their unfamiliarity in dealing with computer terminals. Mizukawa et al. (1992), at the Tottori University School of Medicine in Yonago in Japan, reported a variety of problems with a PACS. They cite a litany of mechanical and technical problems, especially in the image retrieval and display equipment.

Kajiwara (1992), at the Kurume Hospital in Japan, studied the application of JPEG to the digitization of X-ray images and the use of the JPEG file interchange format (JFIF) as a standard format for the digitized X-rays. He argues that 10:1

compression yields a high-quality image that can be communicated over ISDN in hundreds. He admits that highly compressed images are likely to be unsuitable for diagnostic applications, but he argues that the images are quite useful for reference.

The technical literature indicates that Japan and Western Europe (especially The Netherlands, Denmark, and Belgium) are far ahead of the United States in the planning, developing, and implementing of PACS; in the supporting research for image compression, representation, storage, and display; and in the statistical analysis to validate the usefulness of compressed images in medical environments. It cannot be said, however, that any of the systems is yet fully functional or generally accepted by radiologists for ordinary diagnostic work. PACS are making steady gains in popularity for applications such as archival and recall work, and the systems have proved quite useful to physicians wishing to retrieve information from the imaging centers. Many European hospitals already report their intention to become nearly all-digital during the 1990s.

5. Videotelephony

Videotelephony and videoconferencing allow for "face-to-face" meetings without travel. The visual cues available in such meetings generally improve communication efficiency relative to audio-only conversations, and make straightforward the inclusion of visual aids such as charts, graphs, and video tape recordings. Research in implementation of compression-based videoconferencing systems includes efforts by Sharp (Nakabayashi et al., 1992), NTT (Tajiri et al., 1991), and Mitsubishi (Murakami et al., 1989) in Japan, KAIST (Jang and Kim, 1991), and British Telecom (Elliott et al., 1991).

6. High-Definition Television

High-definition television (HDTV) usually refers to transmission techniques that will approximately double vertical and horizontal resolution relative to current video transmission standards (such as NTSC, PAL, and SECAM)[5], as well as incor-

5 NTSC (National Television System Committee) is the television transmission standard in use in the United States, Japan, and several other countries. PAL (Phase Alternation Line) is the standard used in much of Western Europe. SECAM (Sequentiel Couleur avec Mémoire) is used in France, Eastern Europe, and the former Soviet Union.

porating a wider aspect ratio, better colorimetry, and better sound transmission. Such a dramatic increase in required bandwidth is difficult to accommodate with current transmission technology. In the United States, the need for compression has been made even more apparent by the Federal Communications Commission (FCC) decision to standardize on the basis of the ability to fit the HDTV signal into a broadcast slot no wider than current 6-MHz slots for NTSC transmission. Of the remaining proposals before the FCC in the United States, all use digital compression based on motion compensation and DCT.

Major programs aimed at deploying analog-based HDTV systems are underway in Japan (MUSE) and Europe (HD-MAC). The rapid progress of US proposals based on digital compression has surprised HDTV developers in these places, forcing reevaluation of these plans. Research on digital compression for HDTV can be expected to accelerate rapidly as Japan and Western Europe attempt to position themselves in international markets.

Published HDTV research divides into reports on efforts to implement the proposed analog schemes for Japan and Europe (for example, papers by Croojimans [1992] at Philips and Kohiyama et al. [1992] at Fujitsu) and reports on digital HDTV compression. In spite of the European Community's interest in HD-MAC, Italy's Tellettra actually pioneered the development of digital HDTV equipment (Barbero et al., 1991). Hervigo et al. (1992), at EPFL in Switzerland, have described a high-throughput architecture for motion compensation, while Kadono and Yamamitsu (1992) at Matsushita have described an entire digital HDTV compression scheme at 50 Mbps

7. Stereo Image Compression

A relatively new field is efficient compression of stereo image pairs. Such pairs allow for depth perception. Although not found in widespread consumer use (excepting, of course, classic 1950s horror flicks), stereo imagery is used in certain scientific applications, such as aerial and space photography for geological prospecting. Stereo pairs have significant redundancy between the two images. In fact, except for areas occluded in one image, the images are identical up to disparity due to camera location. Efficient compression of stereo image pairs is being pursued using motion-compensation-like techniques in France (Tamatoui and Labit, 1991a) and at

Sony and NTT in Japan (Yamaguchi, 1991). Pastoor (1991), at the Heinrich-Hertz Institut in Germany, goes farther by exploring basic stereo image compression requirements for three-dimensional television (see also Section VII.E.10).

8. Perception-Based Codes and Quality Assessment

Perception-based codes are compression systems that try to consider visual masking effects in order to concentrate quantization noise in regions where it is unlikely to be perceived, usually by adjusting bit allocation in a transform domain so as to put more bits in coefficients important for visual perception. Virtually all published work on perception-based codes is of US origin, especially from AT&T Bell Labs. The one exception is the work of Du Bois et al. (1987) of INRS Communications in Quebec. They experimentally optimized two systems, a DCT and a simple structured VQ, so as to achieve error at perceptual threshold as measured by the CCIR 5 grade subjective testing questionnaire. Although relatively inactive as a research area, this area has the potential for significant improvements in the subjective quality of an image.

A related issue is that of finding meaningful quality measures for compressed images: if one can measure quality in a manner relevant for a particular application, in principle one can change the design to produce better quality. Although popular because of its simplicity, the common variations on signal-to-noise ratio (SNR) are notoriously inadequate as a measure of real image quality. Although the literature on this subject is relatively sparse, a few papers stand out.

Macq and Delogne (1989) at Université Catholique de Louvain related 5 grade subjective scores to a logistic function of SNR. They used weighted SNR with white-noise quantization error approximation to "optimize" transform code quantizer step sizes.

De Valk et al. (1987) published a very remarkable paper, establishing an international (US/non-US) collaborative effort for verifying diagnostic accuracy and image quality through standard tests (based on questionnaires and receiver operating characteristic [ROC] curves). One of the co-authors, D. E. Boekee, is a pioneer in the area of subband image coding, so quite sophisticated compression algorithms are permitted. The cooperative effort is aimed at standardizing image formats and tests.

Ishigaki et al. (1990), at Toshiba Medical Systems in Japan, performed a diagnostic accuracy study of DCT-based compression for a variety of compression ratios using an ROC analysis of the success of finding lung nodules in computerized tomography. They conclude that 10:1 compression was excellent and that the algorithm was useful to 20:1 if suitably optimized.

Lopez-Soler et al. (1990), at the University of Granada in Spain, evaluated subjective performance by mean opinion scores after using a variety of possible distortion measures on LPC speech coefficients. They conclude that the likelihood ratio spectral measure is the best of several distortion measures tested.

Such work on perceptually accurate measurements of the quality of compressed images is important because, in the long run, it can be used as a design tool for better codes for specific applications.

9. Variable-Bit-Rate and Packet Video

Typical video compression schemes lead to a naturally bursty source of digital data, since the appropriate bit-rate increases when complicated motion or scene changes occur. When operated over a fixed-rate channel, these bursts must be smoothed by a buffer. However, buffers are constrained in size by cost and delay considerations. A channel that could accept a variable-bit-rate source might be able to achieve a more constant average quality than a fixed-rate channel.

Packet networks are inherently designed to accommodate variable-rate sources (for example, computer terminals). Unfortunately, they have drawbacks, principally variable delay and the possibility of dropped packets (due to congestion at intermediate nodes). Hence, packet networks offer advantages and disadvantages for video compression. Researchers are attempting to modify existing algorithms or find new algorithms that can exploit the variable-bit-rate advantage while remaining robust against variable delay and packet loss.

Because of the ubiquity of packet networks in computer applications and the trend in telecommunications towards a single-packet-based network for all signals (for example, Broadband ISDN), this area bears careful tracking.

Nomura et al. (1989), at NTT in Japan, and McLaren and Nguyen (1992), at the University of Tasmania in Australia, have provided useful models of variable-rate video sources for validating network models. This is a useful step in attempting to jointly optimize network architectures and compression systems, as full joint optimization is often unwieldy.

Several non-US researchers have proposed complete algorithms and systems for packet video applications. Rossiter and Donovan (1987), at STC Technology Ltd. in the United Kingdom, described a motion-compensated conditional replenishment scheme adapted to variable-rate coding. Ghanbari and Seferidis (1991), at the University of Essex, focused on building a two-layer pyramid system that uses H.261 as a coder for the base layer and a variable rate residual layer; while Sikora et al. (1991), at Monash University in Australia, explore a H.261-based QCIF/CIF pyramid for use over ATM. The residual layer packets are marked as lower priority, and the received picture shows only local loss of resolution when these packets are lost. Hosoda (1991), at the OKI Electric Industry Co. in Japan, has taken a similar approach to broadcast-quality transmission over ATM networks. He differs, however, in using a four-band subband instead of a pyramid structure, and using a non-standard DCT coder in three of the four bands.

Leduc and D'Agostino (1991), at Université Catholique de Louvain and Alcatel, provided a particularly interesting description of a variable bit-rate Codec designed and built for use in a Belgian experimental ATM packet network.

10. Three-Dimensional Television

Kopernik et al. (1992) provided an overview of three-dimensional television. Three-dimensional television poses an interesting compression problem (somewhat similar to stereo audio compression). The basic idea is simply to code one channel, and then to code the discrepancy for the other channel. For interesting work on joint disparity and motion estimation, see Tamatoui and Labit (1991b-c). There is more interest and ongoing work on this subject in Europe and Japan than in the United States, as shown, for example, by meetings on 3-D TV organized in France or European projects like RACE and COST focusing on 3-D TV.

F. SUMMARY BY GEOGRAPHY

The previous sections describe research in compression for images and video on the basis of topic. This section presents a brief alternative breakdown, by geographical location.

1. Pacific Rim

a. Japan

A broad range of research in image and video compression is being pursued in Japan. Among fundamental research techniques, model-based coding receives significant interest, as does vector quantization, although all standards-based techniques are clearly being researched in preparation for implementation. The research literature is dominated by implementation- and application-oriented papers, particularly in special-purpose silicon for video compression/VCR, digital camera, videoconferencing, and HDTV applications.

Although some research, particularly novel and important work on model-based coding, is conducted at universities (Tokyo University, Okayama University, Yamanashi University, Kobe University, Kyoto University), most Japanese research is conducted in the major industrial corporations (Mitsubishi, NEC, NTT, Matsushita, Sony, Fujitsu, Nippon Steel, Ricoh, Sharp, Sanyo, Toshiba, Olympus, OKI).

b. South Korea

South Korea has a mixture of research similar to Japan, but on a smaller scale. Although a significant amount of the research is done at universities (Korea Advanced Institute of Science and Technology, Seoul National University), it is of only moderate novelty and is unlikely to generate breakthroughs. The major South Korean industrial corporations (Samsung, Goldstar, Hyundai) are, like those in Japan, very involved in consumer applications, especially the digital VCR.

c. Taiwan

Some moderately novel research is being pursued at National Tsing-Hua University and at the Industrial Technology Research Institute. Research seems to be concentrated on DCT and vector quantization (especially hybrids). Taiwanese industrial research is not widely published.

d. Other Pacific Rim Nations

There are pockets of research activity of moderate interest in Singapore, Hong Kong, Thailand, and China. Australia supports a concentration of work in variable-bit-rate and packet video. As this will be of increasing importance, Australian activity in this area bears watching.

2. Canada

Canada generates research similar to that in the United States. There are pockets of significant activity in motion estimation and vector quantization. Application-oriented research in the Toronto-Ottawa-Quebec area is likely motivated by Northern Telecom.

3. Europe

a. United Kingdom

The United Kingdom is undertaking a very broad spectrum of research in compression, ranging from model-based compression to fractals. Most research is in universities (for example, University of Essex, Imperial College). Notable industrial institutions include British Telecom and National Transcommunications.

b. France

France also has broad research efforts, but there is a higher concentration of work on higher-resolutior and higher-bit-rate video compression. Major research institutions include CCETT and IRISA. Thomson is the major industrial researcher in the area.

c. Germany

Germany has a higher concentration of work on videotelephony and videoconferencing. The most important university in this regard is the University of Hannover. Industrial research institutions include PKI and Siemens.

d. Italy

Italy has a higher concentration of work in high-quality video and HDTV. Significant research is performed at RAI, Italtel, and Telettra.

e. Benelux

Belgium and The Netherlands produce a surprising amount of research in these areas relative to their size. University centers include the University of Leuven, Université Catholique, and Delft University. Philips research laboratories also generate significant research.

f. Switzerland

Switzerland conducts important research, particularly in boundary- and segmentation-based coding, at EPFL.

g. Scandinavia

The Norwegian Institute of Technology conducts important research in subband coding and motion estimation.

G. PROJECTIONS FOR THE FUTURE

The United States conducts leading research in most areas of image and video compression fundamental techniques. The most significant exception is model-based coding, a technique that is being actively pursued in Japan and Europe, but which will not likely see important application in the next 10 years.

In application-oriented research, however, there is dramatic activity overseas, both in universities and in industrial research laboratories. Research oriented toward generating good implementations of image and video compression standards and the development of high-volume products, such as digital VCRs, digital still cameras, video phone and videoconferencing, and others, is as active abroad as in the United States. As a result, the technology lead implied by fundamental research efforts in the United States is less likely to result in leads in the manufacture and sales of products based on image and video compression technology.

Facsimile compression will be guided primarily by groups that will be able to influence international standards directly. While this will promote interoperability of equipment, it will, and apparently has, stifled innovative research. Thus, during the remainder of this decade, most compression techniques are expected to use standards, except for minor variations in proprietary implementations. Because image and video compression research efforts have centered on different aspects of compression, it seems likely that if facsimile images can be segmented accurately into different types of features, then the most promising solution is to compress each feature separately with the most suitable compression algorithm. Such features as text, line drawings, and gray-scale images could each be compressed with optimum procedures.

H. KEY NON-US RESEARCH PERSONNEL AND FACILITIES

Table VII.1 lists the key non-US research personnel active in image and video compression research, their affiliations, and the image-/video-compression-related areas in which they have been working.

TABLE VII.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—IMAGE AND VIDEO COMPRESSION

Researcher	Affiliation	Area of Expertise
	Australia	
M. H. Chan	Telecom Australia	Motion compensation
D. L. McLaren	University of Tasmania	Packet video
T. Sikora	Monash University	Packet video
	Belgium	
S. D'Agostino J. P. Leduc	Alcatel/Université Catholique de Louvain	Packet video
L. Vandendorpe	Université Catholique de Louvain, Louvain	Progressive transmission
K. Xie	University of Leuven, ESAT, Haverlee	Motion compensation
	Canada	
D. C. Coll Y. Wu	Carleton University	Vector quantization
M. Kamel	University of Waterloo, Waterloo, Ontario	Block truncation codes
	China	
Z.-Y. He L.-N. Wu	Southeast University, Nanjing	Discrete cosine transform
	France	
J. C. Pesquet G. Tziritias	Centre National de Recherché Scientifique (CNRS)	Differential pulse coded modulation
C. Raimondo	IBM–France, La Gaude	Vector quantization
C. Labit N. Tamatoui	IRISA/INRIA, Rennes	Stereo image
N. Demassieux F. Jutland	Thomson	Discrete cosine transform

TABLE VII.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—IMAGE AND VIDEO COMPRESSION (cont'd.)

Researcher	Affiliation	Area of Expertise
	Germany	
S. Pastoor	Heinrich-Hertz Institute, Berlin	Stereo television
J. Achhamer P. C. Jain	Philips	H.261 Discrete cosine transform
P. Pirsch	University of Hannover, Hannover	Motion estimation
	Israel	
N. Efreti	Ben Gurion University, Tel Aviv	Block truncation codes
	Italy	
V. Rampa	Radiotelevisione Italiana (RAI), Milan	Motion estimation
M. Barbero	Tellettra	High-definition television
C. Braccini	Universita de Genova, Genova (Genoa)	Vector quantization
	Japan	
M. Saigusa	Chinon Industries, Inc., Suwa; Yamanashi University, Kofu	Differential pulse coded modulation
M. Konoshima	Fujitsu	MPEG
H. Uwaba	Graphics Communication Technologies (GCT)	H.261
S. Ozawa X. Wang	Keio University, Yokohama	Vector quantization
S. Kadono C. Yamamitsu K. Nobori	Matsushita	High-definition television Digital VCR Model-based

TABLE VII.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES— IMAGE AND VIDEO COMPRESSION (cont'd.)

Researcher	Affiliation	Area of Expertise
	Japan (cont'd.)	
T. Murakami	Mitsubishi Electronic Corp., Kamakura	Digital signal processor (DSP) chips
I. Tamatani	NEC Corp., Kawasaki	DSP chips
M. Nomura Y. Tajiri Y. Suzuki H. Watanabe	Nippon Telephone & Telegraph (NTT)	Packet video H.261 Vector quantization
K. Hosoda	OKI Electric Industry Co. Ltd., Tokyo	Packet video
K. Tasaki	Ricoh	JPEG
K. Ogawa	Sanyo Electric Co. Ltd., Gifu	JPEG
J. Nakabayashi A. Suwa	Sharp Corp., Chiba	H.261 JPEG
H. Yamaguchi	Sony	Stereo image
M. Nakagawa	Toshiba	JPEG
H. Harashima H. Morikawa	University of Tokyo, Tokyo	Model-based
	Norway	
J. H. Husøy	Norwegian Institute of Technology, Trondheim	Subband
	Singapore	
S. N. Koh	Nanyang Technological University, Singapore	Transform

TABLE VII.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—IMAGE AND VIDEO COMPRESSION (cont'd.)

Researcher	Affiliation	Area of Expertise
	South Korea	
M. S. Hong	Samsung	Digital VCR
S.-H. Jang S.-D. Kim	Korea Advanced Institute of Science & Technology (KAIST), Seoul	Motion compensation
D.-H. Kang		Differential pulse coded modulation
	Switzerland	
T. Ebrahimi	Ecole Polytechnique Federale de Lausanne (EPFL), Lausanne	Gabor transform
R. Hervigo M. Kunt		High-definition television Segmentation
	Taiwan	
R. F. Chang	ITRI	Vector quantization
B.-L. Bai J.-F. Yang	National Cheng Kung University	Discrete cosine transform
H.-H. Chen	National Tsing Hua University, Hsinchu	Vector quantization
C.-L. Huang		Segmentation
	The Netherlands	
M. J. T. Reinders	Delft University of Technology, Delft	Model-based
P. H. N. De With F. Sijstermans J. van der Meer	Philips	Digital VCR MPEG

TABLE VII.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—IMAGE AND VIDEO COMPRESSION (cont'd.)

Researcher	Affiliation	Area of Expertise
	United Kingdom	
W. J. Welsh	British Telecom	Model-based
M. J. Biggar A. G. Constantinides	Imperial College, London	Segmentation
G. Knowles A. S. Lewis		Wavelets
P. R. Carmen W. J. Hobsen	National Transcommunications Ltd., Winchester	Motion compensation
S. M. Underwood	National Transcommunications Ltd., Camberley	Systems
D. R. Donovan T. J. M. Rossiter	STC Technology Ltd., Essex	Packet video
M. Bolton	SGS-Thomson Microelectronics, Bristol	Silicon
M. Ghanbari V. Seferidis	Essex University, Colchester	Packet video
A. F. Clark M. Kokuer		Model-based

CHAPTER VII: IMAGE AND VIDEO COMPRESSION REFERENCES

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CHAPTER VIII
RELIABLE COMMUNICATION OF COMPRESSED DATA

A. SUMMARY

This chapter reviews recent non-US contributions to coding theory that may be useful to protect compressed data from channel errors: trellis-coded modulation[1] and algebraic geometric codes. These two research fields originated outside the United States and were chosen because they represent the two most innovative developments of coding theory worldwide during the past decade.

Research in trellis-coded modulation for bandwidth-limited channels started after publication of the work by Gottfried Ungerboeck (1982), a researcher at IBM-Zürich, and has since received considerable attention in Europe, Japan, and the United States. Significant non-US contributions to theoretical and practical aspects of trellis-coded modulation were made by researchers in Western Europe (mainly, Germany, Switzerland, and Italy). The development of trellis-coded modulation had enormous practical implications due to its applicability to high-speed data transmission over bandlimited channels, as the common twisted pair telephone link, in an era when the demand for data exchange between computers is continuously increasing. It is remarkable that, even though coding theory predicted the gains available by combined coding and modulation, no researcher found a practical way to achieve those gains until Ungerboeck's discovery in 1977 (published in 1982). The fact that the discovery was made abroad did not impede its quick assimilation by US researchers and industry, since IBM was the original sponsor of Ungerboeck's research, and AT&T Bell Labs and Codex Corporation (Motorola) rapidly made their own contributions to this field. Several smaller companies, in the United States and abroad, were also able to include trellis-coded modulation in their modem products, since the implementation is not particularly complex. The need for modem standardization led to inclusion of these techniques into CCITT standards. IBM-Zürich is currently still at the forefront of research in high-speed modems. Other contributions from industry come from DLR and AEG in Germany, and Alcatel and Telecom in France. Researchers in Japan (T. Kasami at Osaka University and I. Oka at Osaka

1 E. Biglieri, D. Divsalar, P. J., McLane, and M. K. Simon, *Introduction to Trellis-Coded Modulation with Applications*, New York: MacMillan, 1991.

City University), Italy (Politecnico of Torino and the University of Pisa), Switzerland (ETH Zürich), and Australia (B. Vucetic at the University of Sydney) have made several important theoretical contributions.

Research in algebraic geometric codes, which are currently only of theoretical interest, is based on the seminal work of three researchers in the former Soviet Union, M. A. Tsfasman, S. G. Vladut, and T. Zink (Tsfasman et al., 1982), and received later contributions by researchers in Europe, Japan, India, and China.[2] The original research in the former Soviet Union was done at the Problems of Information Transmission Institute, USSR/Russian Academy of Sciences, and at the Central Economics and Mathematics Institute, both in Moscow. Main contributors in Europe are J. Justesen's group at the University of Denmark and another group at Aarhus University (Denmark), R. Pellikaan at Eindhoven University (The Netherlands), D. Lebrigand at the University of Paris, D. Rotillon at the University of Toulouse, which is also the location of workshops on algebraic coding theory, and M. Perret and G. Lachaud who are members of the French CNRS. Researchers at several Japanese universities have contributed to this field. US researchers did not appear to show much interest in this field until recently, perhaps due to a yet unclear practical advantage of these codes.

Codes for special applications related to compressed data and combined source/channel coding are also addressed in this chapter. Foreign research in these topics is mostly performed in Europe and Japan.

The future of coding research seems to promise more developments in new applications than in theoretical breakthroughs. Progress in non-US applications of coding will be made most noticeably in Japan, Germany, and France. The situation will be similar to the historical collaboration between Sony and Philips that led to the inclusion of Reed-Solomon codes in the compact disk recording technology. Foreign high-definition television (HDTV) research will also be a driver for new developments in coding for compressed data.

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2 S. S. Abhyankar, *Algebraic Geometry for Scientists and Engineers*, American Mathematical Society, 1990.

B. INTRODUCTION

In 1948, Claude Shannon revolutionized communication technology forever with the publication of a classic paper[3] in which he showed that the most efficient solution for reliable communications necessarily involved *coding* the selected message before transmission over the channel. Shannon did not say exactly how this coding should be done; he only proved mathematically that efficient coding schemes must exist. Since 1948, a whole generation of researchers in the United States and abroad has validated Shannon's work by devising explicit and practical coding schemes, which are now part of nearly every modern digital communication system. Satellite communication systems, high-performance military systems, computer communication networks, high-speed modems, and compact-disk recording and playback systems all rely heavily on sophisticated coding schemes to enhance their performance. Mathematicians in the United States and the former Soviet Union have established deep connections between coding and many other areas of mathematics such as combinatorics, lattices, algebraic geometries, and analytic number theory. The latter part of the 1960s and the 1970s was a period of consolidation for the subject as these connections were explored, new applications developed, and new decoding algorithms designed.

Coding traditionally has been divided into two branches, *channel coding* and *source coding*. The goal of channel coding is to protect the transmitted message from the errors and distortions that may be caused by the channel. The goal of source coding (sometimes also called *data compression*), on the other hand, is to ensure that the transmitted message is as dense with information as possible. This division turned out also to have a theoretical value, since it was possible to prove that the source coding problem (representing a source with entropy H by sending information at rate $R > H$) and the channel coding problem (reproducing reliably information sent at rate R on a channel with capacity $C > R$) can be solved separately without loss of performance.

By combining the two main results of information theory, source coding ($R > H$) and channel coding ($R < C$), it is possible to prove that the condition $H < C$ is neces-

3 C. E. Shannon, "A Mathematical Theory of Communication," *Bell Sys. Tech. J.*, 27 (1948), 379–686.

sary and sufficient for sending a source over a channel. If the user can tolerate a certain distortion D, the rate distortion function $R(D)$ represents the amount of channel capacity $C > R(D)$ per source symbol that is both necessary and sufficient for communicating the source to the user with fidelity D.

If, for example, we consider sending digitized speech or images over a discrete memoryless channel, we could design a code to map the speech samples directly into the input of the channel, or we could compress the speech into a more efficient representation, then use an appropriate channel code to send it over the channel. It is not immediately intuitive that we are not losing something by using the two-stage method, since the data compression does not depend on the channel, and the channel coding does not depend on the source distribution.

Information theoretic arguments introduced by Shannon were used to prove that the two-stage method is as good as any other for transmitting information over a noisy channel. This result has an important practical implication: the design of a communication system can be considered a combination of two parts, source coding and channel coding. Source codes can be designed for the most efficient representation of the data independently from the design of channel codes that are suitable for the channel. The overall source channel coding system so designed will be as efficient as any system designed by considering both problems jointly.

Therefore, the source encoder will, independently from the channel encoder, remove redundancy from source sequences, leaving only the most essential information as determined by the fidelity criterion. The channel encoder will insert a suitable form of redundancy designed to provide optimum immunity to channel errors.

The only problem of such a two-stage communication system is that extremely complex equipment is usually required in the encoder and decoder—the joint design of source and channel coding cannot improve the performance but can reduce the implementation complexity.

C. DISCUSSION OF NON-US WORK

During the 1980s, two major developments occurred in coding theory, both originating outside the United States: the development of trellis-coded modulation for bandwidth-efficient transmission, and the generation of new powerful error-correcting codes from curves in algebraic geometries. A brief, intuitive description of these developments is given, with an assessment of their impact.

The transmission of data over existing telephone lines is an important problem, due to the enormous capital investment in developing the present switched telephone network. With the introduction of fiber optic technologies that promise large amounts of relatively cheap bandwidth, this importance will decline, first in the United States, Japan, and Europe, and later in other foreign countries. The capacity of a telephone line has been estimated, using the key results by Shannon, to be between 20 and 30 kbps, using signal-to-noise ratios of 28 dB and higher, and bandwidths between 2,700 and 3,000 Hz.[4] By the early 1970s, first in the United States and later in Japan and Europe, modems became available that transmitted at 9,600 bps, and this was perceived as the practically achievable limit for such channels. The calculation of capacity assumed a simple additive white Gaussian noise channel model, which does not reflect many other impairments present in a telephone channel such as nonlinearities, phase jitter, and time-varying characteristics. Since this calculation was probably optimistic, it seemed that further gains in data rate would be prohibitively expensive and would require the use of low-rate error-correcting codes while the channel was severely bandlimited. It was therefore surprising when Ungerboeck (1982) at IBM-Zürich showed that coding could, in fact, be applied without bandwidth expansion to easily achieve a 3-dB improvement and, with slightly more complex codes, 6 dB. His new combined coding-modulation strategy was based on designing codes directly to maximize Euclidean distance rather than using traditional algebraic codes designed for Hamming distance. He based his results on the two notions of *trellis* representation of codes and of *set partitioning*. To obtain error correction without bandwidth expansion, he proposed expanding the signal set by a factor of 2 and showed, using a capacity argument on conventional one- and two-dimensional signal sets, that most of the available gain could be obtained in this manner. This technique is now referred to as trellis-coded modulation (TCM). For example, an 8-ary phase-shift-keyed system with 16 trellis states has been shown to achieve 4.1-dB coding gain (Ungerboeck, 1982). This combined approach to coding and modulation has led to a revolution in modem design and to

4 G. D. Forney, Jr., R. G. Gallager, G. R. Lang, F. M. Longstaff, and S. U. Qureshi, "Efficient Modulation for Band-Limited Channels," *IEEE J. Sel. Areas Commun., SAC-2,* (1984), 632–647.

an enormous volume of research activity in the United States, Japan, and Western Europe. However, most of the available coding gains were already achieved with Ungerboeck's original designs. Later researchers in the United States and abroad could only slightly improve the performance or construct codes with additional desirable properties (for example, rotational invariance to deal with phase ambiguities). These ideas have already been incorporated into international standards for the design of modems operating at speeds of 9,600 and 14,400 bps (CCITT recommendations V.32 and V.33). The principles of trellis-coded modulation have spread rapidly to other applications, such as satellite and radio channels, and to magnetic recording of data. US modem industries that did not quickly incorporate TCM in their designs suffered substantial loss of market. It is interesting to note that the US modem industry did not suffer erosion or strong competition from Japanese modem manufacturers.

The other non-US development discussed in this chapter involves the construction of a remarkably good class of codes from curves in algebraic geometries originated in the former Soviet Union. This development has a strong theoretical value, but applications are still in the far future. A simple argument, available since the very early stages of coding theory and now called the Varshamov-Gilbert bound, lower bounds the minimum distance d of a linear code of length n and dimension k, and has an equally simple asymptotic version for large n. For many years, no classes of codes that met this bound were known. In fact, there were only negative results on certain classes of codes. For example, it was known that there was no infinite sequence of primitive BCH codes for which both $R=k/n$ and $\delta = d/n$ are bounded away from zero.[5] In 1972, the Danish researcher Justesen[6] gave an interesting construction, using a concatenation scheme, of a class of codes for which δ is greater that zero for large n, but well below the Varshamov-Gilbert bound. Numerous later researchers obtained modest improvements to the construction, but all codes were below the bound. It was not until 1982 that Tsfasman et al. (1982) reported from the Soviet Union a construction based on curves in algebraic geometries that produces codes better than the Varshamov-Gilbert bound.

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5 F. J. MacWilliams and N. J. A. Sloane, *The Theory of Error-Correcting Codes*, Amsterdam: North-Holland, 1983.

6 J. Justesen, "A Class of Constructive Asymptotically Good Algebraic Codes," *IEEE Trans. Inform. Theory*, *IT-18*, (1972), 652–656.

1. Codes for Bandwidth-Limited Channels: Trellis and Lattice Codes

This section examines research outside the United States on the development of error-correcting codes for bandwidth-constrained applications. Such codes are particularly useful when used in conjunction with data compression techniques to achieve overall savings in required data rate for transmission of information.

The need to transmit large amounts of data over a bandlimited channel has led to the development of various data compression schemes, many based on removing redundancy from the data stream. An unwanted side-effect of this approach is to make information transmission more vulnerable to channel noise. Efforts to protect against errors introduced by the channel involve the reinsertion of redundancy and an increase in bandwidth requirements. This section discusses techniques for providing error protection without the additional overhead required for channel coding.

Communication theory provides a firm theoretical background for establishing efficient, reliable, and secure communications systems. Its foundations lie on three important basic developments by US and Russian researchers, which can be referred to as Nyquist theory, Wiener-Kotelnikov theory, and Shannon theory. Nyquist's theory aims at controlling distortion in the transmission of bandlimited pulse sequences. The most brilliant results of this theory, obtained in 1928, established the criteria for distortionless transmission of such sequences.[7] In the early 1940s, the structure of optimum receivers in the presence of noise under the minimum mean-square error criterion was determined by Wiener, and, in the late 1940s, Kotelnikov studied optimum demodulation techniques. The investigation of optimum communication system design had a new beginning when Shannon published his classic paper.[8] Shannon's random-coding arguments established theoretical bounds on the achievable performance of communication systems and launched the field of error control coding. Large research efforts on error control coding in the United States and abroad have resulted from the pioneering work of Shannon. The first compre-

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7 H. Nyquist, "Certain Topics in Telegraph Transmission Theory," *Trans. AIEE*, *47*, (Apr 1928), 617–644.

8 Shannon, 1948, op. cit., 379–343.

hensive treatment of digital communication theory was written in the United States by Wozencraft and Jacobs.[9]

Since the late 1950s, the three theories mentioned above have gradually merged and produced new developments in sophisticated signal designs (constellations). Since the early 1960s, various types of multidimensional signal constellations have been proposed by many US and foreign researchers to more closely approach Shannon's bound. These multidimensional modulation formats were initially impractical due to the presence of severe intersymbol interference. Again, US researchers investigated the design of optimum receivers in the presence of intersymbol interference and found systematic methods for spectrum shaping, also providing the first digital modulation system (partial response continuous-phase frequency-shift keying) that conserved bandwidth. In 1967, another US researcher derived a bound on the performance of convolutional codes using an asymptotically optimum decoding algorithm.[10] This algorithm later became known as the Viterbi algorithm. In the early 1970s, Forney[11] gave a complete account of the importance of the Viterbi algorithm, particularly for the intersymbol interference channel.

Various efforts to improve the signal-to-noise ratio in channels with additive noise and intersymbol interference led to the original development of set partitioning techniques by Imai and Hirakawa in Japan, and culminated in Ungerboeck's development, in the late 1970s, of trellis-coded modulation (TCM) at IBM-Zürich (Ungerboeck, 1982). These are the most important contributions outside the United States to the early development of TCM because they revealed for the first time a practical method to achieve most of the remaining coding gain that Shannon's theory predicted for bandlimited channels. The initial motivation for developing TCM came from work on use of the Viterbi algorithm to improve the signal-to-noise ratio in intersymbol interference channels, where improvements for multilevel signal constellations require an increase in the Euclidean distance between signal points. The combined coding and modulation approach of TCM increases the Euclidean distance by encoding the data sequences themselves.

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9 J. M. Wozencraft and I. M. Jacobs, *Principles of Communication Engineering*, Wiley, 1955.

10 A. J. Viterbi, "Error Bounds for Convolutional Codes and an Asymptotically Optimum Decoding Algorithm," *IEEE Trans. Inform. Theory, IT-13*, (1967), 260–269.

11 G. D. Forney, Jr., "Maximum-Likelihood Sequence Estimation of Digital Sequences in Presence of Intersymbol Interference," *IEEE Trans. Inform. Theory, IT-18*(1972), 363–378.

Ungerboeck showed how the Hamming distance between differing data sequences does not necessarily imply large Euclidean distance between modulated data sequences, unless the assignment of coded signals to modulated signals is cleverly made. He proposed a new method of set partitioning that not only provided a method for the assignment of coded signals to channel signals, but also provided a simple formula for the lower bound of the Euclidean distance between modulated data sequences. In recent years, an enormous amount of work has been reported on coded modulation in the United States and abroad.

Other important US contributions were made by Forney and Calderbank and Mazo. Forney gave a detailed account of the advances made during the 1980s in approaching Shannon's capacity bound for the additive Gaussian noise channel, focusing on precoding combined with trellis coding and spectral shaping to approach channel capacity. Forney[12] studied the properties of lattices (also used for vector quantization, as discussed in Chapter IV) to construct powerful trellis codes. Forney[13] also recently introduced the use of isometries to design trellis codes with practically important properties as rotational invariance. His work on isometries[14] has been recently applied by Benedetto et al. (1993) at the Politecnico of Torino, Italy, to construct several good trellis codes. Calderbank and Mazo[15] looked at codes on large alphabets, where the encoding rule can be used to spectrally shape the transmitted signal. Such encoding techniques would be highly applicable in such areas as magnetic recording, high-data-rate digital subscriber lines, and high-definition television.

There are an enormous number of non-US contributions on trellis-coded modulation. We concentrate on the most important from a theoretical or practical point of view.

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12 G. D. Forney, "Coset Codes 2: Binary Lattices and Related Codes," *IEEE Trans. Inform. Theory*, *34*, 5(1988), 1152–1187.

13 G. D. Forney, "Geometrically Uniform Codes," *IEEE Trans. Inform. Theory*, *IT-37*, (Sept 1991), 1241–1260.

14 Ibid.

15 A. R. Calderbank and J. E. Mazo, "Spectral Nulls and Coding with Large Alphabets," *IEEE Commun. Mag.*, *29*, 12(1991), 58–67.

Imai and Hirakawa (1977), in Japan, and Ginzburg (1984), in the former Soviet Union, showed how algebraic codes of increasing Hamming distance can be combined with nested signal constellations of decreasing Euclidean distance to generate multidimensional signals with large distances. Cusack (1984) and Sayegh (1986) found specific constructions of multidimensional constellations. Based on these earlier contributions, Biglieri (1992), at the Politecnico of Torino, Italy, suggested staged demodulation of these multidimensional constellations. With this procedure, the bits of the signal label protected by the most powerful code are decoded first, then the bits protected by the second most powerful code are decoded, and so on. This procedure is suboptimum, but reduced decoding complexity will result. In particular, staged decoding is amenable to an architecture with pipelined parallelism. Biglieri considered the design of constellations that allow a high degree of parallelism in the staged decoder structure and allow soft decoding of the component algebraic codes based on a systolic algorithm amenable to VLSI implementation. Other contributions from Italian researchers include a study on phase ambiguity resolution by Mengali et al. (1990). Mengali is an expert in synchronization problems.

In Australia, B. Vucetic has made a number of recent contributions to the analysis of trellis codes. He studied the performance of 8-PSK trellis codes over nonlinear fading mobile satellite channels, deriving a new analytical bound for large effective code length (ECL) codes, defining a trellis-code design procedure for fading channels, and estimating the performance degradation of nonlinear communication systems with pilot tone (Vucetic and Nicolas, 1992). He compared the performance of block-coded modulation (BCM) coding schemes for fading channels to that of trellis-coded modulation (TCM—Vucetic and Linn, 1991). He evaluated the performance of 16-QAM TCM schemes with ideal channel state information (CSI) on Rayleigh fading channels (Du and Vucetic, 1991). Vucetic also contributed to the analysis of 16-QAM (Vucetic and Du, 1992) and 16-PSK (Vucetic and Zhang, 1990) trellis codes for fading channels, and analysis of M-QAM coding schemes for digital radio (Vucetic et al., 1990). His work is comparable in quality to similar work done in the United States for mobile communications and contributes several new analytical results.

A similar problem of mobile communications has been addressed in the work on the application of variable rate codes (punctured convolutional codes) by Hagenauer et al. (1990) in Germany. These codes were first developed in the United States, and applications are now being explored in Europe and Japan. In Germany, an active area of interest and research is the design of high-speed Viterbi decoders necessary to decode trellis codes (Fettweis and Meyr, 1989). Chevillat and Eleftheriou (1989a), at IBM-Zürich, have made interesting contributions to the decoding problem of trellis codes in the presence of intersymbol interference and noise. Chevillat and Eleftheriou (1989b) also made important contributions to the design of adaptive receivers for data transmission over dispersive band-limited channels, using concepts from lattice codes.

Schnabl and Bossert (1991), in Germany, have considered coded modulation with multiple concatenation of block codes and claim advantages in performance/complexity tradeoff. A similar concept based on the combination of block-coded modulation and trellis-coded modulation has been proposed by Wu and Su (1991). This work also uses properties of lattices to partition the trellis code. Important non-US work on lattices has been done in Israel by Be'ery et al. (1989) on the fast decoding of the Leech lattice.

An active research group at MacMaster University in Canada (Taylor and So, 1992; Yang et al., 1990) is making progress over work in the United States on the design and performance analysis of continuous phase frequency shift keying (CPFSK) and multilevel-coded modulation. Another interesting contribution from Canada is the application of trellis-coded modulation to the indoor wireless channel (Paiement and Chouinard, 1991).

The only recent contribution from the United Kingdom is a study of land mobile satellite links by Fines and Aghvami (1992), which compares low-bit-rate DE-QPSK (differentially encoded phase shift keying), TCM 8-ary PSK and 16-ary DE-QAM (differentially encoded quadrature amplitude modulation) fully digital demodulators.

The Swiss Federal Institute of Technology (ETH) in Zürich is an active center of high-quality theoretical research in coding, due in large part to the work of J. Massey, who emigrated from the United States to Switzerland in the early 1980s. Particularly innovative is the recent work by Loeliger (1991) on the design of signal sets matched to generalized linear algebraic codes, building on fundamental results on group codes obtained by Slepian in the United States in the 1960s. Purely theoretic problems on minimal and non-catastrophic encoders for convolutional codes over rings were addressed by Mittelholzer (and Loeliger, 1993) also at ETH–Zürich. Beside research at ETH and IBM, researchers at Swiss industrial establishments have produced significant contributions: Ramseier (1990, 1991) studied bandwidth-efficient trellis-coded modulation schemes for commercial applications; Wittenben (1990) studied TCM schemes for time-selective fading channels; and Schlegel and Costello (1989) analyzed TCM schemes on channels disturbed by jamming and impulse noise. In summary, research in Switzerland has produced several theoretical and practical improvements over US research on trellis codes.

Despite the fact that French researchers are not as active in trellis-coded modulation as their Italian, German, and Swiss counterparts, some isolated, minor contributions from France are worth mentioning. The results of work by Chouly and Sari (1988) on the design of a family of six-dimensional (6-D) trellis-coded modulation schemes indicate that 6-D trellis codes achieve a better performance/complexity tradeoff than 8-D codes, and some of them also achieve a significantly better tradeoff than 4-D codes. Research by Karam and Sari (1991) extends an idea generated in the United States on data predistortion to combat transmitter nonlinearities and suggests the use of predistortion rules with memory by using intersymbol interpolation.

The only notable, albeit minor, contribution from Austria is work by Zou and Weinrichter (1989) on simplified designs for trellis-coded modulation and useful new lower bounds on their distance.

Scattered and minor contributions have been made by researchers in Asia. Kam and Ching (1990, 1992) resurrected the use of the Viterbi algorithm for sequence estimation over fading channels with diversity reception. Wang et al. (1990) studied trellis codes for partial response, a subject earlier developed better in the United States and at IBM-Zürich. Wen et al. (1990) studied a modification of the Viterbi decoding that was already known in the United States.

2. Algebraic Geometric Codes

This section examines research outside the United States on the development of a class of error-correcting codes called algebraic geometric codes. Another major development of coding theory during the 1980s, discovered by a group of Russian mathematicians, was the construction of a remarkably good class of codes, called "algebraic geometric codes" from curves in algebraic geometries. These codes have good minimum distance properties, that is, they can correct more errors for a given block length and code size.

Tsfasman et al. (1982) showed that by using algebraic curves, one can construct codes that lie above the Varshamov-Gilbert bound and that can correct more errors than other previously known block codes. This is the most important non-US contribution to the discovery of a new class of powerful error-correcting codes.

The idea of the construction is best explained as a generalization of Reed-Solomon codes, as discussed by van Lint and Springer.[16] Consider a finite field F_q with q elements and the Reed-Solomon code of length q, dimension k and minimum distance d, defined over F_q. Then algebraic geometric codes can be described as a generalization of such Reed-Solomon codes based on a certain projective line when this line is replaced by a projective curve. Each such curve is associated with an integer g called the genus of the curve.

It can be shown that (n,k) codes can be constructed with dimension $k = m - g + 1$ and minimum distance $d \geq n - m$, where m is a positive integer such that $2g - 2 < m < n$. If the codes are based on the projective line, which has genus 0, the Reed-Solomon codes are obtained. The main result is a sequence of projective curves such that $R = k/n \geq 1 - \gamma - \delta$, where $\gamma = \frac{1}{q^{1/2} - 1}$ and $\delta = d/n$. It is possible to show that, for $q \geq 49$, δ meets the Varshamov-Gilbert bound and between the points of intersection one gets a sequence of codes that exceed the bound (Tsfasman, 1991). This is an application of an area of pure mathematics, a particular strength of Russian research, to the solution of a difficult and long-standing problem of coding theory. It is likely that more work will lower the restriction $q \geq 49$, and it may be anticipated that binary codes meeting the Varshamov-Gilbert bound will be found.

Because of the relative youth of this area of research, several of the basic properties of the codes described above can be considered to be recent non-US research

16 J. H. van Lint and T. A. Springer, "Generalized Reed-Solomon Codes from Algebraic Geometry," *IEEE Trans. Inform. Theory*, *IT-33*, (1987), 308–309.

activity, especially the construction described by van Lint and Springer (1987). A related problem is the decoding of these algebraic geometry codes. Practical decoding algorithms exist, although none of these decode up to the full error-correcting capacity of the code. Efforts in devising decoding algorithms are discussed by Justesen et al. (1989); Skorobogatov and Vladut (1990); Kurihara et al. (1991); Vladut (1990); Pellikaan (1989, 1992); Justesen et al. (1992); and Rotillon and Ly (1991). Extending these algorithms to the full error-correcting capacity of the code will be the main challenge.

After the discovery of algebraic geometric codes, a number of non-US contributions to this field were made, and all of them have been significant improvements over related US research, which is somewhat scarce. A summary of the major developments follows.

Janwa (1991) used Weierstrass gaps of points to improve the lower bounds on the minimum distance and covering radius of Goppa codes and Reed-Solomon (RS) codes from function fields over finite fields. As a consequence, he showed the existence of many optimal and sub-optimal codes from algebraic geometry. Chaoping (1992) generalized a result on the minimum distance of Goppa codes to alternate codes. Tsfasman (1991), an inventor of these codes, analyzed the present knowledge on asymptotic bounds in coding theory and discussed the advantages of an algebraic geometric approach to code design. Skorobogatov (1991) proposed a uniform approach to BCH codes, Goppa codes, and subfield subcodes of algebraic geometric codes on curves of arbitrary genus based on exponential sums along a curve. He generalized some classical and recent results on the dimension, the minimum distance, the covering radius, and the spectrum of long BCH codes.

Several important topics related to algebraic geometric codes were covered at the 1989 Toulouse conference on "Applied Algebra, Algebraic Algorithms, and Error-Correcting Codes" (AAAECC, 1989), including results on subcodes of algebraic geometric codes and on asymptotic problems concerning algebraic geometric codes.

Vladut (1987) constructed a new lower bound for the asymptotic parameters of codes arising from modular curves. For $q = 4, 9, 16, 25$, it is identical to the Varshamov-Gilbert bound, whereas for $q = p^{2a} \geq 49$, it improves the best known lower bound in two ranges of δ. Vladut (1990) proved that for algebraic geometric codes on a curve over F_q for $q \geq 37$ or on a curve of sufficiently large genus over F_q for $q \geq 16$, there is a polynomial decoding algorithm capable of correcting up to $(d^* - 1)/2$ errors, d^* being the designed minimum distance. Pellikaan et al. (1991) showed that all linear codes can be obtained from an infinite series of curves, using Goppa's construction. Criteria were derived for linear codes to be algebraic geometric. In particular, the family of q-ary Hamming codes was investigated, and it was proven that only those with redundancy one or two and the binary (7,4,3) code are algebraic geometric in this sense. Hansen and Stichtenoth (1990) constructed a series of algebraic geometric codes using a class of curves that have many rational points. They obtained codes of length q^2 over F_q, where $q = 2q_0^2$ and $q_0 = 2^n$, such that dimension + minimal distance $\geq q^2 + 1 - q_0/(q-1)$. Barg et al. (1987) constructed q-ary codes from algebraic curves of genus 1, 2, and 3. From these codes binary codes were obtained with parameters better than those of known codes.

In the theory of algebraic geometric codes, the length is bounded by the number of rational points of an algebraic curve chosen as the base curve of the code under consideration. Significant progress on this specific topic has been reported by Japanese researchers. Mizuno et al. (1988) showed how information about the number of its rational points can be obtained from the Zeta function associated with the base curve. Mizuno and Ando (1989) constructed new algebraic geometric codes arising from algebraic curves $X_s{:}x^s y + y^s z + z^s x = 0$ defined over $F = GF(2^{31})$. These codes have "good" parameters for certain values of l and $s = 2^l + 1$. Kurihara et al. (1991) gave explicit constructions of parity check matrices and generator matrices of algebraic geometric codes. They also presented a method of decoding these codes. The only other notable contribution from Japan is by Sakata (1991).

Progress in the determination of the weight enumerator of algebraic geometric codes was reported by Katsman and Tsfasman (1987), who derived bounds on the enumerator of an arbitrary algebraic geometric code. They calculated in full the weight distribution of the code constructed from all the points of an elliptical curve. The weight distribution (the enumerator) is found to depend on the group of points of this curve and on an element of this group. The number of minimum-weight vectors is minimized both by elements and by curves. The possibility that such a code is MDS (maximum distance separable) was explored. Sorensen (1992) showed how the minimum distance of certain algebraic geometric codes, in many cases, can be determined exactly using subcodes that are weighted Reed-Muller codes. In France, Lachaud (1992) wrote a survey of some results recently obtained on the distribution of the weights of some linear codes that depend upon the properties of certain algebraic curves defined over a finite field, and Perret (1991) derived a specific algebraic geometric code and its parameters.

The interesting theoretical and practical work on finding efficient decoding algorithms for these codes started with the results reported by Justesen, one of the pioneers in the conquest of the Varshamov-Gilbert bound, and his Danish colleagues K. Larsen, H. Jensen, A. Havemose, and T. Hoholdt (Justesen et al., 1989). This decoding method is limited to a certain subclass of algebraic geometric codes and is a generalization of a decoding method for BCH codes found by Peterson in the United States several years earlier. The next contribution to a decoding algorithm was made by Pellikaan (1989), who considered an algorithm given by Skorobogatov and Vladut and gave a modified version, with improved performance, which he obtained by applying the former algorithm a number of times in parallel. He generalized existing decoding algorithms based on error location for BCH and algebraic geometric codes to arbitrary linear codes and investigated the number of dependent sets of error positions (Pellikaan, 1992). A received word with an independent set of error positions can be corrected. Following Pellikaan's 1989 algorithm that decodes geometric codes up to $t^* = d^* - \frac{1}{2}$ errors, where d^* is the designed distance of the code, Rotillon and Ly (1991) described an effective decoding procedure for some geometric codes on the Klein quartic. Lebrigand (1991) also commented on 1989 Pellikaan's algorithm and presented facts about the Jacobian of a hyperelliptic curve that makes that algorithm practical. In the former Soviet Union, Skorobogatov and Vladut (1990) presented a decoding algorithm for algebraic geometric codes arising from arbitrary algebraic curves. This algorithm corrects any number of errors up to $(d - g - 1)/2$, where d is the designed distance of the code and g is the genus of the curve. The complexity of decoding is $\sigma = n^3$ where n is the length of the code. Also presented was a modification of this algorithm, which in the case of elliptic and hyperelliptic curves is able to correct $(d - 1)/2$ errors. It is shown that for some codes based on plane curves the modified decoding algorithm corrects approximately $d/2 - g/4$ errors. Asymptotically good q-ary codes with a polynomial construction and a polynomial decoding algorithm (for $q \geq 361$ their parameters are better than the Varshamov-Gilbert bound) are obtained. A family of asymptotically good binary codes with polynomial construction and polynomial decoding is also obtained, whose parameters are better than the Blokh-Zyablov bound on the whole interval $0 < \sigma < \frac{1}{2}$. Finally, Justesen made some improvements to an earlier decoding algorithm (Justesen et al., 1992). For codes from an arbitrary regular plane curve, this new algorithm corrects up to $d^*/2 - m^2/8 + m/4 - 9/8$ errors, where d^* is the designed distance of the code and m is the degree of the curve. The complexity of finding the error locator is also discussed.

3. Special Applications, Combined Source/Channel Coding, and Implementations

This section examines non-US research on special applications of coding theory to the protection of compressed data and on the combined optimization of source and channel codes. Emerging implementations and standards are also discussed.

Recent years have witnessed significant progress in the compression of speech, audio, image, and video signals. This has been the result of simultaneous application of insights from coding theory, signal processing, and psychophysics. Further improvements are expected from increased interaction of source coding technology with other communications disciplines, such as channel coding and networking.[17] Early non-US contributions on integrated data communication systems with data compression and error-correcting codes were reported by Benelli et al. (1978). Early US work on joint source and channel coding by Gray et al.[18] was more recently followed by developments[19] based on a concept analogous to Ungerboeck's coded modulation idea discussed in the section on trellis codes. Vaishampayan and Farvardin[20] addressed the problem of designing a bandwidth-efficient, average-power-limited digital communication system for transmitting information from a source with known statistics over a noisy waveform channel with the goal of designing an encoder, decoder, and modulation signal set to minimize the mean squared error (mse) between the source vector and its estimate in the receiver. Significant

17 N. Jayant, "Signal Compression—Technology Targets and Research Directions," *IEEE J. Sel. Areas Commun.*, *10*, 5(1992), 796–818.

18 E. Ayanoglu and R. M. Gray, "The Design of Joint Source and Channel Trellis Waveform Coders," *IEEE Trans. Inform. Theory*, *33*, 6(1987), 855–865.

19 T. R. Fischer and M. W. Marcellin, "Joint Trellis-Coded Quantization Modulation," *IEEE Trans. Commun.*, *39*, 2(1991), 172–176.

20 V. A. Vaishampayan and N. Farvardin, "Joint Design of Block Source Codes and Modulation Signal Sets," *IEEE Trans. Inform. Theory*, *38*, 4(1992), 1230–1248.

performance improvements over the standard VQ-based system are demonstrated when the channel is noisy.

a. Unequal Error Protection (UEP) Codes

A recent standard for image coding (JPEG) is based on the discrete cosine transform (DCT), followed by variable-length coding (VLC). Transmitted information becomes more vulnerable to channel errors after VLC. An efficient source channel coding scheme for this situation is based on the use of unequal error protection (UEP) codes.

In France, Fazel and Lhuillier have investigated the transmission of images over bursty and random channels (1990a) and the application of UEP codes (1990b). Their work is the most important non-US contribution to this field. Their channel encoder is based on the UEP principle and implemented as a concatenated code. To measure the importance of each bit in the transmitted frame after VLC, a factor of sensitivity for each bit-to-channel error is defined. Then, by utilizing this factor, the optimal error rate for each bit that minimizes the effects of channel noise is estimated. This work showed that this concatenated scheme offers higher performance and less complexity than a single UEP code.

Due to increased radio spectral congestion, the trend in future cellular mobile radio systems is toward digital transmission. Recent advances in spectrally efficient modulation techniques and high-quality low-bit-rate speech coding have further aided this move. However, mobile radio channels are subject to signal fading and interference that cause significant transmission errors. The design of speech and channel coding for this application is therefore challenging. Hagenauer, in collaboration with researchers at AT&T Bell Labs, has reported very useful results for mobile radio channel applications of subband speech coding and matched convolutional channel coding (Cox et al., 1991). Among the results, analyses and informal listening tests show that with a four-level UEP scheme, transmission of 12-kb/s speech is possible with very little degradation in quality over a 16-kb/s channel with an average bit error rate of 2×10^{-2} at a vehicle speed of 60 mph and with interleaving over two 16-ms speech frames.

Other minor progress in the application of UEP codes has been reported by Hagenauer (1989) and Mabogunje and Farrell (1991). Englund (1991) has evaluated the parameters of some binary nonlinear two-level UEP codes. In some situations the nonlinear codes prove to be better than linear codes. Chen et al. (1990) reported new results on self-orthogonal UEP codes, and Stevens (1990) has given decoding algorithms for UEP product codes. Jun (1991) presented a systematic procedure for the construction of a class of binary self-orthogonal convolutional codes (SOCC). Unequal error protection SOCCs are then obtained through a minor modification in the parity check matrix.

b. Combined Source and Channel Coding

From both theoretical and practical viewpoints, one of the most basic and important problems in the digital communication system is to minimize the distortion between the output signal from the information source and the destination signal on the receiving side, under the condition that the transmission rate in the channel is fixed. Several non-US contributions to this problem are discussed below.

Fukunaga et al. (1988, 1990) presented a detailed investigation of the tradeoff relation between the vector quantization (source coding) and the error-correcting coding (channel coding), where the transmission rate of the channel is fixed as constant. Assuming a memoryless Gaussian source or an actual speech signal source, the average distortion for various channel error probabilities is discussed. This is a comprehensive study, but it does not add significantly new knowledge to the problem.

Rosenbrock and Besslich (1992) proposed a new design procedure for shape-gain vector quantizers (SGVQ), which leads to substantially improved robustness against channel errors without increasing the computational complexity. This is achieved by including the channel transition probabilities in the design procedure, leading to an improved assignment of binary codewords to the coding regions. The new algorithm is particularly useful for heavily distorted or fading channels.

Improvements in vector trellis quantization for noisy channel applications in which the application of maximum a posteriori (MAP) detection to trellis quantizers operating over additive white Gaussian noise (AWGN) channels is introduced were reported by Soleymani and Khandani (1991) and Soleymani and Nassar (1992). By using MAP detection instead of maximum likelihood, gains as high as 0.57 dB and 2.2 dB can be achieved in terms of signal-to-quantization-noise ratio (SQNR) for Gauss-Markov source and speech samples, respectively. Another application of combined source channel coding to image transmission has been addressed by researchers working for the German aerospace establishment, where tree-structured vector quantization and sequential decoding has been considered (Perkins and Offer, 1991). Another German researcher has presented some interesting techniques for combined source channel coding in adaptive transform coding systems for images (Goetze, 1984). Earlier work by Goetze was reported by Morgera et al. (1989). A theoretical work on the asymptotic convergence of dual-tree entropy codes appears in a paper by Freeman (1991).

Specific advances in joint source/channel coding for speech transmission were obtained by Wong et al. (1990), who considered combined source and channel coding of subband coded speech with a post-enhancement method. Wong et al. (1991) also reported progress in the estimation of unreliable packets in subband coding of speech. They examined a system using 12 kb/s embedded subband coding of speech and rate-compatible-punctured-convolutional (RCPC) codes with generalized Viterbi decoding for combined error detection and correction. System performance was improved (4.4-dB SNR improvement on speech) by exploiting the frame-to-frame redundancy in the subband energy profile used for adaptive bit allocation in the subband coder. The improved error detection capability in the subband side information allows a suitable reassignment of channel error protection bits to the subband main information, leading to an overall improvement in system performance. These improvements are significant, but probably comparable to techniques already in use in the United States

Japanese researchers have not been very active in the theory of combined source/channel coding. One notable exception is the method proposed by Yamane et al. (1989) for rate allocation to bit planes.

Given the ubiquitous application of Huffman codes, the contribution by Honary et al. (1990) on combined Huffman and convolutional codes is worth mentioning.

Applications of combined source/channel coding to high-definition television (HDTV) transmission have been reported by researchers working for the Italian TV National establishment (RAI—Barbero et al., 1990). Frame replenishment coding was proposed by Zhang and Goldberg (1991).

A very specific source channel coding problem for raster document transmission over mobile radio was addressed by Wyrwas and Farrell (1989) in the United Kingdom.

c. Standards and Implementations

Standards and implementations using error correction codes to protect compressed data are driven by design requirements of efficient HDTV broadcast systems and of high-speed modems. For example, General Instrument's DigiCipher System[21] is an all-digital HDTV system implementation (developed in the United States) that can be transmitted over a single 6-MHz VHF or UHF channel. It provides full HDTV performance, with virtually no visible transmission impairments due to noise, multipath, or interference, thanks to powerful error correction coding combined with adaptive equalization. Recent (US-developed) modem implementations[22] incorporate very advanced signal processing techniques to provide high-speed duplex data transmission over two-wire telephone lines. They integrate protocols for error detection and correction, data compression, and remote configuration.

Non-US contributions to coding for compressed data in modem design were considered by Haspeslagh and Sansen (1990), where a design for a 9,600-b/s modem transmitter chip (Alcatel, Belgium) was discussed. This study does not add to the US state of the art in modem design.

In Japan, Kumozaki (1991), at NTT, described an error correction system for digital subscriber loop transmission systems that uses time compression multiplexing. The experimental results for a 200-kb/s system show that burst errors are substantially reduced. Yamazato et al. (1992) proposed a so-called Interlace Coding

21 W. Paik, "DigiCipher—All Digital, Channel Compatible, HDTV Broadcast System," *IEEE Trans. Broadcasting, 36,* 4(1990), 245–254.

22 G. Baudoin and M. S. Mitrani, "The Telsat Range of V32 Modems," *Commutation & Transmission, 12,* 4(1990), 43–56.

System (ICS) involving a data compression code, a data encryption code and an error-correcting code. The proposed system handles data compression, data encryption and error-correcting processes together, that is, adds error-correcting redundancy to compressed (Lempel-Ziv scheme) and encrypted (Data Encryption Standard, DES) words. This is an important new proposal for a completely integrated system. Japanese researchers reported the development of a high-performance single-chip LSI for error correction of optical disk memory (Murai et al., 1989).

Specific, minor advances in implementations using joint source/channel coding for speech transmission were obtained in Hungary by Hanzo (1989). Hanzo et al. (1991) more recently reported new results on the transmission of digitally encoded speech at 1.2-Kbaud for mobile personal communication networks (PCNs). Speech was encoded at 4.8 kb/s using a low-complexity transformed binary pulse-excited LPC Codec. A 64-level QAM modem with three sub-channels that operated with different bit error rates (BERs) was used. The sensitivity of the encoded speech bits to transmission errors was identified, and the bits classified into three groups. Each group was then individually coded by BCH coders with differing power. The output of the BCH coders was gray-coded onto the three QAM channels. By this arrangement, the protection given to the speech bits was dependent on their vulnerability.

Since the decoding of convolutional and trellis codes can be efficiently done by using the Viterbi algorithm, implementations of Viterbi decoders are interesting. Bree et al. (1992) reported on a node-parallel Viterbi decoding architecture with bit-serial processing and communication. An important aspect of this structure is that short-constraint-length decoders may be interconnected, without loss of throughput, to implement a Viterbi decoder of larger constraint length. A proof-of-concept chip was developed using the Queen's University Interactive Silicon Compiler and fabricated by Northern Telecom in association with the Canadian Microelectronics Corporation (CMC). This study seems to ignore more advanced US work in the design of Viterbi decoders. The decoding of Reed-Solomon codes is an equally important practical issue. However, the work reported by Koksal and Yucel (1992) does not seem to add to US-based very advanced knowledge on this subject.

D. PROJECTIONS FOR THE FUTURE

Although US research in coding theory has been strong and productive, the two major developments during the 1980s originated abroad, in Switzerland and Russia. Since both developments were unexpected at the least, if not fortuitous, it is not advisable to attempt to predict future developments. What was not unexpected was that the most theoretical development occurred within the strong school of information and coding theorists in Russia, and the more practical development occurred in a research laboratory of a large US company (IBM), albeit based abroad. The most active area of research in trellis-coded modulation will probably be the construction of trellis codes with the same gain of those already known, but with additional useful properties (for example, rotational invariance, ease of synchronization, variable rate applications). In this respect, the use of isometries will be the most promising tool for such constructions. These methods have been recently introduced in the United States and are pursued actively in Western Europe—and in Italy, in particular. The most challenging directions of research in algebraic geometric codes will be to find fast decoding algorithms that decode to the full error-correcting capability of the codes, to find better and less complex derivations of the codes, and possibly to find similar binary codes that exceed the Varshamov-Gilbert bound. The likelihood for success in this field currently is centered in Denmark, France, and Russia.

Error-correcting codes are finding increasing application in a broad range of systems developed in the United States, Europe, and Japan, for example, hard disks for computer storage and audio compact discs. The use of error-correcting codes in hard disk systems can render some of the defective sectors usable; the tradeoff between the redundancy of the code, the cost of implementing error correction, and the fraction of the disk restored to use can be quite advantageous. Similarly, compact disk technology, pioneered by Sony in Japan and Philips in Europe, contains very sophisticated signal processing that includes Reed-Solomon coding and interleaving to protect against burst errors caused by surface scratches. In addition, error correction techniques are being applied, more widely in the United States than in Japan, to memory chips to overcome defects and improve process yield.

When new applications for coding are found, the codes need to be modified to take into account special constraints. This is a situation in which countries having large volumes of data exchange and processing, notably the United States, will generate the need for new research. In this respect, Russian research will probably remain more theoretical in nature. As much as applications can be the driver for new developments, advances in microelectronics will also be a major influence on the design of future coding and decoding algorithms. In this respect, Japan and the United States will probably take the lead. In general, error correction algorithms will become more and more an integral part of an overall signal processing scheme rather than a separate and specialized component.

E. KEY NON-US RESEARCH PERSONNEL AND FACILITIES

Table VIII.I lists the key non-US research personnel active in reliable communication of compressed data, their affiliations, and the research relevant to this assessment in which they are engaged.

TABLE VIII.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—RELIABLE COMMUNICATION OF COMPRESSED DATA

Researcher	Affiliation	Area of Expertise
	Australia	
B. Vucetic	School of Electrical Engineering, University of Sydney, Sydney	Codes for bandwidth-limited channels
	China	
Xing Chaoping	Department of Mathematics, University of Science & Technology of China, Hefei	Algebraic geometry codes
	Denmark	
A. Havemose T. Hoholdt H. E. Jensen J. Justesen K. J. Larsen	Circuit Theory & Telecommunications Institute, Technical University of Denmark, Lyngby	Algebraic geometry codes
J. P. Hansen H. Stichtenoth A. B. Sorensen	Mathematics Institute, Aarhus University, Aarhus	Algebraic geometry codes

TABLE VIII.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—
RELIABLE COMMUNICATION OF COMPRESSED DATA (cont'd.)

Researcher	Affiliation	Area of Expertise
	France	
G. Lachaud M. Perret	Laboratoire de Mathematique Discrete du CNRS, Luminy, Marseilles	Algebraic geometry codes
G. Battail J.-C. Belfiore	TELECOM, Paris	Codes for bandwidth-limited channels
H. Sari	SAT, Paris	Codes for bandwidth-limited channels
D. Lebrigand	University of Paris, Paris	Algebraic geometry codes
D. Rotillon	University of Toulouse, Toulouse	Algebraic geometry codes
	Germany	
M. Bossert	AEG Mobile Communications GmbH, Ulm	Codes for bandwidth-limited channels
J. Hagenauer P. Hoeher	German Aerospace Research Establishment (Deutsche Forschungs und Versuchsanstalt Luft und Raumfahrt/ DLR), Institute of Communication Technology, Oberpfaffenhofen	Codes for bandwidth-limited channels
H. Meyr	Polytechnic College (Technische Hochschule), Aachen University, Aachen	Codes for bandwidth-limited channels
	India	
H. Janwa	School of Mathematics, Tata Institute of Fundamental Research, Bombay	Algebraic geometry codes
	Italy	
A. Spalvieri	Alcatel–Telettra, Milan	Codes for bandwidth-limited channels
S. Benedetto E. Biglieri	Turin Polytechnic Institute, Turin (Politecnico di Torino, Torino)	Codes for bandwidth-limited channels
U. Mengali	Electronics & Telecommunications Institute, Pisa	Codes for bandwidth-limited channels
M. Luise	University of Pisa, Pisa	Codes for bandwidth-limited channels

TABLE VIII.1
KEY NON-US RESEARCH PERSONNEL AND FACILITIES—
RELIABLE COMMUNICATION OF COMPRESSED DATA (cont'd.)

Researcher	Affiliation	Area of Expertise
	Japan	
S. Sakata	Department of Knowledge-Based Information Engineering, Toyohashi University of Technology, Toyohashi	Algebraic geometry codes
H. Imai	Toyko University, Tokyo	Codes for bandwidth-limited channels
T. Kasami	Osaka University, Osaka	Codes for bandwidth-limited channels
I. Oka	Osaka City University, Osaka	Codes for bandwidth-limited channels
K. Ando T. Ichijo K. Ikeda M. Kurihara H. Mizuno	University of Electro-Communications, Tokyo	Algebraic geometry codes
	The Netherlands	
R. Pellikaan	Department of Mathematics & Computer Sciences, Eindhoven University of Technology	Algebraic geometry codes
	Russia	
S. G. Vladut	Central Economics & Mathematics Institute (Tsentral'nyy ekonomiko-matematicheskiy institut), Moscow	Algebraic geometry codes
A. M. Barg G. I. Katsman A. N. Skorobogatov M. A. Tsfasman	Problems of Information Transmission Institute (Institut problem peredachi informatsii), USSR/Russian Academy of Sciences, Moscow	Algebraic geometry codes
	Switzerland	
G. Ungerboeck	IBM–Zürich	Codes for bandwidth-limited channels
H. A. Loeliger T. Mittelholzer	Swiss Federal Institute of Technology (ETH), Zürich	Codes for bandwidth-limited channels

CHAPTER VIII: RELIABLE COMMUNICATION OF COMPRESSED DATA REFERENCES

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APPENDIX A
ABOUT THE AUTHORS

Robert M. Gray *(Panel Chairman).* Dr. Gray is currently a Professor of Electrical Engineering in the Information Systems Laboratory and Acting Vice-Chairman of the Department of Electrical Engineering at Stanford University. He received a BS and MS in Electrical Engineering from the Massachusetts Institute of Technology in 1966, and a PhD in Electrical Engineering from the University of Southern California in 1969. Since 1969 he has been with Stanford University. His research interests are the theory and design of data compression and classification systems, quantization, oversampled analog-to-digital conversion, and image compression and enhancement. He was an Associate Editor (1977-1980) and Editor-in-Chief (1980–1983) of the *IEEE Transactions on Information Theory.* He is currently an Associate Editor of Mathematics of *Control, Signals, and Systems.* He has been on the program committee of several IEEE International Symposia on Information Theory (ISIT) and all IEEE Data Compression Conferences (DCC). He was an IEEE delegate to the Joint IEEE/Popov Society Workshop on Information Theory in Moscow in 1975. He was co-chair with Jerry Gibson of the 1993 IEEE International Symposium on Information Theory in San Antonio, Texas. He is the co-editor, with L. D. Davisson, of the two benchmark collections *Data Compression* (1976) and *Ergodic and Information Theory* (1977); the coauthor, with L. D. Davisson, of *Random Processes* (Prentice Hall, 1986), and with Allen Gersho, of *Vector Quantization and Signal Compression* (Kluwer Academic Press, 1992); and the author of *Probability, Random Processes, and Ergodic Properties* (Springer-Verlag, 1988), *Source Coding Theory* (Kluwer Academic Press, 1990), and *Entropy and Information Theory* (Springer-Verlag, 1990). He was co-recipient with Lee D. Davisson of the 1976 IEEE Information Theory Group Paper Award and co-recipient with Andres Buzo, A. H. Gray, Jr., and J. D. Markel of the 1983 IEEE ASSP Senior Award. He was elected Fellow of the IEEE (1980) and the Institute of Mathematical Statistics (1992). He has held fellowships from the Japan Society for the Promotion of Science (1981) at the University of Osaka, the John Simon Guggenheim Memorial Foundation (1981–1982) at the University of Paris, XI, and the NATO/Consiglio Nazionale delle Ricerche at the University of Naples (1990). In 1984 he received an IEEE Centennial Medal.

Martin Cohn. Dr. Cohn is Senior Research Associate and Senior Lecturer in Computer Science Research at Brandeis University. He received a BA (1956) in Mathematics and an MA (1958) and a PhD (1961) in Applied Mathematics from Harvard University. Dr. Cohn's teaching covers the theory of computation, design of algorithms, information theory, and cryptography. His research interests include digital data compression, the complexity of finite sequences, and the design of codes for input restricted channels, such as magnetic and optical recording media. Prior to joining Brandeis University, Dr. Cohn was a staff member and later a department head at the Sperry Research Center, Sudbury, Massachusetts, where he led a group that designed codes for magnetic recording, algorithms for automated circuit diagnosis, and digital sequences for spread-spectrum and covert communications. This group also developed and was granted the first patent on practical implementations of Lempel-Ziv data compression. Dr. Cohn has been a Visiting Lecturer at Harvard and at Boston University. For the past two years, he has served as Program chair for DCC '92 and '93, the annual Data Compression Conference.

Larry W. Craver. Dr. Craver is a Senior Electronics Engineer in the Department of Defense (DoD) and serves as a scientific advisor to various projects. He received a BEE (1959) and an MSEE (1963) from the Georgia Institute of Technology and a PhD in Engineering (1972) from the University of South Carolina. He has supervised, managed, and conducted research on various technical DoD projects for the past 25 years. Most recently, his main task has been to develop a common system architecture to promote functional and communications interoperability among the DoD Services by using Open Systems concepts and accepted Standards. He was Visiting Professor at the US Naval Academy (1989–90), where he did research and taught electrical engineering. He has spent several years developing both lossless and lossy audio compression techniques, including Lempel-Ziv and Adaptive Transform Coding for voice and other audio communications. A combination of methods was implemented in a final audio data compression prototype. Dr. Craver has also designed and implemented assembly language software for custom designed systems and has provided mathematical support for a number of projects.

Allen Gersho. Dr. Gersho is Professor in the Electrical and Computer Engineering Department at the University of California, Santa Barbara (UCSB), where he directs the Center for Information Processing Research. He received a BS from the Massachusetts Institute of Technology in 1960, and a PhD from Cornell University in 1963. From 1963 to 1980, when he joined the faculty at UCSB, he was at the Mathematics and Statistics Research Center at Bell Laboratories. In 1984-1989 he directed a project to develop speech coding algorithms at 4.8 kb/s and hardware prototypes for NASA's Mobile Satellite Experiment. His current research activities are primarily in the design of speech and video compression algorithms and in the study of new techniques for signal compression. Dr. Gersho holds patents on speech coding, adaptive quantization, digital filtering, adaptive equalization, and modulation and coding for voice band data modems. More than a dozen of his papers have been reprinted in various books of collected benchmark papers in signal processing, communications, and quantization. He is co-author, with R. M. Gray, of *Vector Quantization and Signal Compression* (Kluwer Academic Press, 1992); and co-editor, with B. S. Atal and V. Cuperman, of *Advances in Speech Coding* (Kluwer, Academic Press, 1991) and *Speech and Audio Coding for Wireless and Network Applications* (Kluwer Academic Press, 1993). He was co-chair of the IEEE Workshop on Speech Coding for Telecommunications (Vancouver, September 1989); member of the technical program committee for both the 1991 and 1993 IEEE Workshops on Speech Coding for Telecommunications (Whistler, British Colombia, 1991, and Ste. Adele, Quebec, 1993); co-chair of the XX Annual IEEE Workshop on Communication Theory (Ojai, California, June 1990); member of the steering committee for the annual SPIE Symposium on Visual Communications and Image Processing. He has been Editor of *IEEE Communications Magazine* and Associate Editor of the *IEEE Transactions on Communications;* guest editor for a special section of the *IEEE Journal on Selected Topics in Communications* (May 1984) on Encryption of Analog Signals. He has lectured frequently on topics in digital networks, telephony, and speech coding, including lectures in Norway, Turkey, and the United States for the NATO AGARD Lecture Series on Man-Machine Speech Communications (May 1990), at the International Center for Mechanical Sciences, as part of the course "Linear Prediction: Theory and Applications" (Udine, Italy, July 1990), and at the NATO Advanced Study Institute on Speech Processing (Bibion, Spain, June 1993). Dr. Gersho's awards include the Guillemin-Cauer Prize Paper Award (1980) from the Circuits and Systems Society, the Donald McClennan Meritorious Service Award ([illegible]3) from the IEEE Communications Society, an IEEE Centennial Medal (1984), NASA "Tech Brief" awards for technical innovation (1987, 1988, 1992), the Video Technology Transactions Best Paper Award (co-recipient, 1992) from the IEEE Circuits and Systems Society. In 1981, he was elected Fellow of the IEEE.

Thomas Lookabaugh. Dr. Lookabaugh is currently Vice President of Research and Business Development at DiviCom, in Milpitas, California. He received a BS in Engineering Physics with High Scholastic Honors from the Colorado School of Mines in 1983. While at the Colorado School of Mines, he completed the Honors Program in Public Affairs for Engineers and was elected to Tau Beta Pi (Engineering Honor Society) and Sigma Pi Sigma (Physics Honor Society). He attended Stanford University as a National Science Foundation Fellow, earning Masters degrees in Electrical Engineering, Engineering Management, and Statistics in 1984, 1986, and 1987, respectively, and a PhD in Electrical Engineering in 1988. He was co-recipient of the best paper by an author under 30 from the IEEE Signal Processing Society for his work with P. A. Chou and R. M. Gray, "Entropy Constrained Vector Quantization." Dr. Lookabaugh worked at Compression Labs, Inc., San Jose, California, from 1988 to 1993, obtaining the post of Executive Director of Research and New Business Technology. He served as an adjunct faculty member at the University of Santa Clara from 1988 to 1992, and has taught several courses in the areas of image processing and image and video compression.

Fabrizio Pollara. Dr. Pollara is a Group Leader in the Communications Systems Research Section, Jet Propulsion Laboratory (JPL), Pasadena. He manages source and channel coding research for NASA's Deep Space Network advanced systems program. He received an MS and a PhD in Electrical Engineering from the University of California, Los Angeles, in 1977 and 1982, respectively. Since 1983, he has been with JPL, where he has contributed to the development of advanced telemetry communication systems for deep space probes, and to the management of related projects. His research work includes development of new error-correcting coding systems and of data compression algorithms for images and telemetry data. Dr. Pollara has contributed to the development of the decoders for very large constraint length convolutional codes (two US patents on this subject) now used for deep space NASA missions. He is a part-time Lecturer in the Electrical Engineering Department at the California Institute of Technology, where he teaches error-correction coding theory. Before coming to the United States with a Fulbright scholarship, he received a doctoral degree in Electrical Engineering from the Politecnico of Milan, Italy (1973), and worked on high-speed digital communication links at GTE Telecomunicazioni, Milan, from 1973 to 1976. He contributed a chapter to the new edition of *Deep Space Telecommunications Systems Engineering* (Plenum, to be published).

Martin Vetterli. Dr. Vetterli is currently Associate Professor (Acting) in the Department of Electrical Engineering and Computer Sciences at the University of California/Berleley. He is on leave from Columbia University, where he is an Associate Professor of Electrical Engineering, a member of the Center for Telecommunications Research, and codirector of the Image and Advanced Television Laboratory. Born in Switzerland, he received a Dipl. El.-Ing. from the Eidgenössische Technische Hochschule Zürich, in 1981, an MS from Stanford University in 1982, and a Doctorat ès Science from the Ecole Polytechnique Fédérale de Lausanne in 1986. Prior to joining the faculty at Columbia University in 1986, he was a Research Assistant at Stanford University and the Ecole Polytechnique and worked for Siemens and AT&T Bell Laboratories. Dr. Vetterli is a senior member of the IEEE, member of SIAM and ACM, the MDSP committee of the IEEE Signal Processing Society and of the editorial boards of *Signal Processing, Image Communication, Annals of Telecommunications, Applied and Computational Harmonic Analysis,* and *The Journal of Fourier Analysis and Applications.* He received the Best Paper Award of EURASIP in 1984 for his paper on multidimensional subband coding, the Research Prize of the Brown Bovery Corporation (Switzerland) in 1986 for his thesis, and the IEEE Signal Processing Society's 1991 Senior Award (DSP Technical Area) for a 1989 Transactions paper with D. LeGall on filter banks for subband coding. He was a plenary speaker at the 1992 IEEE International Conference on Acoustics, Speech and Signal Processing, and is co-author, with J. Kovacevic, of the forthcoming book *Wavelets and Subband Coding* (Prentice-Hall, 1994). Dr. Vetterli's research interests include wavelets, multirate signal processing, computational complexity, signal processing for telecommunications and digital video processing and compression.

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APPENDIX B
GLOSSARY OF ABBREVIATIONS AND ACRONYMS

ACM	Association for Computing Machinery
A/D	analog to digital
ADC	analog-to-digital converter (or conversion)
ADP	adaptive density pulse
ADPCM	adaptive differential pulse-coded modulation
AEEC	Airlines Electronic Engineering Committee
AEU	Asian Electronics Union; Archiv für Elektronik und Uebertragungs-technik
AI	artificial intelligence
AIM	Advanced Informatics of Medicine
AMS	American Mathematical Society
APC	adaptive predictive coding
ARINC	Aeronautical Radio, Inc.
ASCII	American Standard Code for Information Interchange
ASPEC	adaptive spectral perceptual entropy coding of high-quality music
ASSP	Acoustics, Speech, and Signal Processing
ATM	asynchronous transfer mode
ATRAC	adaptive transform acoustic coding
AWGN	additive white Gaussian noise
BCELP	binary code-excited linear prediction
BCH	Bose-Chaudhuri-Hocquenghem codes
BCM	block coded modulation
BER	bit error rate
bpp	bit(s) per pixel
bps	bit(s) per sample /bits per second
BTC	block truncation code/coding
CAT	computer-aided tomography (outdated term for CT/computerized tomography)
CCD	electronic imaging sensor (charge-coupled device)
CCETT	Centre Commun d'Etudes de Télédiffusion et Télécommunications (Cesson Sévigné, France)
CCIR	International Radio Consultative Committee
CCITT	International Consultative Committee on Telephone and Telegraphy; recently changed to Telecommunications Standards Sector (TSS)
CD	compact disk
CDMA	code division multiple access
CELP	code-excited linear prediction
CELP-BB	CELP with base-band coding

CEMI	Central Economics and Mathematics Institute, Moscow (Russia)
CEREMADE	Centre de Recherches des Matématiques de la Décision (Université Paris IX, France)
CLAP	Classified Pel- Pattern
CMC	Canadian Microelectronics Corporation
CNET	Centre National d'Etudes des Télécommunications (Paris, France)
CNRS	Centre National de Recherché Scientifique (France)
CPFSK	continuous phase frequency shift keying
CPU	central processing unit
CSELT	Centro Studi e Laboratori Telecomunicazioni SpA (Italy)
CSI	channel state information
CT	computerized tomography
D/A	digital to analog
DAB	digital audio broadcast
DAC	digital-to-analog converter (or conversion)
dB	decibel(s)
DCC	digital compact cassette
DCME	digital circuit multiplication equipment
DCT	discrete cosine transform
DE-QPSK	differentially encoded phase shift keying
DE-QAM	differentially encoded quadrature amplitude modulation
DES	Data Encryption Standard
DLR	Deutsche Forschungs und Versuchsanstalt Luft und Raumfahrt (German Aerospace Research Establishment)
DPCM	differential pulse coded modulation
DSI	digital speech interpolation
DSP	digital signal processor
ECG	electrocardiogram
ECL	effective code length
EIC	Institute of Electronics, Information & Communication Engineers of Japan
ENST	Ecole Nationale Supérieure de Télécommunications (Paris, France—also called TELECOM)
EOB	end of block
EPFL	Ecole Polytechnique Fédérale de Lausanne (Lausanne, Switzerland)
ETH	Swiss Federal Institute of Technology
EURASIP	European Association for Signal Processing
EUSIPCO	European Signal Processing Conference
FCC	(US) Federal Communications Commission
FIR	finite impulse response

GSM	Groupe Speciale Mobile
HDTV	high-definition television
HIS	hospital information systems
IC	integrated circuit
ICASSP	International Conference on Acoustics, Speech, and Signal Processing
ICS	Interlace Coding System
ICT	integer cosine transform
IEE	Institution of Electrical Engineers, United Kingdom
IEEE	Institute of Electrical & Electronics Engineers
IEICE	Institute of Electronics, Information, and Communication Engineers
IFS	iterated function system
IIR	infinite impulse response
IMBE	improved multiband excitation coding
INMARSAT	International Marine Satellite
IRESTE	Institut de Recherche et d'Enseignement Superieur aux Techniques de l'Electronique (Nantes, France)
IRT	Institut für Rundfunktechnik, München/Munich (Germany)
ISDN	Integrated Services Digital Network
ISIT	(IEEE) International Symposium on Information Theory
ISO	International Standards Organization
ITE	Institute of Television Engineers of Japan
JBIG	Joint Bi-Level Image Experts Group
JDC	Japanese Digital Cellular
JFIF	JPEG file interchange format
JPEG	Joint Photographic Experts Group
KAIST	Korea Advanced Institute of Science and Technology (Seoul, Korea)
kb/s (kbps)	kilobits per second
kHz	kilohertz
KLT	Karhunen-Loeve transform
LATI	Laboratoire d'Analyse et Traitement des Images (France)
LFU	least-frequently used
LMS	least mean square
LOT	lapped orthogonal transform
LPC	linear predictive coding
LRII	Laboratoire de Robotique et d'Informatique Industrielle (Nantes, France)
LRU	least-recently used
LSF	line spectral frequency
LSI	large-scale integration

LSP	line spectral pair
MAP	maximum a posteriori
MASCAM	masking pattern adapted sub-band coding and multiplexing
MAT	Medial Axis Transform
MBE	multiband excitation coding
Mbps	megabytes per second
MDL	minimum description length
MDS	maximum distance separable
MHz	megahertz
MIT	Massachusetts Institute of Technology
MPEG	Motion Picture Expert Group
MP-LPC	multipulse linear predictive coding
MRI	magnetic resonance imaging
ms	millisecond/s
mse	mean squared error
MUSICAM	masking pattern adapted universal subband integrated coding and multiplexing
NEC	Nippon Electric Co. (Japan)
NICAM	nearly instantaneous companding audio multiplex
NSA	(US) National Security Agency
NTSC	National Television System Committee
NTT	Nippon Telephone and Telegraph (Japan)
PACS	picture archiving and communication systems
PAL	phase alternation line
PASC	precision adaptive subband coding; picture archiving and communications systems
PCM	pulse coded modulation
PCN	personal communication network
PCS	personal communication system
PKI	Philips Kommunikations Industrie
PSK	phase-shift keying
PSNR	peak signal-to-noise ratio
QAM	quadrature amplitude modulation
QPSK	quadrature phase shift keying
R&D	research and development
RAI	Radiotelevisione Italiana
RCPC	rate compatible punctured convolutional
RLC	run-length coding
ROC	receiver operating characteristic

RPE	regular pulse excitation
RPE-LPC	regular pulse excitation linear predictive coding
R-S	Reed-Solomon
SECAM	sequentiel couleur avec mémoire
SGVQ	shape-gain vector quantizer
SIAM	Society for Industrial and Applied Mathematics
SMPTE	Society of Motion Picture & Television Engineers
SNR	signal-to-noise ratio
SOCC	self-orthogonal convolutional code
SQNR	signal-to-quantization-noise ratio
SQVG	shape-gain vector quantizer
STC	sinusoidal transform coding
TCM	trellis-coded modulation
TIA	(North American) Telephone Industry Association
TSS	Telecommunications Standards Sector; formerly, International Consultative Committee on Telephone and Telegraphy
TSVQ	tree-structured vector quantization
UEP	unequal error protection
VAD	voice activity detection
VCC	vector chain coding
VCR	video cassette recorder
VLC	variable-length coding
VLSI	very large-scale integration
VME	versatile backplane buse derived from the Motorola VERSAbus with European Standard connectors
VQ	vector quantizer/quantization
VSELP	vector sum excited linear prediction
VXC	vector excitation coding

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APPENDIX C
FASAC REPORT TITLES

(* asterisk before title indicates report is classified)

(in production)

FY-93	Non-US Electrodynamic Launchers Research and Development
	Non-US Advanced Signal Processing System Technology
	Non-US Pattern Recognition and Image Understanding Research and Development
	FASAC Special Study: Non-US Artificial Neural Network Research (III)
FY-90-92	Foreign Research in and Applications of Heavy Transuranics
	FASAC Special Study: Non-US Artificial Neural Network Research (II)

(completed)

FY-90-92	Soviet Chemical Propellant Research and Development
	Optoelectronics Research in the Former Soviet Union
	Parallel Processing Research in the Former Soviet Union
	Nonlinear Dynamics Research in the Former Soviet Union
	Penetration Mechanics Research in the Former Soviet Union
	Foreign Bandpass Radome Research and Development
	* Foreign Research Relevant to Countering Stealth Vehicles
	Pulsed Power Research in the Former Soviet Union
	Climate Research in the Former Soviet Union
	Non-US Data Compression and Coding Research
FY-86/89	Soviet Magnetic Confinement Fusion Research
	Recent Soviet Microelectronics Research on III-V Compound Semiconductors
	Soviet Ionospheric Modification Research
	Soviet High-Power Radio Frequency Research
	Free-World Microelectronic Manufacturing Equipment
	FASAC Integration Report II: Soviet Science as Viewed by Western Scientists
	Chinese Microelectronics
	Japanese Structural Ceramics Research and Development
	System Software for Soviet Computers
	Soviet Image Pattern Recognition Research
	West European Magnetic Confinement Fusion Research
	Japanese Magnetic Confinement Fusion Research
	* Soviet Research in Low-Observable Materials

(completed/cont'd.)

FY-86/89	FASAC Special Study: Comparative Assessment of World Research Efforts on Magnetic Confinement Fusion
	FASAC Special Study: Defense Dependence on Foreign High Technology
	Soviet and East European Research Related to Molecular Electronics
	Soviet Atmospheric Acoustics Research
	Soviet Phase-Conjugation Research
	FASAC Special Study: Soviet Low Observable/Counter Low Observable Efforts: People and Places
	Soviet Oceanographic Synthetic Aperture Radar Research
	Soviet Optical Processing Research
	FASAC Integration Report III: The Soviet Applied Information Sciences in a Time of Change
	Soviet Precision Timekeeping Research and Technology
	Soviet Satellite Communications Science and Technology
	West European Nuclear Power Generation Research and Development
	FASAC Special Study: Non-US Artificial Neural Network Research (I)
	* Radiation Cone Research in the Former Soviet Union
FY-85	FASAC Integration Report: Selected Aspects of Soviet Applied Science
	Soviet Research on Robotics and Related Research on Artificial Intelligence
	Soviet Applied Mathematics Research: Electromagnetic Scattering
	* Soviet Low-Energy (Tunable) Lasers Research
	Soviet Heterogeneous Catalysis Research
	Soviet Science and Technology Education
	Soviet Space Science Research
	FASAC Special Report: Effects of Soviet Education Reform on the Military
	Soviet Tribology Research
	Japanese Applied Mathematics Research: Electromagnetic Scattering
	Soviet Spacecraft Engineering Research
	Soviet Exoatmospheric Neutral Particle Beam Research
	Soviet Combustion Research
	Soviet Remote Sensing Research and Technology
	Soviet Dynamic Fracture Mechanics Research
FY-84	Soviet Physical Oceanography Research
	Soviet Computer Science Research
	Soviet Applied Mathematics Research: Mathematical Theory of Systems, Control, and Statistical Signal Processing
	Selected Soviet Microelectronics Research Topics
	* Soviet Macroelectronics (Pulsed Power) Research

(completed/cont'd.)

FY-82/83

- **Soviet High-Pressure Physics Research**
  Soviet High-Strength Structural Materials Research
  Soviet Applied Discrete Mathematics Research
- **Soviet Fast-Reaction Chemistry Research**

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