PRESTO – Preservation Technologies for European Broadcast Archives

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Key Links Systems Specification Document

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ABSTRACT Based on outcomes of the technology survey performed in WP3.task1, this specification is aimed to provide the detailed requirements for a number or 'new key links' to be developed in WP 4, 5, 6 and 7- the new tools needed to make preservation work more cost effective.

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Table of Contents

Figures and tables

Chapter 1 Introduction

1.1 Purpose and scope of the specification

Based on outcomes of the technology survey as provided by [AD4], this specification is aimed to provide the detailed requirements for a number or 'new key links' to be developed in work packages 4, 5, 6 and 7- the new tools needed to make preservation work more cost effective. The required technological developments are all forms of process automation, in three main groupings: materials handling, quality control, and efficient documentation.

1.2 Document structure

The document consists of the following sections.

Chapter 1, this chapter, providing service information.

Chapter 2, "*System context and requirement analysis*". Starting from the generic preservation process, consisting of composition, digitisation, new media creation and archive update, a first level decomposition of the PRESTO preservation chain is provided:

- Analogue media archive
- Composition, pre-processing and transfer
- Playback and digitisation (multiple formats)
- Quality monitoring and validation
- Features extraction,
- Metadata and documentation management
- Digital media store and browsing

Then, the compliance of the identified preservation chains components to the user requirements as gathered in [AD3], Archive Preservation and Exploitation Requirements is shown.

Chapter 3, "*Identification of key links*". First, paragraph 3.1, the Presto key links, described in detail in Chapter 4, are mapped to the video, audio and film preservation chains derived by the high level functional model described in Chapter 2.
Then, paragraph 3.2, the preservation chains infrastructure, conceived as a distributed solution framework, cooperating to the production and management of media assets, is outlined as part of the overall broadcaster's content management system.
Finally, paragraph 3.3, the role of the preservation chain databases, allowing to physically store and access essence and metadata, is shortly recalled.

Chapters 4 to 6 outline "*Key link technology*", provides the detailed requirements of the preservation chain elements to be developed:

- Video: Chapter 4
    - VT1, VT3-Video quality control over digitisation process
    - VT2-Time base corrector with drop out detection and compensation
    - VT4-Multi-level encoding
    - VT5-Lossless compression for video
- Audio: Chapter 5
    - AT1-Audio playback devices improvement
    - AT2-Audio quality control
    - AT3-Lossless compression for audio
- Film: Chapter 6
    - FT1-Auto-re-splice
    - FT2-Alternative handling, specific scanner & procedures

   - FT3-Film Format Converter
   - FT4-Lossless compression for film
- Metadata management
   - MT1-Common access to broadcast archives (Broadcast OPAC)

1.3 Applicable and reference documents

[AD1] Contract IST-1999-20013, PRESTO, Annex 1 - "Description of Work"

[AD2] PRESTO-T11-JRS-20001006, "Quality Assurance Plan"

[AD3] PRESTO-W2-BBC-001218, "Archive Preservation and Exploitation Requirements"

[AD4] PRESTO-WP3-INA-001218, "Existing and emerging technology"

[AD5] EBU Technical Review, Special Supplement August 1998, "EBU / SMPTE Task Force for Harmonized Standards for the Exchange of Programme Material as Bitstreams, Final Report: Analyses and Results"

1.4 Glossary

The following list of specialized terms and acronyms concerning television production, post-production, broadcasting, telecommunications and computer industries, mainly derives from [AD5]

Glossary

-1,2,3…-

24p	24 progressive sampled frames
2k	spatial resolution of about 2000x1500 pixels
4k	spatial resolution of about 4000x3000 pixels
50i	50 interlaced sampled fields

-A-

A / D	Analogue-to-digital conversion.
A / V	Audio/Video or Audiovisual. This abbreviation is often used on the socketry of consumer equipment.
AAL	ATM adaptation layer. The AAL translates digital voice, images, video and data signals into the ATM cell format and vice versa. Five AALs are defined: • AAL1 supports connection-oriented services needing constant bit-rates (CBRs) and specific timing and delay requirements (e.g. DS-3 circuit). • AAL2 supports connection-oriented services needing variable bit-rates (VBRs), e.g. certain video transmission schemes. • AAL3/4 supports both connectionless and connection-oriented variable-rate services. • AAL5 supports connection-oriented variable-rate data services. Also known as Simple And Efficient Adaptation Layer (SEAL).
Access set-up time	The amount of time taken to set up a transmission path between a source and a destination from the moment of commencing the connection process.
Adaptive predictor	A predictor whose estimating function is made variable according to the short-term spectral characteristics of the sampled signal. For ADPCM in particular, an adaptive predictor is a time-varying process that computes an estimate of the input signal from the quantized difference signal.
Adaptive quantizing	Quantizing in which some parameters are made variable according to the short-term statistical characteristics of the quantized signal.
ADAT	Alesys Digital Audio Tape recorder.
Address Translation	The process of converting external addresses into standardized network addresses and vice versa. It facilitates the interconnection of multiple networks in which each have their own Addressing schemes.
ADPCM	Adaptive Differential Pulse Code Modulation. A compression algorithm that achieve bit-rate reduction through the use of adaptive prediction and adaptive quantization.
AES	Audio Engineering Society.
AES3-198	The AES Recommended Practice for Digital Audio Engineering – a Serial Transmission Format for Linearly Represented Digital Audio Data. This is a major digital audio standard for serial inter-face transfer. It is substantially identical to EBU Tech. 3250-E, CCIR Rec. 647, SP/DIF, IEC 958, EIA CP340 and EIA DAT. These standards describe a unidirectional, self- clocking, two-channel standard based on a single serial data signal. The AES format contains audio samples up to 24 bits in length and non-audio data including channel status, user data, parity and sample validity. The differences between these standards lie in electrical levels, connectors, and the use of channel status bits. The AES3 standard is better known as the AES /EBU serial digital audio interface.
Analogue (video) signal	A (video) signal, one of whose characteristic quantities follows continuously the variations of another physical quantity representing information.
Analogue transmission	A type of transmission in which a continuously variable signal encodes an infinite number of values for the information being sent (compare with "digital").

Glossary

Anisochronous The essential characteristic of a time-scale or a signal, such that the time intervals between consecutive significant instants do not necessarily have the same duration or durations that are integral multiples of the shortest duration.

ANSI The American National Standards Institute is a US-based organization that develops standards and defines interfaces for telecommunications systems.

AOD Audio On Demand.

API Application Programming Interface. A set of interface definitions (functions, subroutines, data structures or class descriptions) which together provide a convenient interface to the functions of a subsystem and which insulate the application from the minutiae of the implementation.

Application A computer program designed to perform a certain type of work. An application can manipulate text, numbers, graphics or a combination of these elements. An application differs from an operating system (which runs a computer), a utility (which per-forms maintenance or general-purpose chores) and a programming language (with which computer programs are created).

Application layer The seventh and highest layer in the International Organization for Standardization's Open Systems Interconnection (OSI) model. The application layer contains the signals sent during inter-action between the user and the application, and that perform useful work for the user, such as file transfer.

ASCII American Standard Code for Information Interchange. A coding scheme that assigns numeric values to letters, numbers, punctuation marks and certain other characters. By standardizing the values used for these characters, ASCII enables computers and computer programs to exchange information. Although it lacks accent marks, special characters and non-Roman characters, ASCII is the most universal character-coding system.

ASP Microsoft's Active Server Pages (ASP) technology. An Active Server Page (ASP) is an HTML page that includes one or more script (small embedded programs) that are processed on a Microsoft Web server before the page is sent to the user. ASP is a feature of the Microsoft Internet Information Server

Asset An Asset is any material that can be exploited by a broadcaster or service provider. An asset could therefore be a complete programme file, or it could be a part of a programme, individual sound, images etc.

Asset transfer The transfer of an Asset from one location to another

Asynchronous transmission A term used to describe any transmission technique that does not require a common clock between the two communicating devices, but instead derives timing signals from special bits or characters (e.g. start/stop bits, flag characters) in the data stream itself. The essential characteristic of time-scales or signals such that their corresponding significant instants do not necessarily occur at the same average rate.

ATM Asynchronous Transfer Mode. A form of digital transmission based on the transfer of units of information known as cells. It is suitable for the transmission of images, voice, video, and data.

ATM Layer The protocol layer that relays cells from one ATM node to another. It handles most of the pro-cessing and routing activities including: each cell's ATM header, cell muxing/demuxing, header validation, payload-type identification, Quality of Service (QoS) specification, prioritisation and flow control.

ATSC (US) Advanced Television System Committee.

-B-

Back channel The segment of a two-way communications system that flows from the consumer back to the content provider, or to a system component, to provide feedback.

Backbone The top level in a hierarchical network.

Bandwidth The frequency range of an electromagnetic signal, measured in hertz (cycles per second). The term has come to refer more generally to the capacity of a channel to carry information, as measured in data transferred per second. Transfer of digital data, for example, is measured in bits per second.

Glossary

Bandwidth reservation The process of setting aside bandwidth on a specific broadcast channel for a specific data transmission. A Content server application reserves bandwidth on a Microsoft Broadcast Router by calling the msbdnReserveBandwidth function. This function forwards the request to a Microsoft® Bandwidth Reservation Server. The server returns a unique reservation identifier if the bandwidth can be reserved.

Baseband Describes transmissions using the entire spectrum as one channel. Alternatively, baseband describes a communication system in which only one signal is carried at any time. An example of the latter is a composite video signal that is not modulated to a particular television channel.

Baud Number of bits per second, a measure of data-transmission speed. Baud was originally used to measure the transmission speed of telegraph equipment but now most commonly measures modem speeds. The measurement is named after the French engineer and telegrapher, Jean Maurice-Emile Baudot.

BER Bit Error Rate

B-frame MPEG-2 B-frames use bi-directionally-interpolated motion prediction to allow the decoder to rebuild a frame that is located between two reconstructed display frames. Effectively the B-frame uses both past frames and future frames to make its predictions. B-frames are not used as reference frames but for further predictions. However, they require more than two frames of video storage in the decoder, which can be a disadvantage in systems where low cost is of the essence. By using bi-directional prediction, B-frames can be coded more efficiently than P-frames, allowing a reduction in video bit-rate whilst maintaining subjective video quality.

Broadband A service or system requiring transmission channels capable of supporting rates greater than the Integrated Services Digital Network (ISDN) primary rate (1.544 Mbit/s (e.g. USA) or 2.048 Mbit/s (e.g. Europe)). Broadband is also sometimes used to describe high-speed networks in general.

Broadcast In general terms, a transmission sent simultaneously to more than one recipient. There is a version of broadcasting used on the Internet known as multicast. In multicast, each transmission is assigned its own Internet Protocol (IP) multicast address, allowing clients to filter incoming data for specific packets of interest.

Broadcast (Messages) Transmissions sent to all stations (or nodes, or devices) attached to the network.

Broadcast Router A component that enables a Content server to send a data stream to a multiplexer (MUX) or other broadcast output device. A Broadcast Router calls a virtual interface to transmit a stream at the appropriate rate and in the appropriate packet format.

Broadcaster (Service Provider) An organization which assembles a sequence of events or programmes, based upon a schedule, to be delivered to the viewer.

Buffer An area of storage that provides an uninterrupted flow of data between two computing devices.

BWF Broadcast Wave File. The EBU has defined a file format, which contains the minimum information that is considered necessary for all broadcast applications. The basic information, together with the audio data, is organized as “Broadcast Wave Format” (BWF) files. From these files, using an object-oriented approach, a higher-level descriptor can be used to reference other files containing more complex sets of information, which can be assembled for the different specialized kinds of applications.

-C-

CA Conditional Access. A system to control subscriber access to services, programmes and events.

CBO Continuous Bit-stream Oriented. Services that require an ordered and uninterrupted sequence of data to represent them. PCM-coded video is an example of a CBO service.

CBR Constant bit rate. A type of traffic that requires a continuous, specific amount of bandwidth (e.g. digital information such as video and digitised voice).

CCITT The Consultative Committee on International Telephony and Telegraphy, part of the ITU, develops standards and defines interfaces for telecommunications systems.

CD Compact Disc, an audio media capable to store sounds in digital form using an optical coding on a policarbonate disk.

Glossary

Cell	A transmission unit of fixed length used in cell relay transmission techniques such as ATM. An ATM cell is made up of 53 bytes (octets) including a 5-byte header and a 48-byte data payload.
Cell Relay	Any transmission technique that uses packets of a fixed length. ATM, for example, is a version of the cell relay technique, using 53-byte cells. Other versions use cells of a different length.
CEPT	The Conference on European Post and Telegraph is a European organization that develops standards and defines interfaces for telecommunications systems.
Channel	A means of unidirectional transmission of signals between two points.
CHAP	Challenge Handshake Authentication Protocol.
Chip-set	Several integrated circuits (ICs), which work together to perform a dedicated task. Subsequent development of the chip-set usually decreases the number of ICs needed, and often a single IC implementation is achieved.
Chroma / chrominance	The colour portion of the video signal that includes hue and saturation information. Hue refers to a tint or shade of colour. Saturation indicates the degree to which the colour is diluted by luminance or illumination.
Chunking	The process of “chunking” converts a large file into two or more smaller ones.
Circuit Switching	A switching technique in which a dedicated path is set up between the transmitting device and the receiving device, remaining in place for the duration of the connection (e.g. a telephone call is a circuit-switched connection).
Class	In general terms, a category. In programming languages, a class is a means of defining the structure of one or more objects.
Class Driver	A standard driver provided with the operating system that provides hardware-independent support for a given class of devices. Such a driver communicates with a corresponding hardware dependent minidriver, using a set of device control requests defined by the operating sys-tem. These requests are specific to the particular device class. A class driver can also define additional device control requests itself. A class driver provides an interface between a minidriver and the operating system.
Client	Generally, one of a group of computers that receive shared information sent by a computer called a server over a broadcast or point-to-point network. The term client can also apply to a software process, such as an Automation client, that similarly requests information from a server process and that appears on the same computer as that server process, or even within the same application.
Clock	Equipment that provides a timing signal.
Closed Captioning	Real-time, written annotation of the currently displayed audio Content. Closed Captioning – mainly used in 525-line countries – usually provides subtitle information to hearing-impaired viewers or to speakers of a language other than that on the audio track.
Codec	A combination of an encoder and a decoder in the same equipment.
COM	Component Object Model. An object-oriented programming model for building software applications made up of modular components. COM allows different software modules, writ-ten without information about each other, to work together as a single application. COM enables software components to access software services provided by other components, regardless of whether they involve local function calls, operating system calls or network communications.
Component	(Elementary Stream) One or more entities, which together make up an event, e.g. video, audio, teletext.
Compression	The process of reducing the number of bits required to represent information, by removing redundancy. In the case of information content such as video and audio, it is usually necessary to extend this process by removing, in addition, any information that is not redundant but is considered less important. Compression techniques that are used include: blanking suppression, DPCM, sub-Nyquist sampling, transform coding, statistical coding, sub-band coding, vector coding, run length coding, variable length coding, fractal coding and wavelet coding.
Connectionless	A type of communication in which no fixed path exists between a sender and receiver, even during a transmission (e.g. packet switching). Shared media LANs are connectionless.

Glossary

Connection-oriented A type of communication in which an assigned path must exist between a sender and a receiver before a transmission occurs (e.g. circuit switching). ATM networks are connection-oriented.

Content Programme Content can be Video Essence, Audio Essence, Data Essence and Metadata. Content can therefore include television programming, data and software applications.

Content provider A person or company delivering broadcast Content.

CPU Central Processing Unit. In a personal computer, the CPU is the microprocessor, which is the computer.

CRC Cyclic Redundancy Check. A common technique for detecting errors in data transmission. In CRC error checking, the sending device calculates a number based on the data transmitted. The receiving device repeats the same calculation after transmission. If both devices obtain the same result, it is assumed the transmission was error-free. The procedure is known as a redundancy check because each transmission includes not only data but also additional, redundant values for error checking.

CVD Cell Delay Variation. A measurement of the allowable variation in delay between the reception of one cell and the next, usually expressed in thousandths of a second, or milliseconds (ms). Important in the transmission of voice and video traffic, CDV measurements determine whether or not cells are arriving at the far end too late to reconstruct a valid packet.

-D-

D / A Digital-to-analogue conversion.

DAB Digital Audio Broadcasting. The new coming standard for radio that defines in a digital broadcasting channel not only audio, but also various formats of information.

DAT Digital Audio Tape. The first consumer standard that makes possible recording audio in digital form. A long battle to protect music authors copyright has delayed the DAT market introduction causing its commercial flops.

Data Link layer The second of the seven layers in the International Organization for Standardization's Open Systems Interconnection (OSI) model for standardizing communications. The Data Link layer is one level above the Physical layer. It is involved in packaging and addressing information and in controlling the flow of separate transmissions over communications lines. The Data Link layer is the lowest of the three layers (Data Link, Network and Transport) that help to move information from one device to another. There is also a Data Link layer in the EBU / SMPTE four-layer object model.

Data service A mechanism offered by a broadcaster (service provider) for sending broadcast data to broadcast clients. Such data can include Programme Guide information, WWW pages, software and other digital information. The data service mechanism can be any broadcast process.

Data streaming The data broadcast specification profile for data streaming supports data broadcast services that require a streaming-oriented, end-to-end delivery of data in either an asynchronous, synchronous or synchronized way through broadcast networks. Data which is broadcast according to the data streaming specification is carried in Programme Elementary Stream (PES) packets which are defined in MPEG-2 Systems. Asynchronous data streaming is defined as the streaming of only data without any timing requirements (e.g. RS-232 data). Synchronous data streaming is defined as the streaming of data with timing requirements in the sense that the data and clock can be regenerated at the receiver into a synchronous data stream (e.g. E1, T1). Synchronized data streaming is defined as the streaming of data with timing requirements in the sense that the data within the stream can be played back in synchronization with other kinds of data streams (e.g. audio, video).

Datagram One packet of information and associated delivery information, such as the destination address, that is routed through a packet-switching network. In a packet-switching network, data packets are routed independently of each other and may follow different routes and arrive in a different order from which they were sent. An Internet Protocol (IP) multicast packet is an example of a datagram.

DAVIC Digital Audio VIsual Council. DAVIC has been convened along similar lines to MPEG but with no affiliation to a standards body; it therefore has the status of a worldwide industry consortium. Its purpose is to augment MPEG and to collect system specifications for the delivery of a range of audio-visual services that can be applied uniformly on a worldwide basis.

Glossary

DB	DataBase, a software component capable to store information's providing tool for complex search operations.
DBS	Digital Broadcasting System.
DCT	Discrete Cosine Transform. A DCT process basically involves dividing the picture up into 8 x 8 pixel blocks, then replacing the discrete luminance and chrominance values of each pixel by the amplitudes of the corresponding frequency components for the horizontal and vertical directions respectively. In this way, the information is transformed from the time domain to the frequency domain. No information is lost in this process, except perhaps by the rounding of the last digit of the frequency coefficient values.
Delivery system	The physical medium by which one or more multiplexes (MUXs) are transmitted, e.g. a satellite system, wide-band coaxial cable, fibre optics, terrestrial channel of one emitting point.
DEMUX	Demultiplexer. A device that performs the complementary operation to that of a multiplexer (MUX).
Descrambler	A device that performs the complementary operation to that of a scrambler.
Device	A unit of hardware, for example a videotape machine or a server.
Device class	A group into which devices are placed for the purposes of installing and managing device drivers, and for allocating resources.
Device driver	A software component that allows an operating system to communicate with one or more specific hardware devices attached to a computer.
Device object	A programming object used to represent a physical, logical or virtual hardware device whose device driver has been loaded into the operating system.
DIF	Digital InterFace. All the DV-based compression schemes share the so-called DIF structure, which is defined in the "Blue Book" (IEC61834).
Digital (transmission) channel	The means of unidirectional digital transmission of digital signals between two points.
Digital connection	A concatenation of digital transmission channels, switching and other functional units, set up to provide for the transfer of digital signals between two or more points in a network, in support of a single communication.
Digital demultiplexing	The separation of a (larger) digital signal into its constituent digital channels.
Digital multiplexing	A form of time-division-multiplexing applied to digital channels by which several digital signals are combined into a single (larger) digital signal.
Digital signal	A discretely-timed signal in which information is represented by a number of well defined discrete values that one of its characteristic quantities may take in time.
Digital transmission	The transmission of digital signals by means of a channel or channels that may assume, in time, any one of a defined set of discrete states.
Digital-S	JVC / Victor Company of Japan trademark. This tape system is based on DV technology, and has a data-rate of 50Mbit/s.
DLT	Digital Linear Tape. A magnetic tape cartridge capable to store in the 7000 version 35 Gbytes of non compressed data. This king of media is a standard de-facto in the multimedia archive based on robot system. DLT is the most compact media today available on the market and the one with the longest media life and number of readout.
Downstream	One-way data flow from the head-end to the broadcast client.
DPCM	Differential Pulse Code Modulation. A process in which a signal is sampled, and the difference between each sample of this signal and its estimated value is quantized and converted by encoding to a digital signal.
DSP	Digital signal processor.
DTD	Document Type Definition. The purpose of a DTD is to define the legal building blocks of an XML document. It defines the document structure with a list of legal elements. A DTD can be declared inline in your XML document, or as an external reference.
DTS	Data Time Stamp.
DV	Digital Video. A digital videotape format originally conceived for consumer applications.

Glossary

DVB	Digital Video Broadcasting.
DVB-C	DVB framing structure, channel coding and modulation scheme for cable systems (EN 300 429).
DVB-S	DVB baseline system for digital satellite television (EN 300 421).
DVB-T	DVB baseline system for digital terrestrial television (EN 300 744).
DVC	Digital Video Cassette.
DVCPRO, DVCPRO50	Panasonic trademarks. Based on DV technology and having a data-rate of 25Mbit/s and 50Mbit/s respectively.
DVD	Digital Versatile (Video) Disk.
-E-	
EBU	European Broadcasting Union. Headquartered in Geneva, Switzerland, the EBU is the world's largest professional association of national broadcasters. Following a merger on 1 January 1993 with the International Radio and Television Organization (OIRT) – the former association of Socialist Bloc broadcasters – the expanded EBU has 66 active members in 49 European and Mediterranean countries, and 51 associate members in 30 countries elsewhere in Africa, the Americas, and Asia.
ECM	Entitlement Control Message.
Enhancement	A multimedia element, such as a hypertext link to a WWW page, a graphic, a text frame, a sound or an animated sequence, added to a broadcast show or other video programme. Many such elements are based on Hypertext Mark-up Language (HTML).
EPG	Electronic Programme Guide.
Error ratio	[error rate]The ratio of the number of digital errors received in a specified period to the total number of digits received in the same period.
Error, digital error	An inconsistency between a digit in a transmitted digital signal and the corresponding digit in the received digital signal.
ESCR	Elementary Stream Clock Reference.
ETR	ETSI Technical Report.
ETS	European Telecommunication Standard.
ETSI	European Telecommunications Standards Institute.
-F-	
FEC	Forward error correction. A system of error correction that incorporates redundancy into data so that transmission errors can, in many cases, be corrected without requiring retransmission.
Field	In broadcast television, one of two sets of alternating lines in an interlaced video frame. In one field, the odd-numbered lines of video are drawn on the screen; in the other, the even-numbered lines are drawn. When interlaced, the two fields combine to form a single frame of on-screen video.
File	An organized collection of related records, accessible from a storage device via an assigned address. The relationship between the records and the file may be that of common purpose, format or data source, and the records may or may not be sequenced.
Frame	In broadcast television, a single screen-sized image that can be displayed in sequence with other slightly different images to animate drawings. In the case of NTSC video, a video frame consists of two interlaced fields of 525 lines; NTSC video runs at 30 frames per second. In the case of PAL or SECAM video, a video frame consists of two interlaced fields of 625 lines; PAL and SECAM video run at 25 frames per second. By way of comparison, film runs at 24 frames per second. A variable-length packet of data is used by traditional LANs such as Ethernet and Token Ring, as well as WAN services such as X.25 or Frame Relay. An edge switch will take frames and divide them into fixed-length cells using an AAL format. A destination edge switch will take the cells and reconstitute them into frames for final delivery.
FTP	File Transfer Protocol. A protocol that supports file transfers to and from remote systems on a network using Transmission Control Protocol / Internet Protocol (TCP/IP), such as the Internet. FTP supports several commands that allow the bi-directional transfer of binary and ASCII files between systems.

Glossary

FTP+	FTP+ is an enhanced version of FTP, and uses the same base set of commands. FTP+ includes new commands that enable traditional features and which also provide the ability to embrace network protocols other than IP.
-G-	
Gbit/s	Gigabit per second. A digital transmission speed of billions of (i.e.10 9) bits per second.
Genre	A category of broadcast programmes, typically related by style, theme or format, e.g. TV movies or television series.
GoP	Group of Pictures. An MPEG-2 GoP begins with an I-frame and extends to the last frame before the next I-frame. The GoP sequence is known as an open GoP – the last frame in the GoP uses the first frame of the next GoP as a reference. Another type of GoP is a closed GoP, which has no prediction links to the next GoP and, by definition, always ends in a P-frame.
GSM	Global System for Mobile communication.
Guaranteed bandwidth	Bandwidth that is reserved only if the requested bandwidth is available for the requested period. Once reserved, such bandwidth can be relied upon to be available.
-H-	
HDTV	High Definition TeleVision. Television that is delivered at a higher screen resolution than that of NTSC, PAL or SECAM.
Head-end	The origin of signals in a terrestrial, cable, satellite or network broadcast system. In Broadcast Architecture, the server infrastructure that gathers, coordinates and broadcasts the data is generally located at the broadcast head-end.
HEX	Hexadecimal. A numbering system with a base of 16 (binary numbers have a base of 2, and decimal numbers have a base of 10). In HEX notation, the decimal numbers 0 to 9 are extended by the addition of the uppercase letters A to F, i.e. 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F (which is equivalent to the numbers 0 to 15 in decimal notation).
Host	A device where one or more modules can be connected, e.g. a VTR, a PC ...
HTML	Hypertext Mark-up Language. A mark-up language used to create hypertext documents that are portable from one platform to another. HTML files are text files with embedded codes, or mark-up tags, that indicate formatting and hypertext links. HTML is used for formatting documents on the WWW.
HTTP	Hypertext Transport Protocol. The underlying, application-level protocol by which WWW clients and servers communicate on the Internet.
-I-	
ID	Identifier
IDL	Interface Definition Language. Used to describe interfaces that client objects call, and object implementations provide. It is a purely descriptive language which has mappings provided for several programming languages such as C++, C and Java. It has the same lexical rules as C++.
IEC	International Electrotechnical Commission. Based in Geneva, the IEC is the world organization that prepares and publishes international standards for all electrical, electronic and related technologies.
IEEE	(US) Institute of Electrical and Electronic Engineers. The world's largest technical professional society, with more than 320,000 members. The technical objectives of the IEEE focus on advancing the theory and practice of electrical, electronic and computer engineering, and computer science.
IETF	Internet Engineering Task Force. The IETF is a large open international community of network designers, operators, vendors and researchers concerned with the evolution of the Internet architecture and the smooth operation of the Internet. It is open to any interested individual.
I-Frame	Intra-coded Frame. I-frame pictures make use only of information already contained within that frame. They are not dependent on other frames and can act as the starting point to enable decoders to begin working on a GoP containing a sequence of other types of frame. The amount of compression achievable is typically less than for the other types of frame.
IIOP	Internet Inter-ORB Protocol.

Glossary

Interactive television The interactive combination of a video programme and multimedia enhancement elements such as hypertext links, graphics, text frames, sounds and animations.

Interface The common boundary point where two elements connect so that they can work with one another. In computing, the connection between an application and an operating system or between an application and a user (the user interface) are examples of an interface. In C++ programming, an interface is a collection of related methods exposed by a given class of objects. These methods are procedures that can be performed on or by those objects.

Interlacing / interlaced A video display technique, used in current analogue televisions, in which the electron beam refreshes (updates) all odd-numbered scan lines in one field and all even-numbered scan lines in the next. Interlacing takes advantage of both the screen phosphor's ability to maintain an image for a short period of time before fading, and the human eye's tendency to average subtle differences in light intensity. By refreshing alternate lines, interlacing halves the number of lines to update in one screen sweep. An alternative video display technique, used in computer monitors, is progressive scanning. In progressive scanning, the image is refreshed one line at a time.

Internet Generically, a collection of networks interconnected with routers. The Internet is the largest such collection in the world. It has a three-level hierarchy composed of backbone networks, mid-level networks and stub networks.

IOR Interoperable Object Reference.

IP Internet Protocol. The primary network layer of Internet communication, responsible for addressing and routing packets over the network. IP provides a best-effort, connectionless delivery system that does not guarantee that packets arrive at their destination or that they are received in the sequence in which they were sent.

IP Address An identifier for a network node; expressed as four fields separated by decimal points (e.g. 136.19.0.5.); IP address is site-dependent and assigned by a network administrator.

IPCP Internet Protocol Control Protocol.

IP-over-ATM The adaptation of TCP/IP and its address resolution protocol for transmission over an ATM network. It is defined by the IETF in RFCs 1483 and 1577. It puts IP packets and ARP requests directly into protocol data units and converts them to ATM cells. This is necessary because IP does not recognize conventional MAC-layer protocols, such as those generated on an Ethernet LAN.

IS Interactive Service.

ISDN Integrated Services Digital Network. A type of dial-up service. Data can be transmitted over ISDN lines at speeds of 64 or 128 kbit/s, whereas standard phone lines generally limit modems to top speeds of 20 to 30 kbit/s.

ISO International Organization for Standardization, based in Geneva.

Isochronous A term used to describe signal-timing techniques that require a uniform reference point (usually embedded in the data signal).

ITU International Telecommunication Union, part of the United Nations, based in Geneva.

-J-

Java An object-oriented, platform-independent computer programming language developed by Sun Microsystems. The Applet subclass of Java can be used to create Internet applications.

JDBC Java Database Connectivity. JDBC (Java Database Connectivity) is an application program interface specification for connecting programs written in Java to the data in popular database.

Jitter Short-term non-cumulative variations of the significant instants of a digital signal from their ideal positions in time.

Jitter, delay, latency See Latency

JSP Java Server Pages. Java Server Page (JSP) is a technology for controlling the content or appearance of Web pages through the use of Servlet, small programs that are specified in the Web page and run on the Web server to modify the Web page before it is sent to the user who requested it. Sun Microsystems, the developer of Java, also refers to the JSP technology as the Servlet application program interface (API). JSP is comparable to Microsoft's Active Server Page (ASP) technology.

-K-

Glossary

Kbit/s	Kilobits per second. A digital transmission speed expressed in thousand of bits per second.
-L-	
LAN	Local Area Network. A network dispersed over a relatively limited area and connected by a communications link that enables each device on the network to interact with any other.
LAN Emulation	The process of implementing enough of the media access control layer protocol of a LAN (e.g. Ethernet or Token Ring) to allow existing higher layer protocols (and applications) to be used unchanged over another network, such as an ATM network.
Latency	The time delay inherent in a manipulative process. In particular, the time that it takes to process an input bitstream through a compression and decompression process. Buffering and trans-mission can be major contributors to processing delays.
Link	Any physical connection on a network between two separate devices, such as an ATM switch and its associated end point or end station.
Log on	To provide a user name and password that identifies you to a computer network.
LSB	Least Significant Bit. In any related grouping of bits (i.e. a word), there will be one that quantifies the zeroth power of 2 (i.e. the value is 0 or 1). This bit is the LSB of the word.
Luminance	A measure of the degree of brightness or illumination radiated by a given source. Alternatively, the perceived brightness component of a given colour, as opposed to its chroma.
-M-	
MAA	MPEG ATM Adaptation.
MAC	Media Access Control.
MAN	Metropolitan area network.
Master clock	A clock that is used to control the frequency of other clocks.
Mbit/s	Megabits per second. A digital transmission speed expressed in millions of bits per second.
MBONE	Multicast backbone. A virtual, multicast-enabled network that works on top of the Internet. The most popular application for the MBONE is video conferencing, including audio, video and whiteboard conferencing. However, the essential technology of the MBONE is simply multicast – there is no special support for continuous media such as audio and video. The MBONE has been set up and maintained on a co-operative, volunteer basis.
MD	Metadata Data i.e. data that are describing other data.
Metadata	Data describing other data.
MIB	Management Information Base.
MIME	Multipurpose Internet Mail Extensions.
MJD	Modified Julian Date.
MMDS	Microwave Multi-point Distribution Systems (or Multichannel Multi-point Distribution Systems). Also known as wireless cable.
MMI	Man Machine Interface. The MMI of a door is its doorknob. That of a PC is a combination of keyboard, mouse and monitor.
Module	A small device, not working by itself, designed to run specialized tasks in association with a host – for example, a conditional access sub system, or an electronic programme guide application module – or to provide resources required by an application but not provided directly by the host
MPEG	Motion Picture Experts Group. MPEG-1 is a standard designed for video playback from CD-ROM. It provides video and audio compression at rates up to 1.8 Mbit/s. MPEG-2 refers to the ISO/IEC 13818 standard, and it provides higher video resolutions and interlacing for broadcast television and high-definition television (HDTV). Both standards were created by the Motion Pictures Experts Group, an International Standards Organization / International Telegraph and Telephone Consultative Committee (ISO/CCITT) group set up to develop motion video compression standards. The MPEG system makes use of three different types of compressed video frames, (I, P and B frames), which are stored so as to enable temporal prediction of missing or incomplete frames as received by the decoder.
MPEG TS	MPEG Transport Stream.

Glossary

MPI	MPEG Physical Interface.
MPLS	Multi-protocol Label Swapping.
MSB	Most Significant Bit. In any related grouping of bits (i.e. a word), there will be one that quantifies the largest power of 2. This bit is the MSB of the word.
MTBF	Mean Time Between Failure. The expected time between two possible independent failure of a hardware component.
MTTR	Mean Time To Repair. The average time required to repair a given system from a hardware failure.
MTU	Multiport Transceiver Unit.
Multicast	A point-to-many networking model in which a packet is sent to a specific address, and only those computers that are set to receive information from this address receive the packet. On the Internet, the possible IP multicast addresses range from 224.0.0.0 through 239.255.255.255. Computer networks typically use a unicast model, in which a different version of the same packet is sent to each address that must receive it. The multicast model greatly reduces traffic and increases efficiency on such networks.
Multicast Messages	A subset of “broadcast” in which a transmission is sent to all members of a pre-defined group of stations, nodes or devices.
Multimedia	Online material that combines text and graphics with sound, animation or video, or some combination of the three.
Multipoint	Generally encountered in the term "point-to-multipoint" which describes a broadcast topography.
MUX	Multiplex or multiplexer. A stream of all the digital data carrying one or more services within a single physical channel. In general terms, a multiplexer is a device for funnelling several different streams of data over a common communications line. In the case of broadcasting, a multiplexer combines multiple television channels and data streams into a single broadcast.
MVDS	Multipoint Video Distribution System.
-N-	
NE	Network Element. Network In computing, a data communications system that interconnects a group of computers and associated devices at the same or different sites. In broadcasting, a collection of MPEG-2 Transport Stream multiplexes that are transmitted on a single delivery system, e.g. all the digital channels on a specific satellite or cable system. NFS Network File System. This is defined in RFC 1813. File system access is different from file transfer, in that Network File Systems generally employ a client-server model in which the server computer actually has the file system as local data. The client-host is allowed to “mount” the network file system to get access to the directories and files as if they were locally available. Multiple clients are permitted to simultaneously “mount” the server's file system and get access to its Content.
NCITS	National Committee for Information Technology Standards. NCITS T11 is responsible for standards development in the areas of Intelligent Peripheral Interface (IPI), High-Performance Parallel Interface (HIPPI) and Fibre Channel (FC). NNI Network-to-Network Interface. In an ATM network, the interface between one ATM switch and another, or an ATM switch and a public ATM switching system. NTSC National Television System Committee – which originated the NTSC standard for analogue television signals in North America, and which has also been adopted in Japan and parts of South America. The NTSC system is based on a power supply frequency of 60 Hertz (Hz) and can display 525 scan lines at approximately 30 frames per second. However, non-picture lines and interlaced scanning methods make for an effective resolution limit of about 340 lines. The bandwidth of the system is 4.2 Megahertz (MHz).
Network	In computing, a data communications system that interconnects a group of computers and associated devices at the same or different sites. In broadcasting, a collection of MPEG-2 Transport Stream multiplexes that are transmitted on a single delivery system, e.g. all the digital channels on a specific satellite or cable system.
NFS	Network File System. Communication standard between computers on a LAN introduced by Sun that allows a remote computer to see a storage unit of a remote computer as a local one.

Glossary

-O-

Object	A computer programming term describing a software component that contains data or functions accessed through one or more defined interfaces. In Java and C++, an object is an instance of an object class.
Octet	A group of eight binary digits or eight signal elements capable of representing 256 different values operated upon as an entity (also known as a "word").
ODBC	Open Database Connectivity. Open Database Connectivity (ODBC) is an open standard application programming interface (API) for accessing a database.
OPAC	Online Public Access Catalogue
Operating system	Software responsible for controlling the allocation and usage of computer hardware resources such as memory, CPU time, disk space and peripheral devices.
Opportunistic bandwidth	Bandwidth granted whenever possible during the requested period, as opposed to guaranteed bandwidth that is actually reserved for a given transmission.
OSI	Open Systems Interconnection. This refers to the ISO / OSI seven layer model for standardizing communications.

-P-

Packet	A unit of information transmitted as a whole from one device to another on a network. In packet-switching networks, a packet is defined more specifically as a transmission unit of fixed maximum size that consists of binary digits (bits) representing both data and a header containing an identification number, source and destination addresses, and sometimes error-control data.
Packet Switching	A switching technique in which no dedicated path exists between the transmitting device and the receiving device. Information is formatted into individual packets, each with its own address. The packets are sent across the network and reassembled at the receiving station.
PAL	Phase Alternation by Line standard. The analogue television standard for much of Europe – except France, Russia and most of Eastern Europe, which use SECAM. As with SECAM, PAL is based on a 50 Hertz (Hz) power supply frequency, but it uses a different encoding process. It displays 625 scan lines and 25 frames per second, and offers slightly better resolution than the NTSC standard used mainly in North America and Japan. The PAL bandwidth is 5.5 Megahertz (MHz).
Partial Transport Stream	Bitstream derived from an MPEG-2 TS by removing those TS packets that are not relevant to one particular selected programme, or a number of selected programmes.
PCM	Pulse Code Modulation. A process in which a signal is sampled, and each sample is quantized independently of other samples and converted by encoding to a digital signal.
PC	Personal Computer.
PCR	Programme Clock Reference.
PDH Plesiochronous Digital Hierarchy.	
PDU	Protocol Data Unit. A unit of information (e.g. a packet or frame) exchanged between peer layers in a network.
PES	Packetized Elementary Stream.
P-frame	MPEG-2 P-frames use a single previously-reconstructed frame as the basis for temporal prediction calculations; they need more than one video frame of storage. Effectively the P-frame uses the nearest previous frame (I or P) on which to base its predictions, and this is called forward prediction. P-frames serve as the reference frame for future P- or B-frames, but if errors exist in a particular P-frame, they may be carried forward to the future frames derived from them. P- frames can provide a greater degree of compression than I-frames.
PHP	PHP is a script language and interpreter that is freely available and used primarily on Linux Web servers. PHP (the initials come from the earliest version of the program, which was called "Personal Home Page Tools") is an alternative to Microsoft's Active Server Page (ASP) or SUN's Java Server Page (JSP) technology.
Physical Layer	The first of the seven layers in the International Organization for Standardization's Open Systems Interconnection (OSI) model for standardizing communications. It specifies the physical interface (e.g. connectors, voltage levels, cable types) between a user device and the net- work.

Glossary

PID	Packet Identifier.
Plesiochronous	The essential characteristic of time-scales or signals such that their corresponding significant instants occur at nominally the same rate, any variation in rate being constrained within specified limits. Two signals having the same nominal digit rate, but not stemming from the same clock, are usually plesiochronous.
PLL	Phase Locked Loop.
Plug and Play	A design philosophy and set of specifications that describe changes to hardware and software for the personal computer and its peripherals. These changes make it possible to automatically identify and arbitrate resource requirements among all devices and buses on a computer. Plug and Play specifies a set of application programming interface (API) elements that are used in addition to existing driver architectures.
Point-to-point	A term used by network designers to describe network links that have only one possible destination for a transmission.
Port	Generally, the address at which a device such as a network interfaces card (NIC), serial adapter or parallel adapter communicates with a computer. Data passes in and out of such a port. In Internet Protocol (IP), however, a port signifies an arbitrary value used by the Transmission Control Protocol / Internet Protocol (TCP/IP) and User Datagram Protocol / Internet Protocol (UDP/ IP) to supplement an IP address so as to distinguish between different applications or proto-cols residing at that address. Taken together, an IP address and a port uniquely identify a sending or receiving application or process.
PRBS	Pseudo Random Binary Sequence.
Predictor	A device that provides an estimated value of a sampled signal, derived from previous samples of the same signal or from a quantized version of those samples.
Printf	A symbol in the C programming language.
Programme	A concatenation of one or more events under the control of a broadcaster, e.g. a news broadcast, entertainment show.
PSI	MPEG-2 Programme Specific Information (as defined in ISO/IEC 13818-1).
PSK	Phase Shift Keying.
PSTN	Public Switched Telephone Network.
PTS	Presentation Time Stamp.
Push model	A broadcast model in which a server sends information to one or more clients on its own schedule, without waiting for requests. The clients scan the incoming information, save the parts they have been instructed to save, and discard the rest. Because the push model eliminates the need for requests, it eliminates the need for a back channel from the client to the server. The push model contrasts with the pull model, in which each client requests information from a server. The pull model is more efficient for interactively selecting specific data to receive, but uses excessive bandwidth when many clients request the same information.
PVC	Permanent Virtual Circuit. A generic term for any permanent, provisioned, communications medium. Note that PVC does not stand for permanent virtual channel. In ATM, there are two kinds of PVCs: permanent virtual path connections (PVPCs) and permanent virtual channel connections (PVCCs).
-Q-	
QAM	Quadrature Amplitude Modulation.
QoS	Quality of Service. The ATM Forum, for example, has outlined five categories of performance (Classes 1 to 5) and recommends that ATM's QoS should be comparable to that of standard dig-ital connections.
QPSK	Quadrature Phase Shift Keying.
Quantizing / quantized	A process in which a continuous range of values is divided into a number of adjacent intervals, and any value within a given interval is represented by a single predetermined value within the interval.
Query	A request that specific data be retrieved, modified or deleted.

Glossary

-R-

RAID	Redundant Array of Independent Disks. A means of constructing a server by interconnecting several hard disk units such that the data is distributed across all of them. If an individual hard disks fails, the remainder can continue working and the defective unit can be replaced, usually without taking the server out of service.
RAM	Random access memory. RAM is semiconductor-based memory within a personal computer or other hardware device that can be rapidly read from and written to by a computer's microprocessor or other devices. It does not generally retain information when the computer is turned off.
Reference clock	A clock of very high stability and accuracy that may be completely autonomous and whose frequency serves as a basis of comparison for the frequency of other clocks.
Regeneration	The process of receiving and reconstructing a digital signal so that the amplitudes, waveforms and timing of its signal elements are constrained within specified limits.
Registry	A hierarchical database that provides a repository for information about a system's hardware and software configuration.
Resource	A unit of functionality provided by the host for use by a module. A resource defines a set of objects exchanged between the module and the host by which the module uses the resource. An example of a resource is a piece of static data, such as a dialog box, that can be used by more than one application or in more than one place within an application. Alternatively, it is any part of a computer or network, such as a disk drive, printer or memory, that can be used by a program or process.
RFC	Request For Comment.
RFT	Request for Technology.
RMS / rms	Root Mean Square.
Router	A device that helps local-area networks (LANs) and wide-area networks (WANs) to connect and interoperate. A router can connect LANs that have different network topologies, such as Ethernet and Token Ring. Routers choose the best path for a packet, optimising the network performance.
RS-422	A serial data interface standard. RS-232 has been around as a standard for decades as an electrical interface between Data Terminal Equipment (DTE) and Data Circuit-Terminating Equipment (DCE) such as modems, and is commonly the serial interface found on PCs. The RS-422 interface is a balanced version of the interface, and it is much less prone to interference from adjacent signals.
RSVP	Resource reSerVation Protocol. RSVP is a QoS signalling protocol for application-level streams. It provides network-level signalling to obtain QoS guarantees.
RTP	Real-time Transport Protocol. RTP permits real-time Content transport by the inclusion of media-dependent Time Stamps that allow Content synchronization to be achieved by recovering the sending clock.

-S-

S / N	(SNR) Signal-to-Noise Ratio. The amount of power by which a signal exceeds the amount of channel noise at the same point in transmission. This amount is measured in decibels and indicates the clarity or accuracy with which communication can occur.
Sample	A representative value of a signal at a chosen instant, derived from a portion of that signal.
Sampling / sampled	The process of taking samples of a signal, usually at equal time intervals.
Sampling rate	The number of samples taken of a signal per unit of time.
Satellite uplink	The system that transports a signal up to a satellite for broadcasting. Signals usually come to the uplink through multiplexers (MUXs).
SCSI	Small Computer System Interface. Popular standard for peripheral interconnection to a computer. Used mainly for disk and tape devices has evolved in the past years from the initial 2 Mbytes/sec to the today 80 Mbytes/sec.
SCPC	Single Channel Per Carrier transmission.
Scrambler	A device that converts a digital signal into a pseudo-random digital signal having the same meaning and the same digit rate.

Glossary

SDH	Synchronous Digital Hierarchy. International version of SONET that is based on 155 Mbit/s increments rather than SONET's 51 Mbit/s increments.
SDTV	Standard Definition TeleVision. Television service providing a subjective picture quality roughly equivalent to current 525-line or 625-line broadcasts.
SECAM	Séquentiel Couleur á Mémoire, or Sequential Colour with Memory. The television standard for France, Russia and most of Eastern Europe. As with PAL, SECAM is based on a 50 Hertz (Hz) power supply frequency, but it uses a different encoding process. Devised earlier than PAL, its specifications reflect earlier technical limitations.
Server	A computer or other device connected to a network to provide a particular service (e.g. print server, fax server, playout server) to client devices connected to the network.
Service	A set of elementary streams offered to the user as a programme. They are related by a common synchronization. They are made of different data, e.g. video, audio, subtitles and other data. Alternatively, it is a sequence of programmes under the control of a broadcaster, which can be broadcast as part of a schedule.
Service_id	A unique identifier of a service within a TS.
SI	Service Information. Digital data describing the delivery system, Content and scheduling / timing of broadcast data streams etc. It includes MPEG-2 PSI together with independently defined extensions (ETS 300 468).
Signalling	(ATM) The procedures used to establish connections on an ATM network. Signalling standards are based on the ITU's Q.93B recommendation.
Slip	The loss or gain of a digit position or a set of consecutive digit positions in a digital signal, resulting from an aberration of the timing processes associated with transmission or switching of a digital signal.
SMPTE	(US) Society of Motion Picture and Television Engineers. The Society was founded in 1916, as the Society of Motion Picture Engineers. The T was added in 1950 to embrace the emerging television industry. The SMPTE is recognized around the globe as a leader in the development of standards and authoritative, consensus-based, recommended practices (RPs) and engineering guidelines (EGs). The Society serves all branches of motion imaging including film, video and multimedia.
SNMP	Simple Network Management Protocol.
SNMP2	Simple Network Management Protocol version 2. An enhancement of the simple gateway monitoring protocol, and which was designed as a connectionless application-level protocol within TCP/IP that uses UDP as a Transport layer.
SOAP	Simple Object Access Protocol (SOAP). SOAP is a lightweight protocol for exchange of information in a decentralized, distributed environment. It is an XML based protocol that consists of three parts: an envelope that defines a framework for describing what is in a message and how to process it, a set of encoding rules for expressing instances of application-defined data types, and a convention for representing remote procedure calls and responses. SOAP can potentially be used in combination with a variety of other protocols.
SONET	Synchronous Optical NETwork. A set of standards for the digital transmission of information over fibre optics. Based on increments of 51 Mbit/s.
SQL	Structured Query Language
Station	An establishment equipped for radio or television transmission.
STM	Synchronous Transfer Mode / Synchronous Transport Module. In ATM, a method of communications that transmits data streams synchronized to a common clock signal (reference clock). In SDH, it is "Synchronous Transport Module" and is the basic unit (STM-1 = 155 Mbit/s, STM-4 = 622 Mbit/s, STM-16 = 2.5 Gbit/s) of the Synchronous Digital Hierarchy.
Streaming	A collection of data sent over a data channel in a sequential fashion. The bytes are typically sent in small packets, which are reassembled into a contiguous stream of data. Alternatively, it is the process of sending such small packets of data.
Streaming architecture	A model for the interconnection of stream-processing components, in which applications dynamically load data as they output it. Dynamic loading means data can be broadcast continuously.
String	Data composed of a sequence of characters, usually representing human-readable text.

Glossary

SVC	Switched Virtual Circuit. A generic term for any switched communications medium. Note that SVC does not stand for switched virtual channel. In ATM, there are two kinds of SVCs: switched virtual path connections (SVPCs) and switched virtual channel connections (SVCCs).
Switch	Device used to route cells through an ATM network.
Symbol rate	The number of signal elements of the signal transmitted per unit of time. The baud is usually used to quantify this, one baud being equal to one single element per second.
Synchronization	The process of adjusting the corresponding significant instants of signals to make them synchronous.
Synchronous	A term used to describe a transmission technique that requires a common clock signal (or timing reference) between two communicating devices to co-ordinate their transmissions.
Synchronous network	A network in which the corresponding significant instants of nominated signals are adjusted to make them synchronous.
-T-	
Task Scheduler	A scheduling service and user interface that is available as a common resource within an operating system. A Task Scheduler manages all aspects of job scheduling: starting jobs, enumerating currently running jobs, tracking job status, and so on.
TCP	Transmission Control Protocol.
TCP / IP	Transmission Control Protocol / Internet Protocol. A networking protocol that provides reliable communications across interconnected networks made up of computers with diverse hardware architectures and operating systems. The TCP portion of the protocol, a layer above IP, is used to send a reliable, continuous stream of data and includes standards for automatically requesting missing data, reordering IP packets that might have arrived out of order, converting IP datagrams to a streaming protocol, and routing data within a computer to make sure the data gets to the correct application. The IP portion of the protocol includes standards for how computers communicate and conventions for connecting networks and routing traffic.
TDM	Time-division multiplexing. Multiplexing in which several signals are interleaved in time for transmission over a common channel.
Telecommunication	Any transmission and/or emission and reception of signals representing signs, writing, images and sounds or intelligence of any nature by wire, radio, optical or other electromagnetic systems.
TFHS	The Joint EBU / SMPTE Task Force for Harmonized Standards for the Exchange of Programme Material as Bitstreams.
Theme	A category to which individual television programmes are assigned within the Guide database. A theme allows a programme episode to be associated with multiple genre / subgenre pairs.
Timing recovery	[timing extraction] The derivation of a timing signal from a received signal.
Timing signal	A cyclic signal used to control the timing of operations.
Traffic Policing	A mechanism used to detect and discard or modify ATM cells (traffic) that do not conform to the Quality of Service (QoS) parameters specified in the call setup procedure.
Traffic Shaping	A mechanism used to control traffic flow so that a specified QoS is maintained.
Transmission	The action of conveying signals from one point to one or more other points.
Transparency, digital transparency	The property of a digital transmission channel, telecommunication circuit or connection, that permits any digital signal to be conveyed over it without change to the value or order of any signal elements.
Transport layer	The fourth of the seven layers in the International Organization for Standardization's Open Systems Interconnection (OSI) model for standardizing communications. The Transport layer is one level above the Network layer and is responsible for error detection and correction, among other tasks. Error correction ensures that the bits delivered to the receiver are the same as the bits transmitted by the sender, in the same order and without modification, loss or duplication. The Transport layer is the highest of the three layers (Data Link, Network and Transport) that help to move information from one device to another.
Transport_stream_id	A unique identifier of a TS within an original network.

Glossary

TS	Transport Stream. A TS is a data structure defined in ISO/IEC 13818-1 for the MPEG-2 Transport Stream. It is the basis of the ATSC and DVB standards.
TV	Television.
Twisted-pair cable	A communications medium consisting of two thin insulated wires, generally made of copper, that are twisted together. Standard telephone connections are often referred to as "twisted pair."
-U-	
UDP	User Datagram Protocol. UDP, as defined in RFC 768, can be used as an option to enable bounded-quality transfers on top of the IP layer. It allows broadcast transmissions and is a datagram-oriented protocol.
UDP/IP	User Datagram Protocol / Internet Protocol. A networking protocol used to send large unidirectional packets across interconnected networks made up of computers with diverse hardware architectures and operating systems. The UDP portion of the protocol, a networking layer above IP, is used to send unidirectional packets of up to 64 kilobytes in size and includes standards for routing data within a single computer so it reaches the correct client application. The IP portion of the protocol includes standards for how computers communicate and conventions for connecting networks and for routing traffic.
UML	Unified Modelling Language. The UML is a language for specifying, visualizing, constructing and documenting the artefacts of software systems. It assists the complex process of software design, making a "blueprint" for construction.
UNI	User-to-Network Interface. A connection that directly links a user's device to a network (usually through a switch). Also, the physical and electrical demarcation point between the user device and the switch.
Unicast	A point-to-point networking model in which a packet is duplicated for each address that needs to receive it.
UNO-CDR	Universal Networked Object – Common Data Representation.
Upstream	One-way data flow from the broadcast client to the head-end.
URI	Uniform Resource Identifier. Also known as a URL.
URL	Uniform Resource Locator. URLs are short strings that identify resources on the WWW: documents, images, downloadable files, services, electronic mailboxes and other resources, etc. They may be thought of as a networked extension of the standard filename concept, in that not only can you point to a file in a directory, but that file and that directory can exist on any machine on the network, can be served via any of several different methods, and might not even be something as simple as a file.
User mode	Software processing that occurs at the application layer.
UTC	Universal Time Co-ordinated.
UTF	The Unicode standard is the universal character encoding standard used for representation of text for computer processing. Unicode provides a consistent way of encoding multilingual plain text. This standards are a transformation format of ISO 10646.
U-U	User-User
-V-	
VBI	Vertical Blanking Interval. The time period in which a television signal is not visible on the screen because of the vertical retrace (that is, the repositioning of the trace to the top of the screen to start a new scan). Data services can be transmitted using a portion of this signal. In a standard NTSC signal, perhaps 10 scan lines are potentially available per channel during the VBI. Each scan line represents a data transmission capacity of about 9600 baud. In 625-line systems, about 20 scan lines are available in the VBI.
VBR	Variable Bit-Rate. A type of traffic that, when sent over a network is tolerant of delays and changes in the amount of bandwidth it is allocated (e.g. data applications).
VBV	Video Buffer Verifier. The MPEG concept defined in ISO/IEC 13818-2 (MPEG-2, Annex C) employs a fixed-size buffer to handle the transition of the channel bit-rate to the rapidly fluctuating coded bit-rate of individual MPEG pictures. The scope of the VBV is only within a sequence. The VBV is built upon a framework of several axioms of decoder behaviour, which are unfortunately not very well described in the specification.
VC	Virtual Circuit. A generic term for any logical communications medium.

Glossary

VCC Virtual Channel Connection. A logical communications medium identified by a VCI and carried within a VPC.

VCI Virtual Channel Identifier. The field in the ATM cell header that labels (identifies) a particular virtual channel.

VCR Video Cassette Recorder.

VHF Very High Frequency.

VHS Video Home System.

Virtual LAN A logical association of users sharing a common broadcast domain.

VPC Virtual Path Connection. A logical communications medium in ATM identified by a Virtual Path Identifier (VPI) and carried within a link. VPCs may be permanent virtual path connections (PVPCs), switched virtual path connections (SVPCs), or smart permanent virtual path connections (SPVPCs). VPCs are uni-directional.

-W-

WAN Wide Area Network. A communications network that connects geographically separated areas.

Wander Long-term non-cumulative variations of the significant instants of a digital signal from their ideal positions in time.

Wrapper A function that provides an interface to another function.

WWW World Wide Web / the Web. A hypertext-based, distributed information system created in Switzerland and used for exploring the Internet. Users may create, edit or browse hypertext documents on the Web.

-X-

XML Extensible Markup Language. The Extensible Markup Language (XML) is the universal format for structured documents and data on the Web.

XTP eXtended Transport Protocol. A network-level interface appropriate for file transfer. XTP can operate in a “raw” mode in which it encompasses both the Network and Physical layers, or it can operate on top of IP. XTP in raw mode achieves some efficiency and has the possibility of using features of the underlying physical media (such as the QoS for ATM) that is not possible when XTP is used on top of IP.

-Y-

YUV True-colour encoding that uses one luminance value (Y) and two chroma values (UV).

Chapter 2 System context and requirement analysis

2.1 PRESTO generic preservation process

The preservation process as considered in the PRESTO project is given by a nominal chain consisting of more steps:

- Composition
- Digitisation
- New media creation
- Archive update

Composition	Digitisation	New Media Creation	Update Archive
Identify and Assemble materials	Create a digital master copy plus low-data rate versions	Create a new archive item (physical or electronic)	Replace old item with new; update metadata

Figure 2.1: Reference process

At each step, a number of actions has to be performed both at media processing and at metadata processing level. They are summarised by the following figures.

Stage	**Identify**	**Get Basic Items**	**Get assoc. items**	**Dispatch**
		Legacy System "Issue"	New Process	
Meta-data Processing	Material needing processing	Extract metadata from existing documents relative to digitising	Get any additional information	Prepare all labels for old and new physical media
Media Processing		Pick media from archive	Program docs. + physical items relevant to digitising	Dispatch media

Figure 2.2 Breakdown of Composition phase of preservation process

Stage	**Check**	**Digitise media**	**Digitise associated media**	**Process digitised media**	**Format physical media**	**Coding of associated versions, e.g. "browse"**
Metadata Processing	Ensure metadata complete and accurate	Automatic or manual quality check (of digitisation) during process	Capture of entire documents or essential new data	Create shot detection, key frames, speech recognition, MPEG-7 data	Arrange metadata to fit standard file format	Transfer metadata to any subsidiary media
Media Processing	Ensure media matches Meta-	Transfer content form original	Scanning or other processing	(media must be on or written to	Media on server made into "im-	Create images or media for all

Metadata	media to server or directly to new media	of any associ-ated media or documents	a server); new data stored	"image" of final output (if any)	subsidiary data

Table 2.1.1 Breakdown of Digitisation phase of preservation process

Stage	Label	Write	Label	Dispatch
	Electronic metadata goes with media			
Metadata Processing	Ensure all required identifying names or numbers are in the metadata	Metadata written to same media as audio and video	Produce paperwork	To accompany new media
Media Processing	Create electronic image of new media	Burn or record new media item	Physical print of Identifying data onto media item and onto packaging	Media grouped in batches, packaged for transport

Table 2.1.2 Breakdown of New Media Creation phase of preservation process

Stage	Check	Accession	Store	Update Metadata	Final	Remove old media
		Legacy system “Accesssion”	Standard Function	New, possibly very labour intensive		
		Legacy system “Accesssion”	Standard Function	New, possibly very labour intensive		
Media Processing	Ensure media matches metadata	Scan barcodes inot stock control system	New media placed in physical archive		Last chance to ensure that new media matches original media	Dispose of Old Media

Table 2.1.3 Breakdown of the Update of physical Archive process

2.2 High level functional model

2.2.1 System context

The diagram below sketches the relationships among the PRESTO preservation chain (as outlined in the previous paragraph) and the external world (its context)

The *Preservation Chain* counterparts are:

- The analogue media archive
- The user communities
- The legacy system that hosts data relating to the analogue media, to be ingested into the digital media storage
- The pre-processing sites

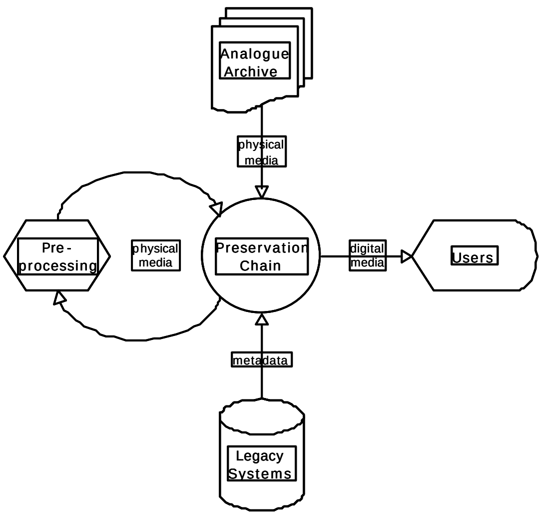

Figure 2.1: Preservation process system context

2.2.2 High level processes

Figure 2.4 shows the main processes within the preservation chain.

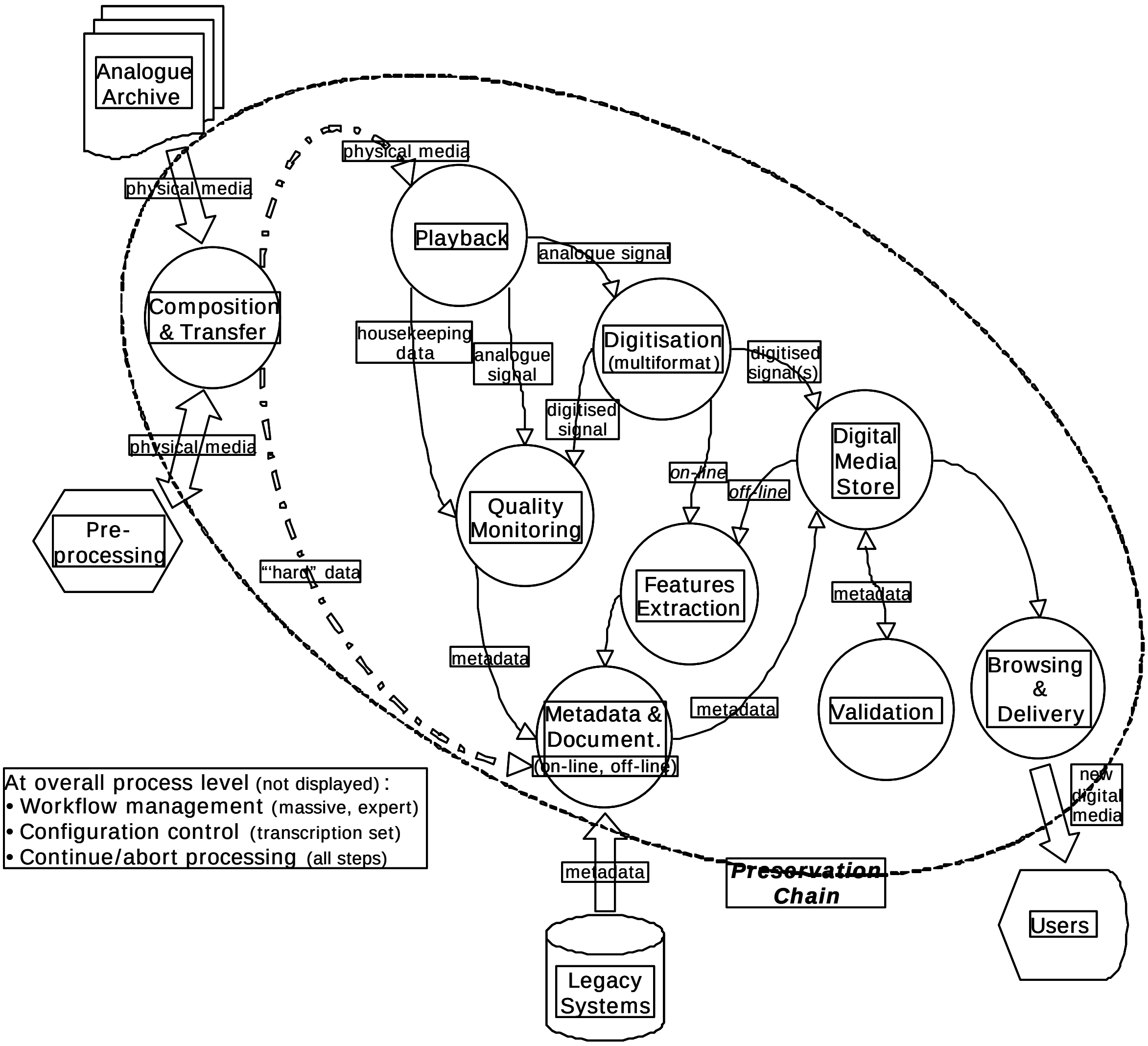

Figure 2.1: First level decomposition

Analogue media archive

It is important to conserve original analogue media in case a better digitisation process becomes available.

Pre-processing

Preparation processing can be seen as a form of basic restoration, allowing media to be suitable for playback. Separate techniques could apply to different physical items, depending on media type and its actual conservation status, e.g.:

- Recovery of improperly stored audio tapes

- Chemical film cleaning
- Vacuum cleaning, "baking", or even editing could apply to old videotapes.
- Vinyl surface cleaning

Composition and transfer

Composition: The collection of all related media before conversion creating a *transcription set*, including Metadata, which can be extracted from existing documents (legacy system).

Transfer: The download and upload of media to or from storage areas. It is essential in this phase that new media is correctly identified and that analogue material is correctly returned back to the archive after processing to conserve information within the system.

Playback

Media handling: Some media can be manipulated using robotic devices e.g. video cassettes, however older media may require manual handling.

Reproduction: It is essential that playback devices read the original accurately, to achieve quality digitised media.

Digitisation (multiple formats)

Digitisation: The conversion of the existing analogue signal to a digital representation, or the migration from old to new digital formats.

Compression – When digitising, it is possible to remove redundancy by compressing the output digital stream. However in video, it is usually necessary to remove information that is not redundant but is considered less important. Reconstruction from the compressed bit stream thus leads to the addition of *distortions* or *artefacts* therefore the process is not normally lossless.

Currently it is not necessary to compress audio for preservation purposes.

Multiple formats - Multiple new items occur when multiple versions (formats) are being created of the same inherent item. The main example is creation of a master copy and low-data rate versions for edit or browse or web access.

The processing of such lower rate digital media can be either performed on-line, by processing the ongoing digitisation stream, or off-line, applied to the already stored digital master. In both cases (and as far as applicable to the specific media), should more formats be needed, they could be either produced at the same time (parallel processing) or serialised, one at the time. On-line processing is a more efficient throughput of the storage system, since it needs to store data only once, at the end of processing, while off-line processing requires separate manipulation. On the other hand, on-line processing is costly in terms of computer processing power.

Parallel and serial processing strategies imply a trade-off evaluation. Parallel processing is simpler to manage: you have different codecs processing at the same time a master media, to obtain different formats. Since the time required to perform different types of compressions greatly vary, the overall process speed will dictated by the slowest codec. Unfortunately, this approach is inefficient: faster codecs will spent the most of their time in idle state. Furthermore the system is costly to scale up and rebuild.

Serial processing overcomes such problems: you can saturate each codec by processing all the media it can sustain; if some format requires reprocessing, only the associated codec need be engaged. System scalability is optimal. However, the model is more complex, vast increases in workflow can arise endangering the system and resources

Quality monitoring

Metadata for quality - The key to a cost-effective preservation is to provide automatic quality detection during digitisation, eliminating the need for "manned" playback. Restoration systems are already equipped to provide quality analysis. However detection systems must be robust and reliable, incorrect assessment leads to unnecessary preservation or the loss of valuable information.

Quality control should take into account:

- *The behaviour of the transcription chain*.
- *The status of the original media*. The 'original format and its characteristics' are an important part of the original material; the preservation processing decisions vital for understanding the new 'artefact' and for any further processing (such as signal restoration).
- *Information about the digitisation/ conversion process*. The key aspect of media preservation is to create a new item that is as close as possible in signal content to the original.
- *Master copy and low-data rate versions comparison*. Enusring consistency across multiple versions e.g. time codes

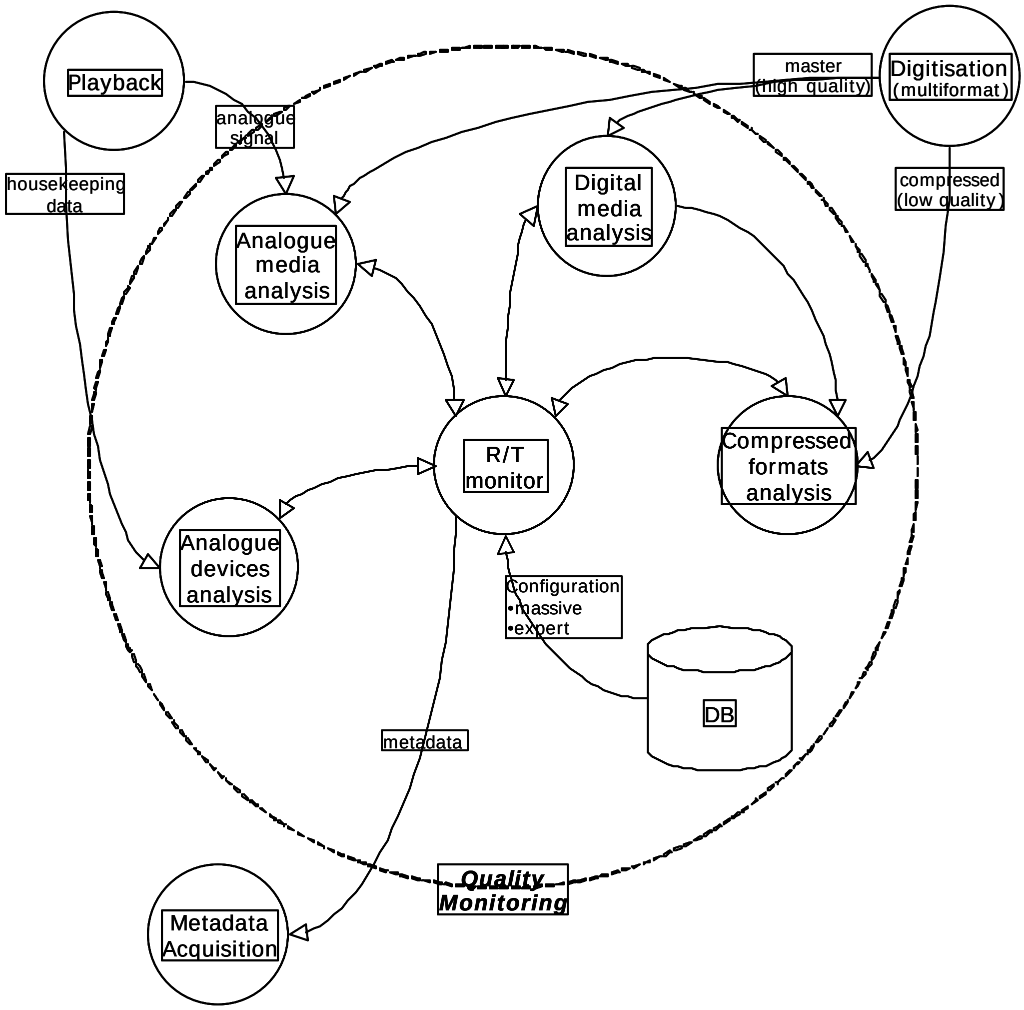

Figure 2.1: Quality monitoring diagram

Massive vs. Expert transcription – It is possible to define two separate transcription approaches: one specially optimised in terms of speed and cost reduction (***Massive Transcription***), and a second involving a high degree of specific expertise for endangered valuable materials that requires accuracy instead of speed (***Expert Transcription***). The two transcription approaches are actually described by the same high-level functional model. At this level, they are supported by different configurations of the quality monitoring chain, driven by a parameters database

Features extraction

Metadata for retrieval – additional enhanced metadata (contents-oriented) can be produced while digitising, to fully exploit online media, e.g.: time coding, key frame extraction, indexing, shot-level description, and automatic speech recognition.

Features extraction could be performed either on-line, concurrently to the digitisation process, or off-line, applied to the already stored digital master.

Generaly, as many features as possible should be extracted on-line from the digitisation stream, in order to optimise the storage system throughput, even though not all types of processing are suitable. However, features validation should always be performed off-line, since intervention of a technician is required.

Metadata and documentation

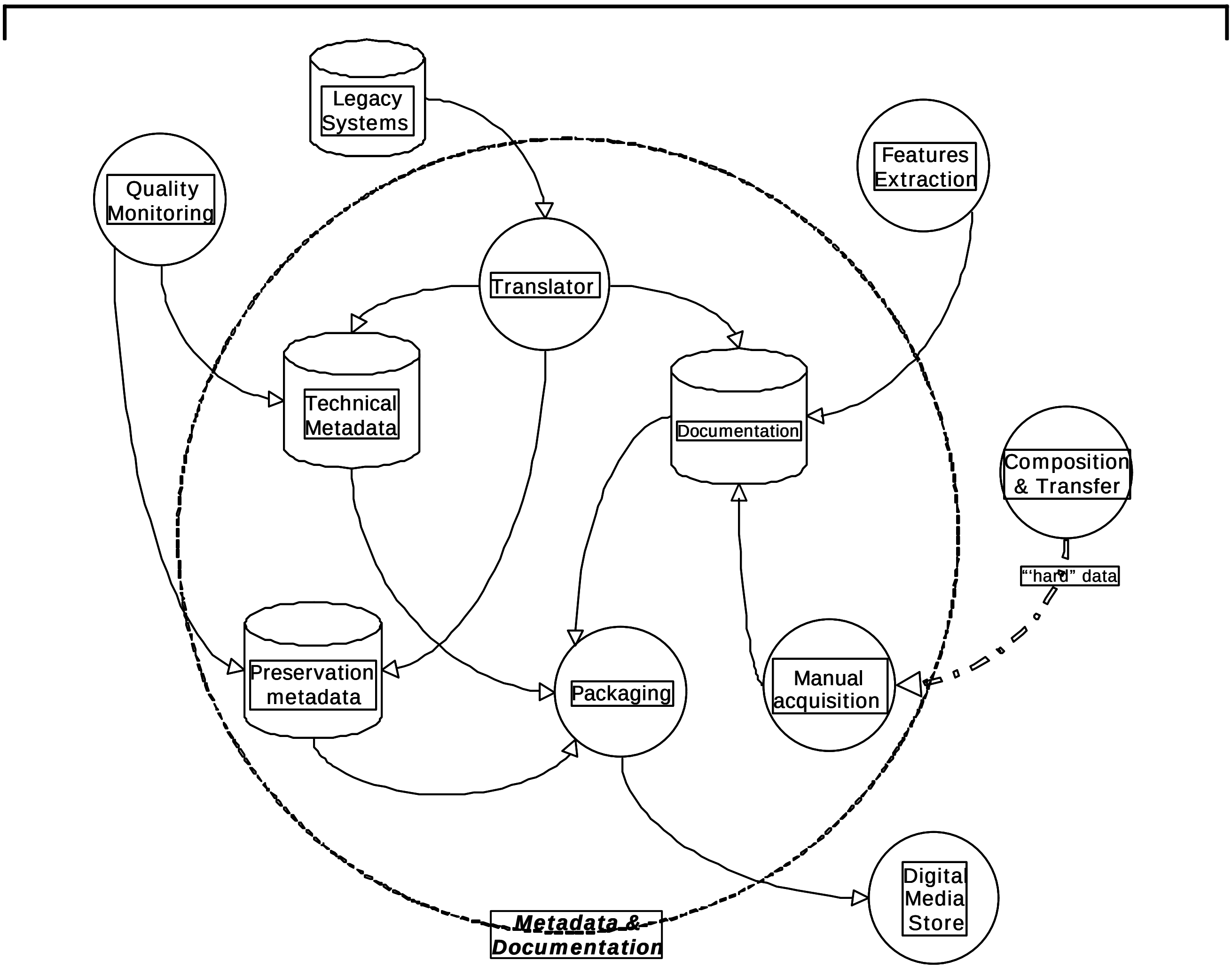

Figure 2.1: Metadata and Documentation diagram

This process joins all the activities related to metadata preparation.

Metadata can be spit into three main categories:

- Technical: media with embedded metadata (e.g timecode)
- Preservation: concerning information about the original material format, its characteristics, and the preservation (digitisation / conversion) process, i.e., information for understanding the new 'artefact' and for any further processing (e.g., signal restoration).
- Documentation: mainly related to the media contents description, including media original box/ folders or additional enhanced metadata (indexing, …). Documentation can also be provided by operator, by means of optical scanning of paper images, and textual data entry.

In general, data produced inside the preservation chain are identified by their source. In the case metadata retrieved from external legacy systems, some translation and mapping effort may be required to properly import data (see also *Generalized browsing* in 0). Data elements in a legacy system need to be mapped to elements of standard schema, such as the SMPTE metadata dictionary, or EBU P/META, or Dublin Core. Missing elements, such as a Universal

Media Identifier (SMPTE UMID) need to be created, possibly using components from legacy identifiers.

Metadata can be achieved:

- On-line (during processing): mainly data concerning quality monitoring and features extraction; also the acquisition of available documents packaged with the media (e.g., acquisition of LP covers via optical scanner) could be performed on-line, to avoid multiple handling of the source materials.

- Off-line (after media digitalisation and storage). Should be mainly reserved to documentation: storage of material (audio or browse video) on a server would allow the cataloguing staff to perform this task, rather than the technicians. Digital-to-digital processing could be performed off-line, so that related technical metadata will be completed simultaneously.

Validation

Based on the *Quality Monitoring* detected parameters, a sort of go/no go response should be provided within the preservation system, informing the user whether subsequent restoration processing is required on the digitised pieces or not.

As a tentative basis, the response could be:

- Digitised media is suitable for immediate broadcasting “as is”
- Digitised media can be stored in the media archive, but some digital restoration is needed
- Digitised media quality is too poor, maybe restoring is required at source level, before attempting a new digitisation. Probably, the digitised media should be removed from the archive

The validation could be performed either on-line, or off-line. In the former case, it could be possible to avoid the archival at all.

Digital media store and browsing

Digital media store consists of two main branches:

- *Raw temporary storage*, hosting data and metadata after digitisation, waiting for validation.

- *Long term new digital media storage*. On its own, this storage provides two main facilities: a *Catalogue*, storing metadata, accessed by users to select media to retrieve, and an A*rchive*, storing the digitised media. The user’s access to the catalogue can be improved by hosting previews of the archived material, i.e., video and audio samples produced with a very low quality.

Upon satisfactory validation, media and metadata are ingested into the long term storage system (actually, moved from raw storage). They can also be simply removed from the system if the validation fails.

End-user’s access is restricted to the long term archive; digital media can only be accessed after entries have been inserted into the *Catalogue*, through an OPAC facility.

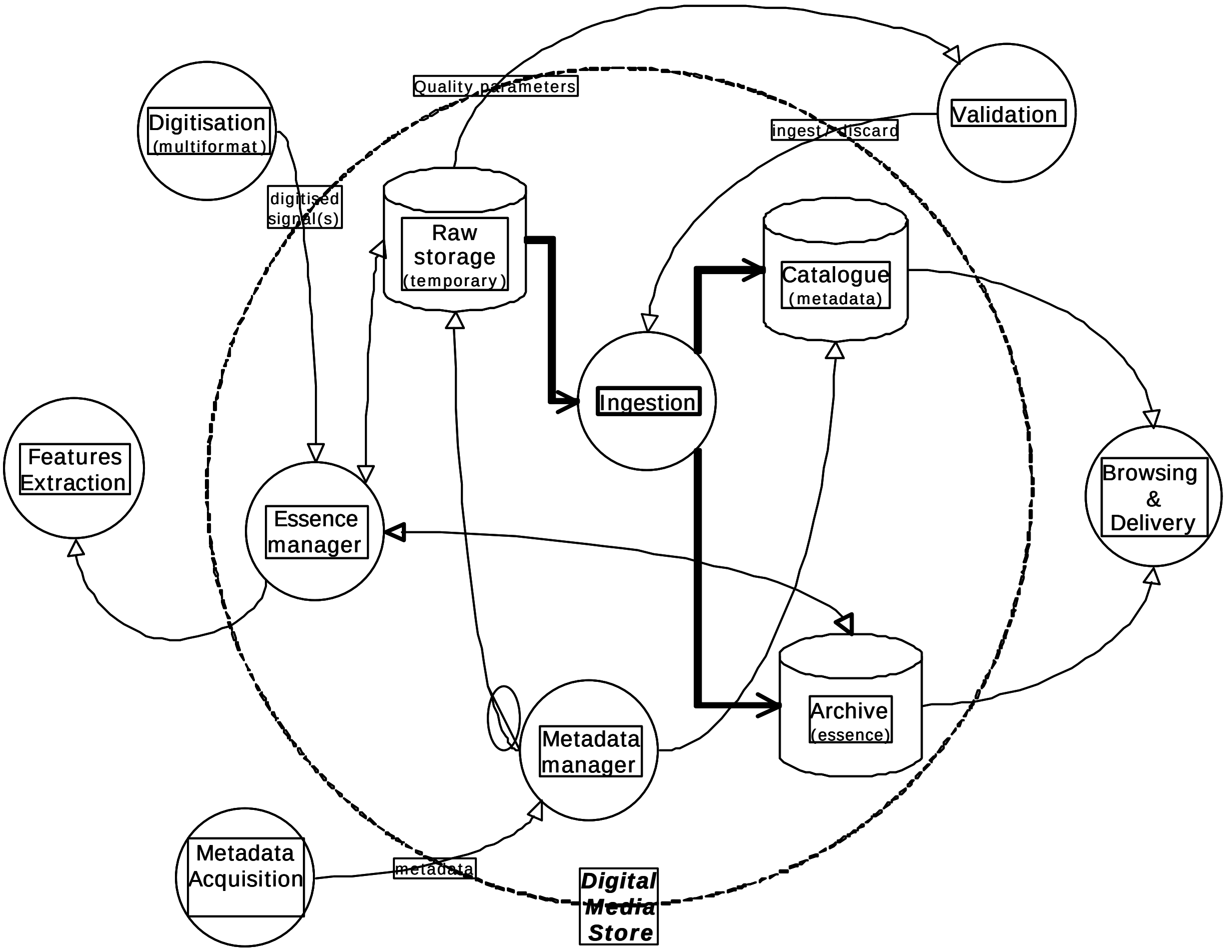

Figure 2.1: Digital media store diagram

On the other hand, local media and metadata management functions allow access at any time when required by preservation staff (technicians performing digitisation, cataloguing staff and others) and processes, to the proper set of data (either raw or long term).

The physical architecture of the digital media store depends on the size of the archive and expected performance. It can be based on some HMS (Hierarchical Storage Management), automatically and transparently managing customer data across a storage hierarchy, consisting of high performance disks, compressed disks, and tape libraries. Robotic libraries can also be used to fully automate media access.

Raw and the *long-term* stores can be either different physical archives or logical areas of the same storage system.

Generalized browsing – Catalogue inter-operability is one of the hottest issues in data sharing over widespread systems.
The problem is simple to describe: a generic user performing a search, should be allowed to retrieve data from many heterogeneous (and remote) information systems with a unique single query, regardless of the different data structures and models implemented all over its search domain. In other words, the user needs to issue a 'title' or 'author' or 'subject' query against a database, without knowing anything equivalent to the specific logical data model.
The solution is a bit more complex. It's mainly based on the capability of the different systems to "explain" each other what they need and what they have. This can be achieved by means of complex software architectures. One of them, widely used, is the American NISO standard Z39.50, providing a general purpose basis for inter-operability. Different "profiles" can be built on top of the Z39.50 protocol, specific to the application domain. The most widely known profile is BIB, used in libraries since long. A second profile is CIP, still under standardization, specific to the field of catalogues of products deriving from satellite remote sensing (i.e., earth observation from satellite).

Chapter 3 Identification of key links

3.1 Preservation chains and key links

The high level functional model described in Chapter 2 apply as well to video, as to audio and film preservation chains. The existing differences (even deep) do not appear at that level.

The following diagrams map the Presto key link, described in detail in Chapter 4 , to the three preservation chains.

3.1.1 Video preservation

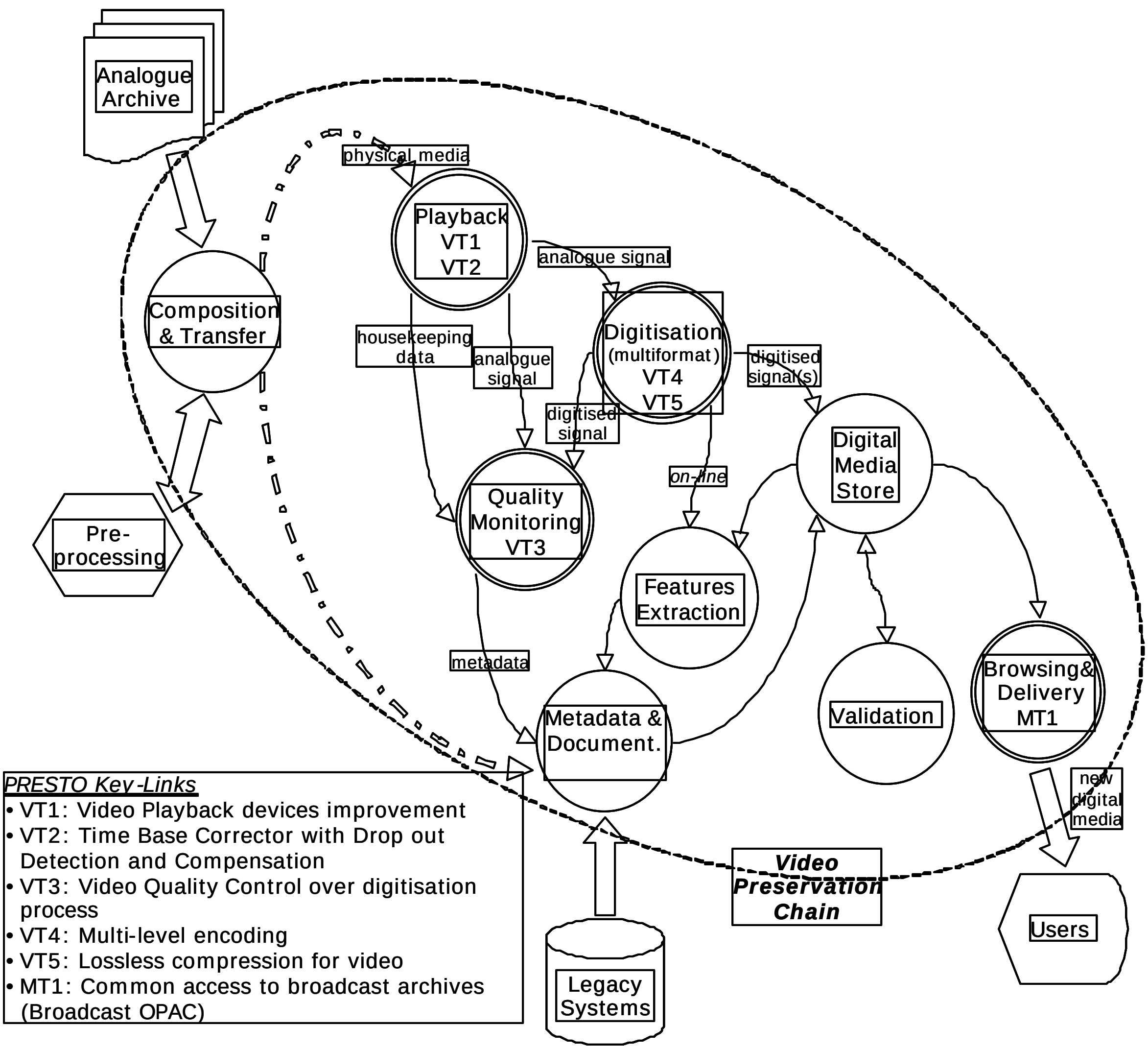

Figure 3.1: Key links for video preservation

3.1.2 Audio preservation

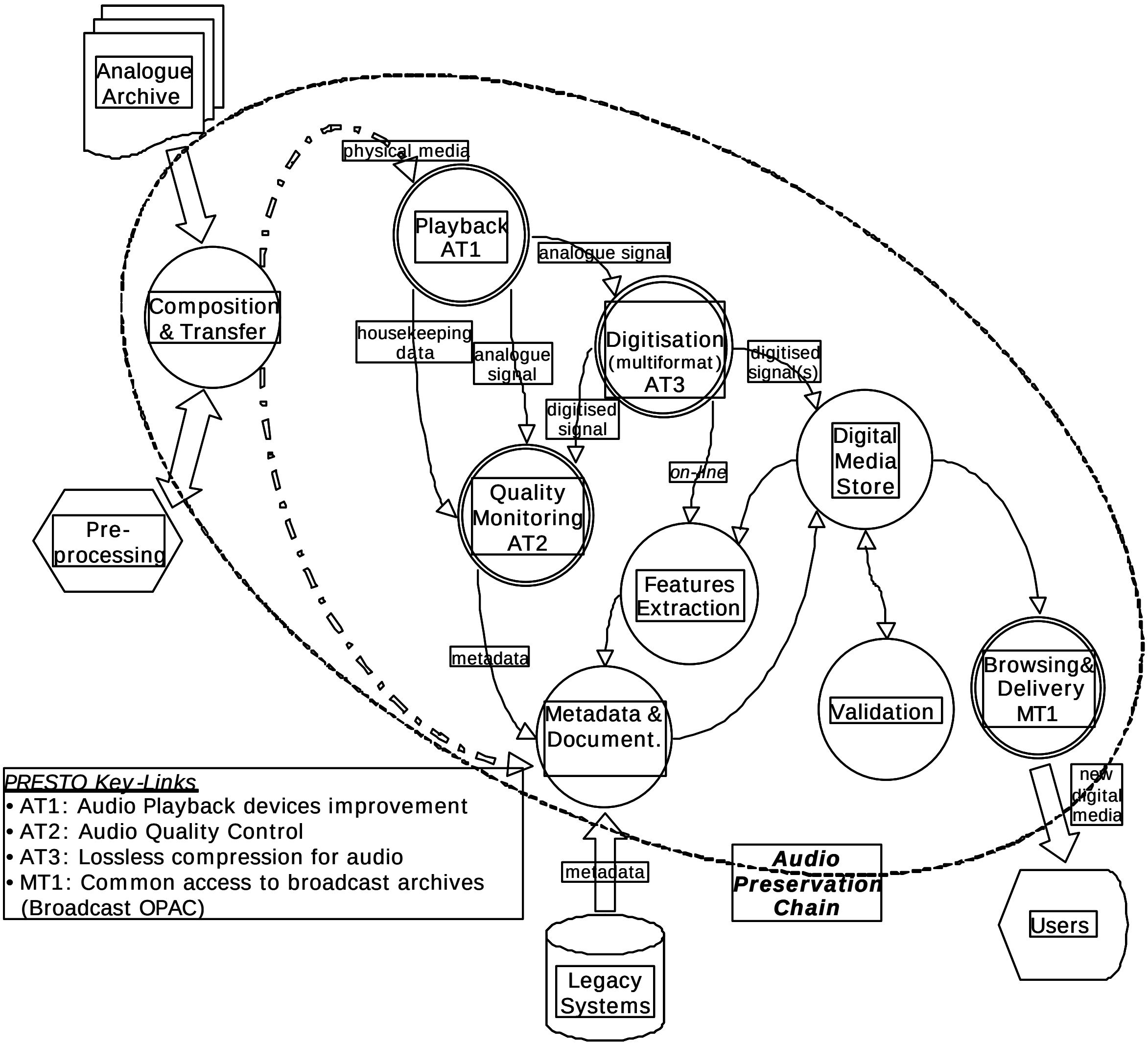

Figure 3.1: Key links for audio preservation

3.1.3 Film preservation

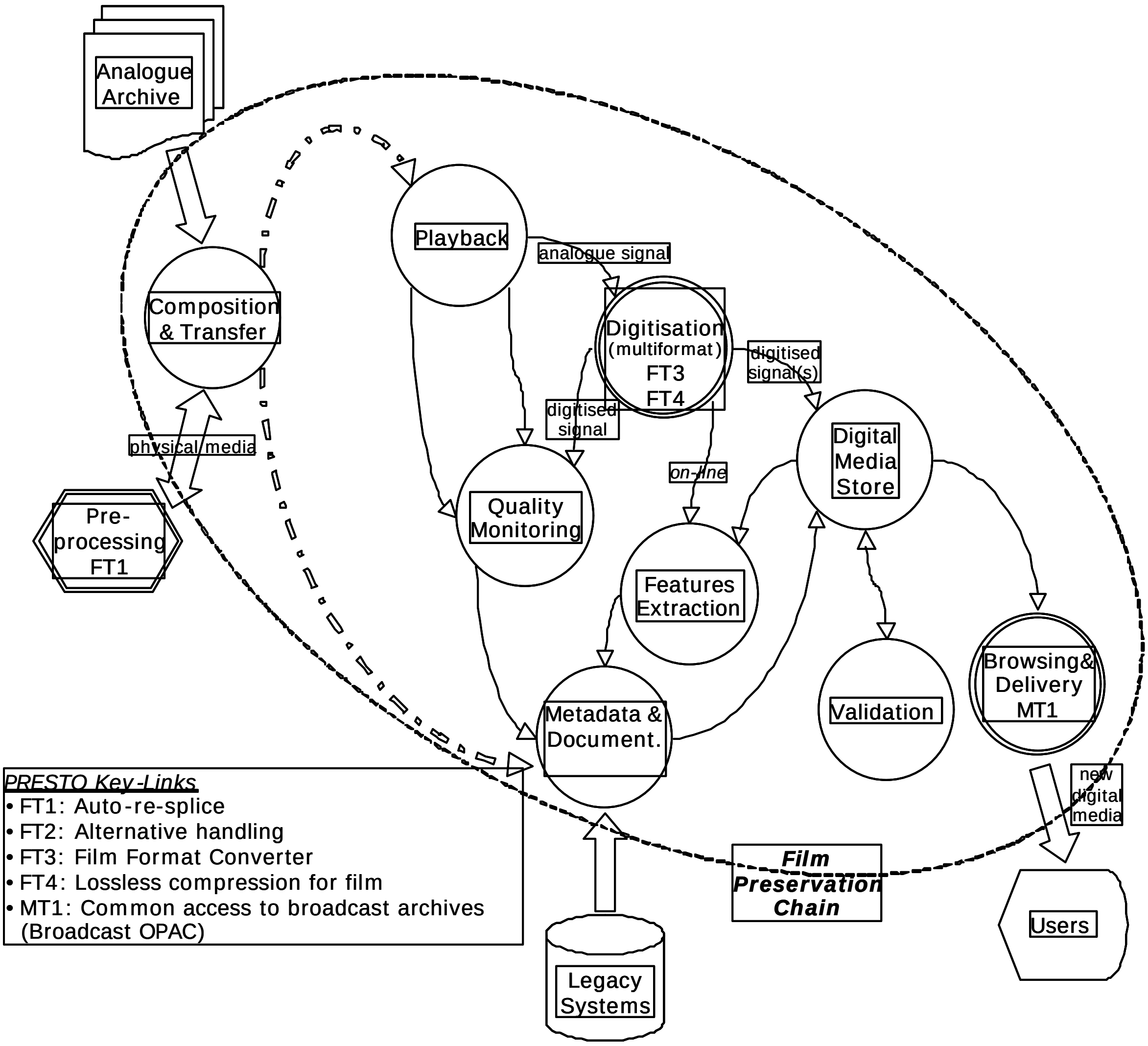

Figure 3.1: Key links for film preservation

3.2 Preservation chains infrastructure

The *PRESTO preservation chain infrastructure* is part of the overall broadcaster's content management system. It is conceived as a distributed solution framework, co-operating to the production and management of media assets, by:

- Providing means to store, organise and retrieve assets
- Providing support to facilities implementing the preservation steps in accessing the assets archived, to manipulate them or derive information
- Supporting all relevant steps in the work flow

In order to satisfy such needs, the preservation chain infrastructure architecture is constrained by the following requirements:

- It has to be open-ended, providing well–defined interfaces in order to facilitate the integration of legacy systems
- It has to be flexible and component based, providing a clear specification of the functionality that needs to be supported by each of its components.
- It needs to be a distributed to allow integration of systems at different physical locations and to inherently provide scalability

The Main target of an effective content management system is to maximise the re-use of archived material, reducing the cost of new production. The quality of the results depends on the quality of the metadata available for the material — especially on the quality of the cataloguing process. The PRESTO preservation chain provides tools that automatically extract a substantial metadata set from the content itself by computer–based content analysis. The preservation infrastructure should also support the creation of the most important metadata at all — descriptions entered by the experienced, highly qualified professionals working for the broadcaster, that can ensure the required quality of the descriptive information.

When starting the preservation process, a large amount of metadata is typically already available, created and entered by the archive or cataloguing. This information must be transferred to the new digital archive. As discussed in 0 , Metadata and documentation, the preservation infrastructure has to help in collecting and keeping metadata during the full preservation cycle.

This puts additional requirements to the preservation chain infrastructure:

- Automatically generate as much metadata as possible in order to enrich the documentation without increasing the human work load
- Get involved throughout the full media usage cycle by providing applications supporting the various steps in the cycle
- Preserve the integrity of metadata while the content moves along the cycle and enable the cataloguing staff to easily evaluate and modify all metadata as required.

The requirements mentioned can best be solved by a highly modular system architecture, that

- is extensible by adding modules,
- ensures investments by allowing to replace modules when technology improves, and
- is scalable enough to meet the ever growing demands

The infrastructure architecture should also integrate third party solutions by abstracting from proprietary API's and interfaces, so that these systems behave like native components of the infrastructure. All interfaces provided should be specified using a standard description language, e.g., the Interface Definition Language (IDL, ISO/IEC 14750 | ITU-T Rec. X.920). Wherever available the system shall build on open standards or recommended practices.

The modular approach to system design also requires that each module to be integrated is defined by:

- a clear interface design specification (IDS) and
- a clear functional design specification (FDS).

This way, the functionality of the system can be enhanced by providing new modules, together with a definition of the jobs the new module can perform, and a work flow specification that identifies how the new component needs to be addressed in a complex process.

3.3 Databases and servers

Preservation chain database and servers physically store and access media, metadata and control devices.

- to handle computer readable copies of essence objects, kept in a distributed storage environment,
- to control studio equipment and other devices, and
- to access and query metadata stored in heterogeneous distributed information systems

The data management system handles access to all metadata storage systems that can be used to identify content during retrieval. Metadata is either introduced automatically by services of the service layer or entered manually via the applications.

Often, these components will interface to existing solutions on the market that have proven their stability and performance. Hence, Foundation Layer components typically are third party components, integrated into via suitable CORBA wrappers.

Local metadata databases - The data structure should reflect the business needs of the organisation and hence should be targeted towards support of business processes, not purely towards persistent storage of metadata. A good example for such a data structure is the BBC's Standard Media Exchange Framework (SMEF™), which also is input to the EBU project P/META.

Full text search - To enhance the full text search capabilities of databases, a user configurable set of attributes should be capable of indexing for full text retrieval using off-the-shelf full text search engines.

Legacy databases - In many cases organisations that want to introduce a content management system already have an existing documentation system or catalogue installed that is widely used within the organisation. Often these systems store metadata describing a large part or even all of the assets of the organisation. They need to be interfaced and often even be treated as the master database.

Rights databases - The CMS needs to provide a possibility to integrate rights management solutions into the overall framework. This can be done, e.g., by considering a rights management system to be a specific kind of legacy database that provides rights related information.

Chapter 4 Key Links Technology in Video Preservation

4.1 Video quality control over digitisation process (VT1, VT3)

4.1.1 Introduction

Context

As previously seen in documents (WP2 and WP3.1), the preservation process is a succession of processes tied together. The transfer of contents from analogue media to digital media – maintaining quality, is the key step of preservation and preserving.

Although quality checking currently occurs at different steps of the preservation process, the majority of quality control is performed in a specific step, viewing and listening to the material. Human work is identified as the major cost driver for quality control. Currently, one person monitors one transfer. The mean effort rate is 1,5 hours of monitoring per hour of programme).

Purposes

In the preservation process, problems may come from:

- The physical status of the media
- The recording status on the media
- Dysfunction of playback
- Dysfunction of decoding and digitisation
- Dysfunction of the new recording system

Quality control has 3 main objectives:

- Ensure That original artefacts have not been amplified by playback, by digitisation or compression …
- Ensure that the transfer has not brought its own artefacts and that it was made in the best possible conditions.
- Ensure that the new recording can be the “new original”
- Reject failing transfers.

There are 3 ways to reduce costs:

- Perform all quality control during a linear transfer, thus avoiding a further specific step. This means that failing transfers have to be detected and rejected. Quality control system should help to decide if the transfer can be retried or if the transfer should be made in an expert chain.

- Quality control systems should be as automated as possible. Automating quality control will allow the operator to survey more than one transfer at the same time (until 4 chains max).

- Assessing and collecting quality information (quality metadata) during the transfer can add value to the asset and may help for re use (will an asset require further restoration or not?)

The future quality control system will be:

- A more or less automated tool “listening” to what is happening on the transfer chain. Once the transfer is started the system will automatically survey several predefined parameters or information coming from different parts or devices of the transfer chain that characterises dysfunction of the devices or impairments that are happening during the transfer. Parts of the control tool may be specific of a particular device.

- A tool that is able to deliver a global quality assessment of a programme or that can help in providing it. For this purpose, it analyses the most important parameters of picture and sound that could help to qualify the programme

- A reporting tool able to deliver an indexed report (time code) of the problems occurring during transfer. It will list impairments that may cause the transfer to be stopped, and impairments that are important or interesting to keep. The tool will also indicate the degree of seriousness of the detected impairment. A global quality assessment report will be also available.

The following specifications will be adapted to ¾ inch cassettes for massive preservation. Although part of the future system will be adapted to a specific playback device, the system should be easily applied (with little adaptation or development) to other formats as 2 inch, 1 inch Beta, SP.

4.1.2 Analysis of current practices

Control during preparation phase

Cleaning, heating, visual inspection of the media are the first steps of the process and contribute to quality checking, delivering information about physical status of the media, given by cleaning machines, or only seen. The operator can check if:

- **The tape is dirty or deposits**. Currently, the cleaning of tapes is systematic and should prevent head clogging. Unfortunately cleaning the tapes may not be sufficient. Current experience on ¾ inch transfers lets appear that a preliminary cleaning is not necessary efficient. Sometimes, a complete play back on a VTR offline and rewind before real transfer seems more efficient. This is not a satisfactory solution but one of the ways operators have found, to successfully perform the transfer. Heating or baking is usually required for sticky tapes but rarely used. Baking stabilises coating and removes humidity.

- **The tape is physically damaged.** During the cleaning step, it is possible to survey the physical status of the tape, edges winding, easy operation of the tape… for example, sometimes, adhesive tape at the beginning of the tape breaks and needs to be repaired manually. Checking strength of the adhesive tape may prevent problems at the transfer. A physical survey of the tape is really important to avoid irreversible damage during playback.

Control during the transfer phase

Safety of the tape and best playback is required. Before the transfer, an operator usually checks the VTR that will play the tape. Then he checks:

- If the play back at the beginning of the tape is OK
- The tracking and the RF level
- If reference signals (bars and tone) are present, playback levels are checked. If they are not present, video and audio levels are checked at the beginning of the programme.
- Audio tracks allocation switching to the right channel to digitise.
- The presence (or not) of time code or the "horloge parlante" on original recordings(specific
- The exact location of start and end of recording (this operation gives to the operator the exact duration)
- He checks if the content is the right one and corresponds to the description given in the database for this item.
- He rewinds the tape.

All previous steps may last 5 to 10 minutes per cassette/tape and are performed on the VTR that will play the tape.

- Then he can start the transfer.

  The best VTR with optimal set-up is required. It is well known that some VTRs are more tolerant than others are from another trademark or type.

- The operator surveys the transfer. Because he cannot stop the transfer, except for major failures he notes on the fly only major artefacts he is seeing and their location. (if really seen):
- Head clogging
- Saturation
- Losses or variation of tracking
- Playback synchronisation losses
- Lack of video and or audio
- Global quality of audio and video levels (saturation, very noisy pictures, distortion …)
- Drop outs (too many dropouts, bursts of drop outs, very large drop out…)
- Presence of black and white shots
- Loss of colour on picture (colour killing)
- End of recording

- Other impairments…
- Checks if decoding and digitisation are not introducing additional artefacts on video
- On digital video recorders, the operators survey the picture and listen to the sound in confidence mode.
- He checks the error rate of the digital video recorder (green orange and red LEDs indicators)
- Other major impairments seen during the transfer are noted.

Quality control phase

Quality control after transfer, occurs generally in a further and separated step:

- The digital copy is checked in a reference control chain.
- During this phase the program is played and a complete quality sheet is edited. The quality control operator has a reference list of impairments and notes them with their location. Sometimes it may be required to return to the original to check quality conformance.

Because there are too many programs to control, some companies have decided to check completely only samples in each set of tapes

Another solution is currently used:

- The continuous control of digital masters is performed while exploitation copies are made (for example the operator pushes pre defined buttons, each one corresponding to a major impairment).
- A list of the impairments and the related time code is updated.
- Operator may also mark a time code and cue the tape to this point when the transfer is completed.
- Operator checks the problem. According the nature and gravity of the problems, the operator can decide to stop the transfer and invalidate the digital master.

4.1.3 Typical transfer chains

First of all, it is worth noticing that the recording could be stored in a specific storage medium or as a generic file. In more detail, the recording could be:

- Video and audio streams onto a digital video tape. (digital betacam, IMX …). In this case, recording of digital audio and video is direct to the tape and recorded video can be immediately monitored (path1).
- Files onto a data tape or disk. In this case the video and audio files are stored on temporary hard disk and then backed up, to data tape or disk storage through a high-speed network (path 2). The back up system may be local (A DLT drive or a DVD writer for example) or distant. (a mass storage system with disk servers and libraries with HSM).
- Both ways (path 1 and path 2) may be simultaneously considered in the preservation process. The expected quality control system should be able to operate in the two cases.

Principles of the different paths:

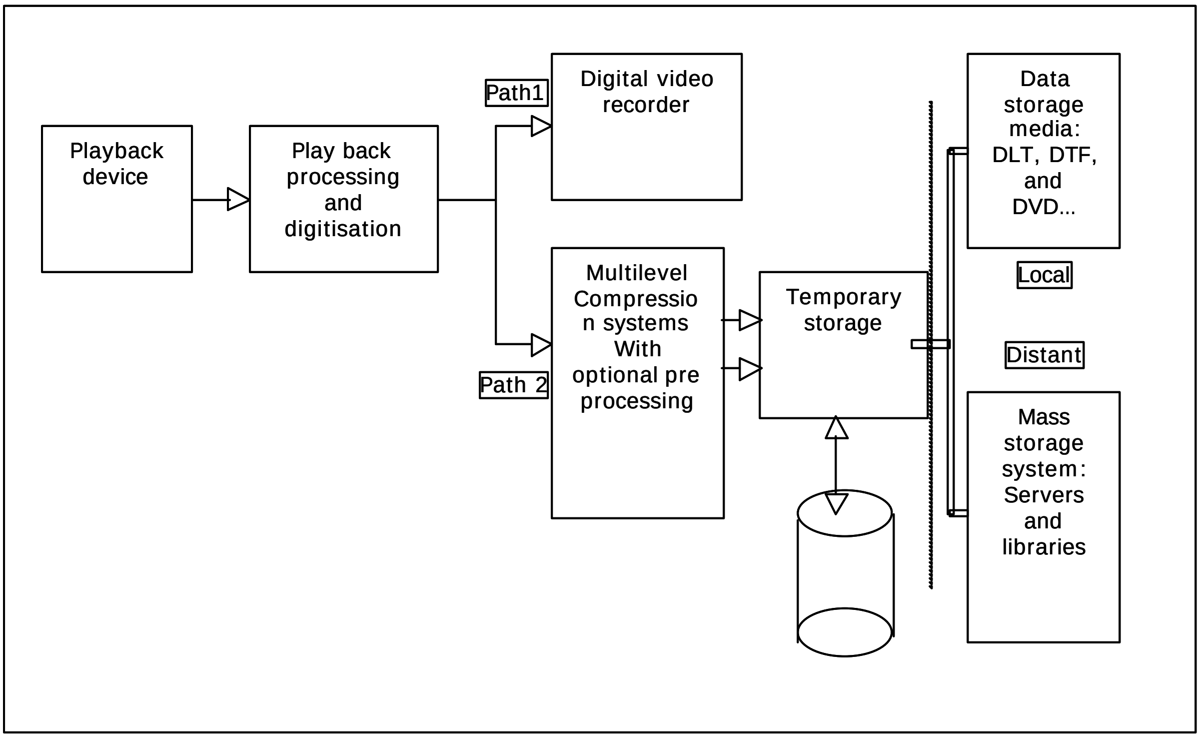

Figure 4.1: Typical transfer chains

4.1.4 Use case

An integrated system will allow the operator to perform transfers, quality control and final validation in addition to other tasks.

Usually, the operator receives the tapes and reports on their physical status. Tapes are bar-coded and the operator has all the information of title identification…

If this is not the case, cleaning physical inspection, bar-coding and information captures are required.

With the automated system the operator will:

Load the cassette and make initial set-ups: check of tracking, audio and video levels audio track allocation, check presence of time code. Then the operator re-cues the cassette and starts the transfer. Automatically the control and reporting tools will begin and the operator can proceed on another transfer chain.

The system will alert the operator to a major problem by sound or visual alarms (optionally the system will stop the transfer). The problem will be reported on the user interface and analysed by the operator. If too serious, the transfer is delayed and sent on to an expert chain. If not, the operator can restart after having corrected the problem (cleaning, modifying set-up etc.)

During the transfer, the operator may have a look at the report that is generated continuously. If he notes something out of the pre-listed impairments (or unknown by the system) he should add it manually to the report.

At the end of a transfer the operator will analyse the report and, depending on the seriousness of the impairments, may replay the new recording on the affected parts and add notes (manual quotation) in the reports, then he validates the new recording or not.

After validation the report is sent to a database.

An example of use case can be:

- From a BVU tape, a digital BETACAM (or everything else for example DVC pro) is recorded, and simultaneously a MPEG1 file and/or a MPEG2 file are stored.

- If there is no problem the digital video format and the files are validated then the files are backed up on a mass storage system. (DLT, DTF DVD, server.)

- If there are very few problems, the digital BETACAM can be continued on the transfer chain (by assemble editing). Then, in a further step, if the digital BETACAM is validated, the digital BETACAM can be played to generate the MPEG files.

- If many problems are encountered, the program will be processed on an expert chain (for example on an editing suite between BVU and digital BETACAM) then, the digital BETACAM can be played to generate the MPEG files.

The quality control system should be able to operate on these 4 situations and allow, at the end, the validation of the digital BETACAM, the mpeg files, the new media.

4.1.5 Failures and impairments to be detected

Contrary to other industrial processes where an input of a process is supposed to be without defect, the archive media is far from perfect and is affected by various problems. The transfer process will react differently according the original defect.

As previously mentioned, in the preservation process, various problems may come from:

- The physical status of the media

- The recording status on the media

- A dysfunction of playback

- A dysfunction of decoding and digitisation

- A dysfunction of new recording system

A problem could be:

- A failure in the transfer process itself

- A recorded impairment (existing in the content)

Identification of problems may come from or detected on:

- The cleaning device or “physical inspector”

- Various signals coming out from the play back device itself

- On the video and audio output signals

- On the picture and the sound itself (for example for recorded impairments)
- On the decoding and digitising devices
- On the recording devices

The following table shows impairments that may occur during the digitisation process and that should be reported and the device in which useful signals could be picked up and the nature of signals

Table 4: VT1,3-Failures and impairments to be detected

Impairments *(list subject to modifications)*	**Definition and effect on picture or sound**	**Signals from**	**Signals that can be used**
Problem related to the Physical media			
dirtiness	May come from dirt on the tape but from oxide particles too. One of the causes of Head clogging	Tape cleaner	No signal available. see head clogging
Damaged edges	When transport of the tape is not aligned, edges can be damaged. Audio tracks are on edges of tapes and may be affected.	Tape inspector Playback device	Signals or report delivered by tape inspection or cleaner. Detection of effect on audio signal?
Folded tape	A tape can be folded by a misaligned tape transport. Scratches appear on the picture and may last several seconds When a folded tape is played: The capstan servo unlocks. There are bursts of dropouts. Control track is absent or damaged. There is a temporary decrease of RF level. **This is an impairment to report, even if there is no way to correct it during the transfer, but, because the resulting defects will be hard to correct in a further restoration**	Tape inspector Playback device	Signals or report delivered by tape inspection or cleaner. Servo lock signal Drop out pulses Control track RF envelop duration
Head clogging	The head gaps are filled with dust or oxide from the tape according seriousness of head clogging, the RF level will be affected, dropout rate will increase. Severe head clogging may result in picture disappearance. Audio head clogging may affect the audio level and its bandwidth RF decreases progressively, drop out rate increases dramatically One head may clog before another.	Playback device	RF level / Picture content analysis/ Servo signals Drop out pulses Head identification or head switching pulses **Detection of beginning of head clogging will activate a cleaning system in the playback device (VECTRACOM)**
Playback or original recording impairments			
variation of tracking shot by shot	Recording has been made on a tape recorder with misaligned servos or editing has been made on different recorders with different tracking setups. Noise or noise bands appear on top or bottom of the picture. RF level changes. Contrary to head clogging, RF level changes straight after the edit point. **If such severe variations are detected with a damaging action on picture The only way to operate, is to perform transfer on an expert chain**	Playback device	RF level Capstan servo

Table 4: VT1,3-Failures and impairments to be detected

Impairments *(list subject to modifications)*	**Definition and effect on picture or sound**	**Signals from**	**Signals that can be used**
Off locks	This occurs if CTL (control track) is missing, if there is a break or discontinuity in CTL or if reference is lost during playback. Crash record between two sequences. Damaged Tape		Servo lock signals CTL missing or discontinuous
Recorded off locks	Transmission breaks Playback off lock while dubbing Sync signals and play back of the VTR is OK but the video signal is disturbed: vertical rolling of the picture Severe horizontal shifts during several frames	Picture capture	Picture content analysis
dropouts	Tape is in poor condition (looses oxide) Too used tape Heads are clogged while playing	Playback device Synchroniser with DO corrector	DO pulses RF signal
Recorded drop outs	Occurs when the played Tape is a dub of an original tape with uncorrected drops. This can occur for example during a tape to tape editing process editing	Picture capture	Picture analysis
Video levels out of range	Video exceeding 1 Volt peak to peak. Picture has severe losses of dynamics in whites. an excessive video input level may cause unwanted artefacts in compressed streams or A/D conversion.	Playback device	Video output
Chrominance level out of range	When chroma level is too low, mainly on identification bursts, problems may occur on decoding		Video output Bursts
Audio levels out of range	Overloaded audio at play back. Saturation of sound. losses of details and sound is distorted. An excessive audio output may cause unwanted artefacts in the audio stream	Playback device	Audio output
Saturation of video (recorded)	Occurs when video has been recorded with clipped video levels but also can be affected by horizontal black stripes on high frequency transitions. This may occur when the record current level was excessive or recording was made with too much used heads	Playback device	Picture analysis
Saturation of audio (recorded)	Sound has been recorded with saturation and is originally distorted even if the audio output level is OK	Playback device	Audio analysis
Video missing	Video can be missing on the tape for several reasons Severe losses of oxide but also recorded breaks during transmission. Error of manipulation of the operator	Playback device	Video output RF CTL Servo signals
Audio missing	Same reasons than video missing	Playback device	Audio output

Table 4: VT1,3-Failures and impairments to be detected

Impairments *(list subject to modifications)*	**Definition and effect on picture or sound**	**Signals from**	**Signals that can be used**
Time code missing	Break or absence in the TC	Playback device	Time code out Time code presence indicator of the device
Flagging, hooking, skew problems	This is a defect due to bad tape tension or heads that are not perfectly at 180°	Playback device	Servo signals?
Decoding artefacts			
Colour losses			Colour Killer signals? Chroma level
SECAM anti phase	SECAM decoding may lead to magenta pictures if 4 fields sequence has not been respected in editing		
Compression artefacts			
DCT blocks	Also called blocking		Picture analysis
New recording			
BER Bit error rate	On digital recordings there are concealment systems for lost information If BER is under a step value output picture may appear good If not, the picture may be affected of artefacts, audio may mute ... The BER varies if heads are clogging and if the tape is damaged according number of uses	Recording device	ISR (Sony's report status) LED indicators Auto inspection results ...
Integrity (file inspection)	Encoder fails during transfer	Storage device	
Global picture quality assessment			
Noise on picture S/N ratio	Noise can be on the video signal and measured on a black line in the vertical blanking Noise is part of the picture and cannot be measured on the video signal	Video recorder output De compressor	Picture analysis Noise on video signal
Noise on audio			
Mean video level And peak levels Mean audio level And peak levels		Video recorder De compressor	Video output of recorder Decoded streams
Drop out rate	statistics		
Number of off locks	statistics		
Audio bandwidth	Will assess average spectrum width	Recorder	Audio output
Video bandwidth	Will assess if picture is detailed or not	Recorder	Video output
Conformance to standards	Checking the conformance to the norm. If encoded streams are not fully compliant to the norm, future decoding may be impossible or difficult by some decoders		MPEG streams Use of specific analysis software

Table 4: VT1,3-Failures and impairments to be detected

Impairments *(list subject to modifications)*	**Definition and effect on picture or sound**	**Signals from**	**Signals that can be used**
Conformance of time code and synchronicity of multiple streams encoding	Automatic checking that same picture has same time code and same I frames on different streams of different encoding formats		Specific software to develop
Other Useful automatic detection			
End of recording	Alarms the operator or stops the transfer		No RF No video No time code duration
Tracking (optimisation)	Auto track at the beginning of tape		RF level

4.1.6 Functional requirements

The system will ensure:

The capture and processing of various signals:

- The capture and the processing of specific signals output from Playback device or picked into the device: RF, servo signals, dropout pulses …(see list in table) reshaped and converted into usable digital data

- The capture and the processing of analogue video signals, audio signals and time code, analysed and converted to usable data

- The processing/ combination/analysis of different detections: for example, to match and address a specific impairment, a given impairment could be identified by one or by cross checking several detected values. A global qualification of the program can be got in the same way … these processing or analysis could be processed by specific hardware or by software. In some cases real time is required

The system will:

- Include sets of reference values per impairment: for example max number of dropouts/minute, max acceptable video level, min value of RF, reference S/N ratio, min or max duration…

- compare detected values with references values and evaluation of the seriousness of the problem.

- deliver flags addressing detected impairments and their location (recording time code and playback time code if available).

The system will deliver a report (operator's interface), gathering and displaying "sub reports":

- A first sub-report will list continuously detected impairments as events with their identification (name), location, and degree of seriousness

- A second sub-report will monitor quality evolution of the program (Video and audio levels, phase, peak drop out rates and duration etc… and finally delivers a global quality status re-

report. A continuous display of the global quality assessment is not required but may occur every 30 seconds (to be defined). History of the quality assessment should be stored.

- The operator should be able to add a non listed impairment to the first sub report during transfer, and add on a comment about global quality
- The report should be editable and exportable to a database and tied to the new media

The system will allow a validation function of the successful transfer, activated by the operator.

The system will be able to import information coming from legacy database or from another media and technical database. They could be imported and included in the report (title, identification numbers, and bar code information….)

The system should be able to import data coming from the preparation step (if available).

In addition, a video and audio monitoring system should be built for a global view on the transfer (switching or multi views, or windows …)

4.1.7 Hardware architecture

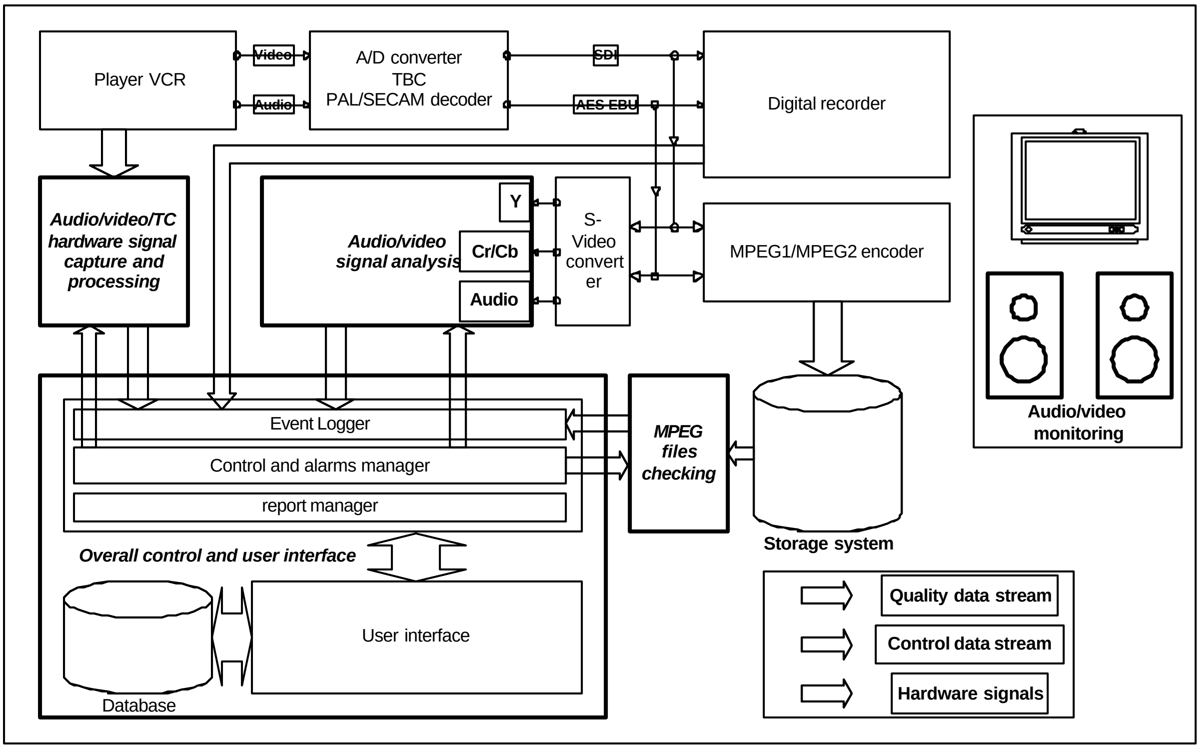

Figure 4.1: VT1,3-Hardware architecture

The system will have 4 main components:

- The soft real time **<AUDIO/VIDEO SIGNAL ANALYSIS SYSTEM>** is a computer sub-system with specific hardware (audio/video PCI capture card) to capture and process audio and video signals. For a given type of digital recording and compression profile, artefacts are related to input quality signal and to its spatial/temporal characteristics (such as frequency content), so its main function is to analyse audio/video content before compression or digital recording and report a set of values, recorded defects, and interpre-

interpretation on audio/video quality. SDI to PAL S-Video or RGB converter should be provided to meet the input requirement of the video capture card.

- The **<MPEG FILES CHECKING SYSTEM>** is a computer sub-system (but it could be hosted by the storage system). Its main function will be MPEG1 and MPEG2 files synchronism and integrity checking. A subsidiary function of MPEG conformance checking should be used.

- The **<AUDIO/VIDEO/TC HARDWARE SIGNAL CAPTURE AND PROCESSING SYSTEM>** is a sub-system based on a micro controller device to capture and process various signals picked on connectors of the playback device or picked from the device. Its main function is to dynamically adjust VTR settings to provide the best quality for reproduction of current videotape. During playback it monitors electrical audio, video and hardware signals and reports continuously all measured values and playback defects.

- An **<OVERALL CONTROL SYSTEM>** is used to control all sub-systems. It receives all quality data and displays graphic user interface.

A standard video and audio monitoring system is also required.

4.1.8 Software architecture

The **<OVERALL CONTROL SYSTEM>** will be the manager for other quality control subsystems. It will receive data about quality, log them and display a synthesis of them. It will produce alarms on their combination and a final report in a database. It will host a local database but a distant database system should be used. It will offer to the user the graphical interface to:

- Define a set of monitored parameters and a set of alarm conditions on them.

- Display continuously quality control parameters during transfer and a quality assessment of transferred programme.

- Produce sound or visual alarms on user defined conditions.

- Add on the fly in the log list, any visually detected impairment.

- Store, display and edit current and previous quality reports.

The **<AUDIO/VIDEO ANALYSIS SYSTEM>** will be a soft real time, “black box” subsystem (no user display), starting and stopping to capture and analyse audio and video content on request of <OVERALL CONTROL SYSTEM>. It will send audio/video analysis data continuously and at defined intervals an audio/video signal quality assessment. All data will be time stamped with a frame number, counted since the beginning of the analysis session. Communication to and from the <OVERALL CONTROL SYSTEM> will be provided by network interface.

<HARDWARE SIGNALS CAPTURE AND PROCESSING SYSTEM> will be a real time, micro-controller subsystem dedicated to VTR electrical signals capture and processing. It will monitor continuously electrical values of these signals and will modify VTR settings in order to guarantee the best quality for tape reproduction. It will send continuously all measured values, playback defects, time code and VTR status to the <OVERALL CONTROL SYSTEM> through a RS232 interface.

<MPEG CHECKING SYSTEM> will be “a black box” subsystem connected to the storage system or a specific process hosted by storage system. It will start on request of the <OVERALL CONTROL SYSTEM> at the end of transfer. It will check MPEG1 and MPEG2 files, searching on image con-

content and GOP header to verify time code synchronism and duration of record. It should provide a brief checking of MPEG standard conformance. It will report a logical indicator (YES/NO), at the end of its process. If it could not be hosted by the MPEG storage system, its MPEG file access should be provided by the network interface. Communication to and from the <OVERALL CONTROL SYSTEM> will be also provided by network interface.

4.1.9 Hardware items

Overall control system

<The Overall control system> will be based on:

- PC PIII 1Ghz processor, 512 Mo RAM, 1x 10Go UltraSCSI 3 system disk + 1 x 36 Go Ultra SCSI 3 data disk. 100Mb/s network interface card. Graphic adapter card and 21" monitor supporting 1280x1024 display.

MPEG file checking system

The <MPEG file checking system > will be based on:

- PC PIII 800Mhz to 1Ghz processor, 512 Mo RAM, 1x 10Go UltraSCSI 3 disk. 100Mb/s network interface card. Standard graphic adapter card and monitor.

*For information only, because it will probably operate on another hardware item.

Audio/video analysis system

The <Audio Video analysis> system will be based on:

- PC Dual Processors Pentium III 933MHz or higher, 1GB RAM, 1 HD 36GB UltraScsi3, Windows 2000 Professional or Linux RedHat 6.2 and supported frame grabber able to digitise S-video signal.
- A audio digitising card (e.g. SoundBlaster Live)
- A 100Mb/s network interface card.
- Modern standard graphic card and monitor.

Hardware signal capture and processing system

The <Hardware signal capture and processing system> will be an "on board" device on the player itself and will be based on a micro controller with a minimum of:

- 32 in /out (TTL).
- 8 in/out analogue to digital and digital to analogue converters.
- 2 serial ports.
- A keyboard and display device.

The system will require minimum mechanical and electrical connections.

Block diagram of the system

Architecture developed and provided by VECTRACOM

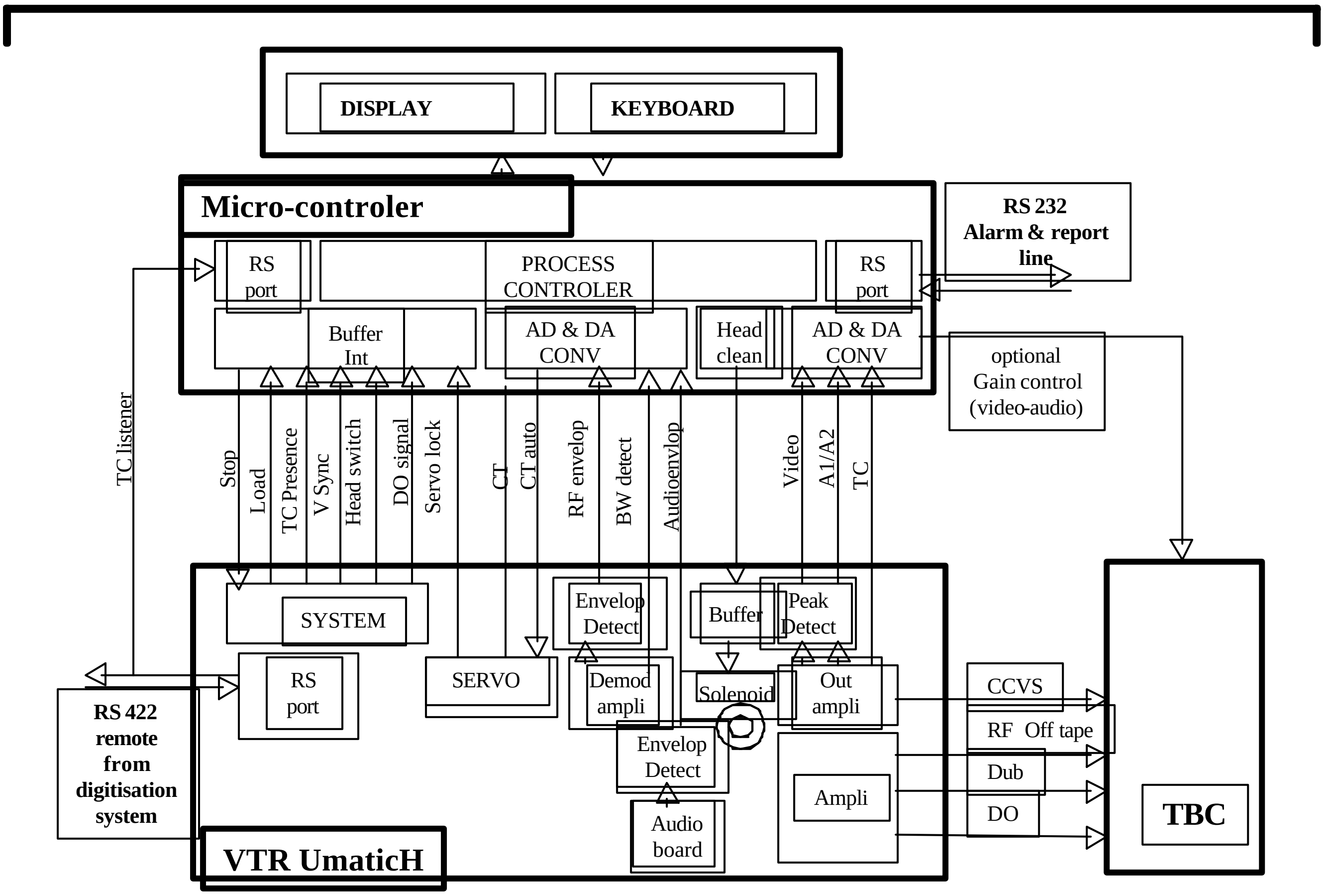

Figure 4.1: VT1,3-Block diagram of the system

Signals, data collected from the VTR

Many signals can be extracted from the VTR. A limited number of signals will be used. This is, of course, a non-restrictive list (*signals written below in italics are not considered for the moment but may be*). The micro controller can handle more signals.

With these signals or their combination it will be possible to detect and qualify the major part of the impairments listed before.

All the used signals or data have always a default, a minimum and a maximum value. For example the default value for video (peak) level is 1Vpp, or DO (drop out) number and duration are under a value of xx per second...

Table 5: VT1,3-Signals for synchronisation or real time management

Vertical sync	<Vsync>
Time code (serial)	$ TC
Head switching (odd/even field)	<headsw>
Play	<play>

Table 6: VT1,3-Analogue signals periodically sampled

RF (or RF envelop) with possible distinction of head	(RF)
Control track	(CT)
Video out (or peak envelope)	(Vid)
A1/A2 output	(An)
Time code (analogue out)	(TC)
Tape beginning sensor	
Tape end sensor	
Tape tension sensors	

Table 7: VT1,3-Logic signals periodically sampled

TC presence	<Tcpre>
Play on	<PLAY>

Table 8: VT1,3-Binary signals used as interruption

DO (drop detected)	<DO>
Servo lock (unlock)	<servolock>

All these signals are collected on the back plane of the Umatic VTR or on plugs available on the rear panel.

Data, commands resulting from the micro controller process

A lot of information or commands can be issued from these signals and/or any combination through the micro controller and the software.

Basically there are 3 types of commands or signals:

- Commands returned to the Umatic, its associated TBC or any external device designed for audio or video processing. The aim of such commands are to activate corrective processes. For example: activation of an additional head cleaning system (to be developed and installed by VECTRACOM) installed into the VTR. Another automatic action could on tracking adjustment or simply activation of a stop command if too many errors or troubles are detected. Eventually, activation of an automatic gain control driving the external TBC or any audio processor device could be considered.

- Signals transmitted to the <OVERALL CONTROL SYSTEM> through an RS 232 interface according a simple protocol. Based on Time Code extracted (or interpolated) from the Umatic, the objective and measurable data is collected, computerised for interpretation, and reported into a database. Data that can be transmitted are: drop out quantity, dropout duration, servo status, RF level, audio-video levels out of range, information regarding the edges of tape (if damaged or folded).. ..

- Assistance to the operator for the adjustment of the Umatic during test phase before launching the digitisation process (i.e. CT adjustment, audio track assignment...) or also during the process in case of troubles (servo lock, DO rate, out of range levels, BW sequence....). A larger range of alarm could also be imagined (tape tension, skew, humidity...). Alarms Assistance information or alarms are given to the operator through a small display or may be a buzzer signal.

Commands or data transmitted:

A command has always a default value, a minimum, a maximum and an error value (respective to the default value).

For example the head cleaner system (installed in the VTR) is set to off, head cleaner is turned ON only when needed.

Default value for video level is the value needed to have 1Vpp on the output of the device controlled (i.e. TBC). These analogue values are used only if an external process is done as for example an automatic gain for the video level...

Table 9: VT1,3-

CT error	(CTerr)	(CTdef)	(CTmin)	(CTmax)
Video level	(Vidleverr)	(Vidlevdef)	(Vidmin)	(Vidmax)
Audio n level	(Anleverr)	(Anlevdef)		

Required information on RS232:

Information or alarms are displayed on the micro controller screen and/or send to <OVERALL CONTROL SYSTEM > through the RS232 line.

Information send to <OVEREALL CONTROL SYSTEM>will be in a simple protocol whose exact format cannot be completely defined now. It should be based on following format:

`TIME|SIZE|ERRORS_FLAG|VTR_STATUS|[LABEL1|TYPE1|VALUE1|[LABEL2|TYPE2|VALUE2]…].`

- TIME will be the current VTR time code in ASCII or binary form (transmitted on every frame).
- SIZE will be the current stream length in bytes (1 byte coded)
- ERROR_FLAG will be reported on 1 or 2 bytes.
- VTR_STATUS will be reported on 1 byte (binary coded).
- LABEL will be the binary coded value name (1 byte).
- TYPE will be A for ASCII and B for binary (1 byte).
- VALUE (1 to several bytes depending on coding and value range)

TIME, ERRORS_FLAGS and VTR_STATUS will be transmitted on every frame even though no defects are detected or values are transmitted (to reduce overall communication traffic, it should be more effective to transmit only values changed since the previous time). Header and checks should be defined and added to this protocol, to provide communication error detection.

Reported logical flags on RS422 and reported values on RS422 (digital values) will be as follows:

Table 10: VT1,3-Reported logical flags on RS422

Tape damaged edges	Flag <Edge>
Folded tape	Flag <Folded>
Tracking variation	Flag <trackerr>
Servolock	Flag <unlock>
Video level out of range	Flag <Vidrange>
Audio level out of range	Flag <Audrange>
Chroma level out of range	Flag <chrrange>
Video missing	Flag <Vidmis>
Audio missing	Flag <Audmis>
TC missing	Flag <TCmis>
Black and white detection	Flag <BW>
Flagging hooking	Flag<hook>

Table 11: VT1,3-Reported values on RS422 (digital values)

RF level	
Chroma level	
Audio level out of range duration	If audio level is out of range, cumulated duration per frame.
Video level out of range duration	If video level is out of range, cumulated duration per frame.
dropout pulses Number	If drop out pulses are detected, number of drop per frame.
Drop out duration	If dropouts are detected duration of the longest pulse per frame.

4.1.10 Software Items

Overall control system

It will be a Microsoft Windows application with user interface.

- Operating system: Windows2000 pro
- Development software tools: Microsoft Visual C++
- Software used will also include specialised tools to provide emulation, writing, loading and debugging of micro code on the <HARDWARE SIGNALS CAPTURE AND PROCESSING SYSTEM> micro-controller card, during software integration step.

Overall description

The management of the control quality system includes four main steps:

A “configuration mode”:

- Allow the user to define a set of parameters to be monitored during transfer, among all captured parameters.
- Allow the user to define nominal value, range and unit for each monitored parameters.

-

Software architecture

*Architecture developed and provided by INA

At present time, several modules could be identified but architecture may be slightly changed on further studies.

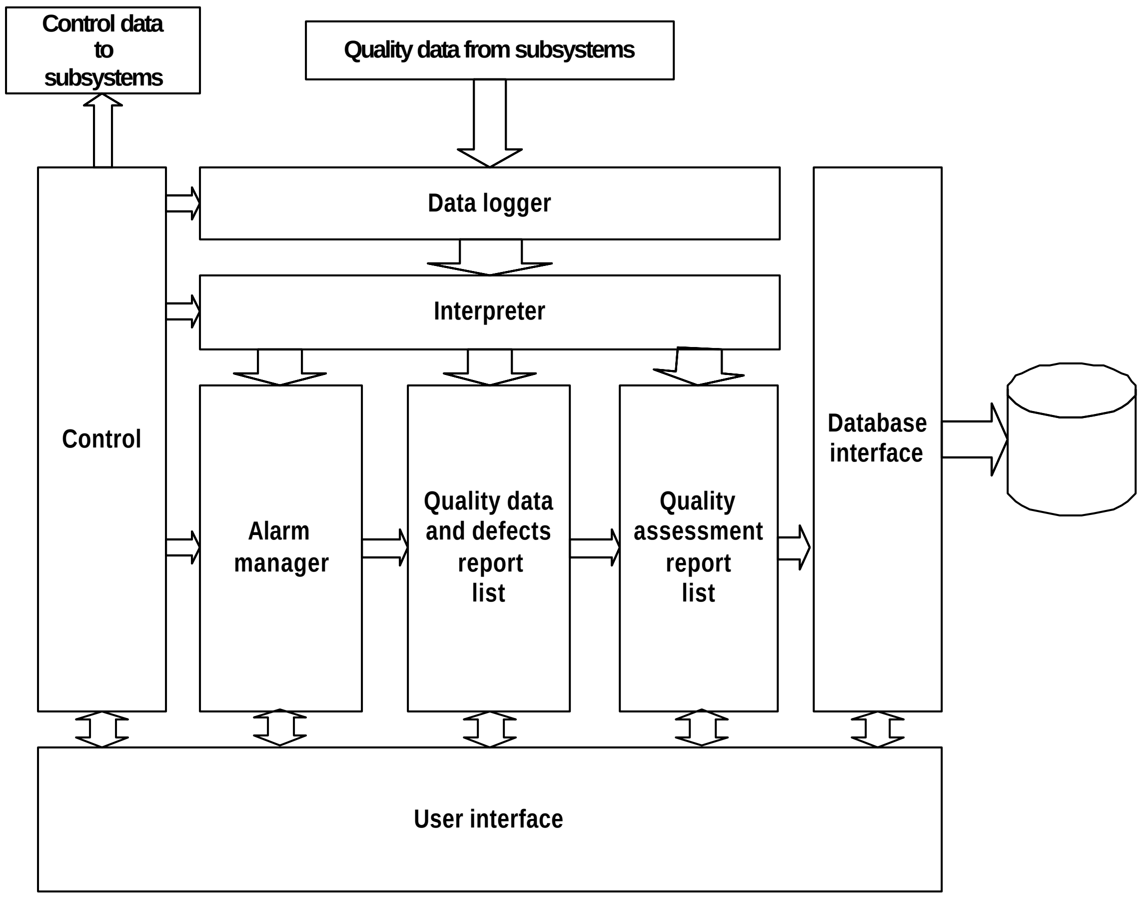

Figure 4.1: VT1,3-Software architecture

Control manager

This module will provide control communications with all other modules or subsystems through RS232 or network interface with their specific protocols. It will decode and dispatch transfer chain status to other modules and subsystems to provide overall synchronism.

Data logger

This module will receive all data streams from subsystems through RS232 or network interface with their specific protocols. It will extract time code from data transmitted by the <HARWARE

SIGNAL SYSTEM>. It will convert elementary time stamps of data issued from the <AUDIO/VIDEO ANALYSIS SYSTEM> and will tie and convert them to the same reference.

Interpreter

This module will provide a second level interpretation of basic data and defects provided by <DATA_LOGGER> module. It will be able to do a temporal or difference based filtering on elementary data received, a scaling operation and a combinative and/or temporal interpretation of them (It could be necessary to convert and store elementary data in intermediate formats - to be defined -) . To do that, it will also take as input, a “settings” file defining combinations, scaling, filtering on the basic data types and defect types returned by sub-systems.

This module will output two streams of data:

- basic data and user defined combination of them to monitor.
- quality assessment data received and quality assessment data deducted from basic data received..

Alarm manager

This module will receive <INTERPRETER> outputs to add severity qualifier or to produce alarm on parameters values or defects. It will take also as input a settings file defining severity qualifiers and alarms conditions.

It will output:

- sound and/or visual alarm through <USER INTEFACE>.
- severity qualifier (to be defined) to <QUALITY and DEFECTS REPORT LIST>.
- Alarm description and time to <QUALITY and DEFECTS REPORT LIST>.

Quality and defects list

Dynamic list of monitored parameters values, alarms an qualifiers will be implemented in this module. Add and edit of comments lines will be also provided.Various sorting on time code, parameters type, and severity will be implemented.

It will take as inputs:

- Monitored parameters data from <INTERPRETER> module.
- Severity qualifiers and alarm description from <ALARM MANAGER> module.
- Special impairments label and time outputted by user defined shortcuts.

Quality assessment list

Dynamic list of quality assessments issued during transfer, will be implemented in this module. Add and edit of comments lines will be also provided by this module. Sorting on quality assessment type will be implemented.

It will take quality assessment data from <INTERPRETER> module as input.

Database interface

This module will provide necessary 'glue' to and from local and/or distant databases. It will allow user to import and export external technical data and reports. Due to the interdependency with database structure used in the transfer chain context, it will be a DLL (Dynamically Loaded Library) module to provide an easy changeable module. For demonstration purposes we will connect the <CONTROL QUALITY SYSTEM> to a local or distant database (MySQL or Access) with standard SQL request commands. This database structure will be defined on further studies.

It will take <QUALITY DATA LIST> and <QUALITY ASSESSMENT LIST> objects as inputs and write formatted fields in the database on <USER INTERFACE> request.

It will read database fields to build <QUALITY DATA LIST> and <QUALITY ASSESSMENT LIST> on <USER INTERFACE> request.

User interface

As previously seen in the description of the < OVERALL CONTROL SYSTEM>, four steps of user interactions could be distinguished. Each will have a specific user interface design based on windows, scroll list, button, etc …with Microsoft standard look and feel. (Action will be only given by keyboard and mouse).

Currently, the design is a first draft of the main monitoring step; it could be modified according further studies.

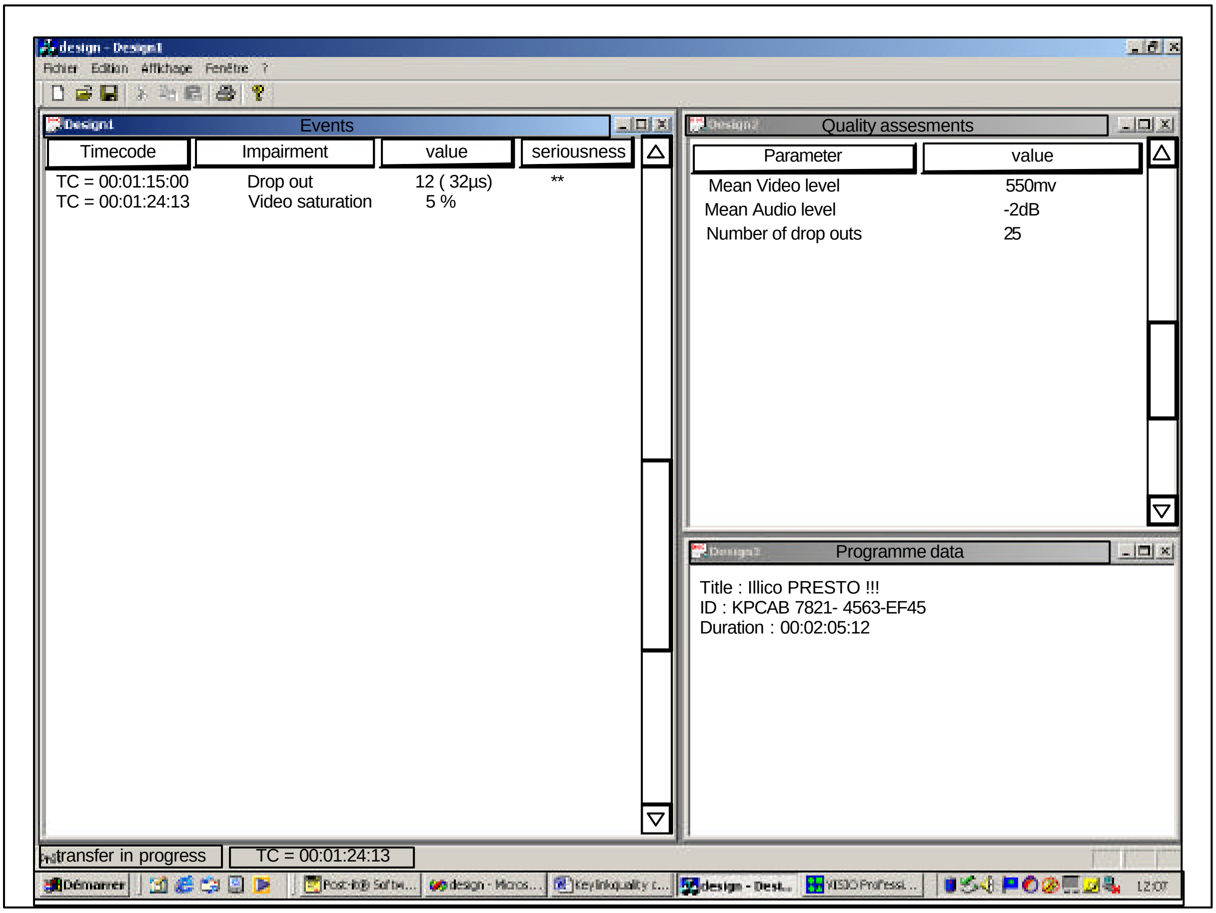

Figure 4.1: VT1,3-User interface

MPEG files checking system

It will be a process starting on request of the <OVERALL CONTROL SYSTEM>. Communications will be provided by network sockets. It will take as inputs the names and location of MPEG files and duration of transferred programmes. Depending on the hardware hosting this item, file access could be provided by network or direct access to storage disks. It will search along each file content for corresponding frames and stored time code. It will parse MPEG headers to check their conformance with MPEG standard.

Output will be an ASCII stream (to be defined) and will report status on MPEG standard conformance, file integrity, and synchronism checking (to be added on final report of <OVERALL CONTROL SYSTEM>.

Hardware signal system

For speed reasons, the programming language will be assembler. For maintenance and possible evolutions, an appropriate software modularity level is applied (one subroutine per task). As real time tasks are managed, the software handles priority interruption level.

All the programmed functions can be bypassed and all the signals or commands returned to the VTR will not interrupt the standard functions of the VTR.

The modularity of the software gives the possibility to add incoming signals if necessary as well to improve the number of impairment detected as to modify a corrective process with new parameters (or to achieve better accuracy for this process).

Definitive algorithms will be described later (after performance evaluation of a given and chosen micro controller).

Algorithms (examples):

According to the signals measured (periodically checked or managed as interruption) computer algorithms, the following commands can be deducted as described here under for example.

Algorithm for Head cleaning command (clogging):

- IF {(RF) decreasing according to a simple law (a percentage during an interval of time= number of <Vsync>) AND <DO> increase each frame} THEN activate <cleaner>

Algorithm for CT (help to the operator for tracking adjustment):

- DO (CT) = (CTdef)
- WHILE <play> is true THEN display on screen a graphic representation of (RF)
- MEMORIZE (RF) for all legal Value of (CT): (CTmin)<(CT)<(CTmax)
- LOOK for value of (CT) which maximise (RF)
- APPLY (CT) = (RF)max

Algorithm for auto tracking:

- Same as above, except that an iterative process is initiated when (RF) decrease suddenly (tracking error due to insert)

- SEND Flag <track err>

Algorithm for Tape damaged edges control:

- IF Audio level AND CT have fluctuation at frequency lower than 100 Hz THEN
- SEND Flag <Edge>

Algorithm for Video level control:

- IF Video peak greater than 1V THEN
- SEND Flag <Vidrange>

Audio/video analysis system

Developed and provided by ITC-irst

This system is a soft real time subsystem. It detects audio/video impairments, and provides information on audio-visual quality.

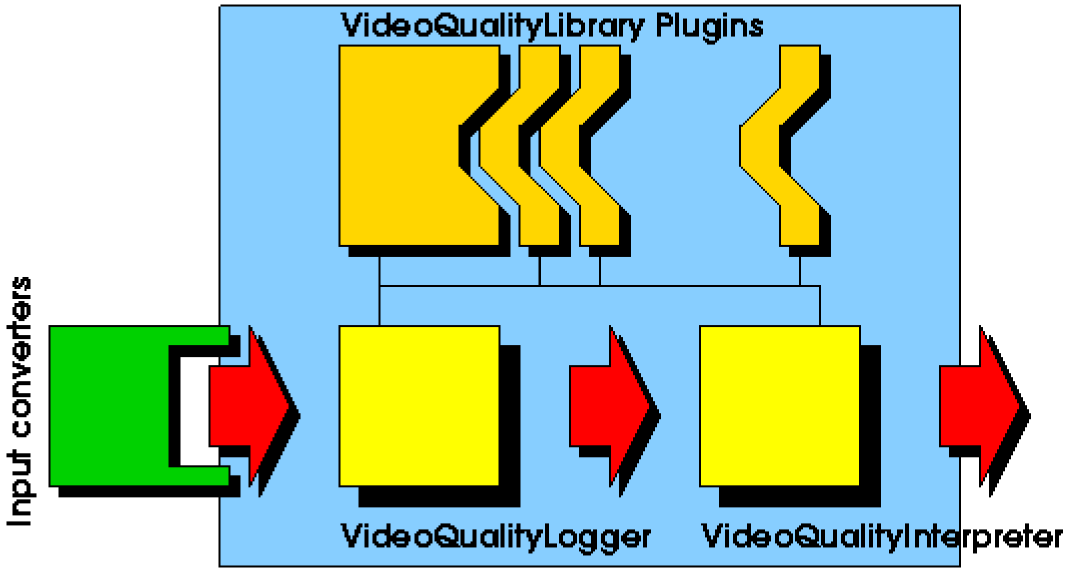

Figure 4.1: VT1,3-Architecture of the Video Quality Analyser modules

Video Quality Library

This module comprises a dynamic library for the (single) selected operating system and accompanying documentation and header files. Additional modules will extend the system relying on a plug-in architecture.

This item provides the basic image and video analysis functions needed to enhance PRESTO with additional capabilities in monitoring the quality of tape playback and assessing the quality of footage. In order to provide an easy path to future upgrades, the structure of the supporting library will be that of a set of plug-ins extending the functionality of the video quality analysis system.

This module provides an API for the developed Video Quality Analysis Modules. The API will include functions for several numerical transforms (wavelet, DCT), template matching algorithms

algorithms and specific image impairments detectors (dropouts, recorded saturation, offlocks). Support for some basic audio quality parameters will also be provided.

This module depends on the required operating system.

This module is required by:

- **<VIDEO QUALITY LOGGER>**
- <**VIDEO QUALITY INTERPRETER** >.

A dynamic library (with supporting header files) and interfaces with other modules by appropriate linking. The basic library will be extended through plug-ins to incorporate new functionality for the <**VIDEO QUALITY LOGGER**> and the <**VIDEO QUALITY INTERPRETER**>. In total there will be 30 MB of hard disk space.[1]

The rules and algorithms cannot be completely specified yet but they will most probably include wavelet/DCT transforms, spectral analysis, robust statistical estimators, template-matching techniques, and HyperBasis Functions classifiers.

Video Quality Logger

An executable function, (with corresponding source code) and usage documentation. This item analyses an incoming uncompressed digital video stream and/or digital audio stream to provide a low level VQ (Video Quality) profile containing preliminary information on video quality and supporting numerical data needing further analysis.

The module will take as input a digital video stream (uncompressed) and/or a digital audio stream, the ID of the stream (a unique identification of the item), the specs of the input streams, and will provide a binary stream containing information extracted from the (audio) video stream(s) such as:

- Wavelet and DCT transform coefficients for assessing S/N ratio

1

[ArKo98] S. Armstrong, A. C. Kokaram, and P. J. W. Rayner. *Restoring video images taken from scratched 2-inch tape*. In Workshop on Non-Linear Model Based Image Analysis (NMBIA'98), Editors: Stephen Marshall, Neal Harvey and Druti Shah, pages 83-88. Springer Verlag, July 1998.

[StMu98] J.-L. Starck and F. Murtagh, *Automatic noise estimation from the multiresolution support*, Publications of the Astronomical Society of the Pacific, 110, 193-199, 1998.

[Wi99] S. Winkler, *Issues in vision modeling for perceptual video quality assessment*, Signal Processing 78(2), 1999

[FiGo99] W.J. Fitzgerald, S. J. Godsill, A. C. Kokaram, J. A. Stark, *Bayesian Methods in Signal and Image Processing*. In J.M. Bernardo, J.O. Berger, A.P. Dawid, and A.F.M. Smith, editors, Bayesian Statistics VI. Oxford University Press, 1999.

- wavelet and DCT transform coefficients for assessing picture quality
- still frames classified as possible picture impairments but needing further analysis for proper assessment (be it from the module <**Video Quality Interpreter**> or a human supervisor)
- log of detected basic picture impairments (e.g. dropouts, recorded saturation, off locks)
- log of basic audio parameters (i.e. noise, bandwidth, saturation, silence)

Log of other impairments based on the availability of VQ plug-ins.

This module depends on:

– <**Video Quality Library**> and its extension plug-ins.

Modules interact with this item by providing a digital uncompressed video stream on its specified input socket/file (each frame is a stream of pixels, each represented by three bytes providing RGB components), and providing some control parameters in ASCII format directly on the command line:

Video Quality Logger
-m ID
-c <configuration file>
-a <audio input stream>
-v <video input stream>
-x <control input stream>
-l <log output stream>

The control stream will trigger appropriate actions from the <**Video Quality Logger**> and will operate as a remote controller (e.g. starting/ending audio/video analysis). The input audio video streams (based on sockets or files) will have the following structure:

4 bytes:	magic number
4 bytes:	data size
12 bytes:	time code info
4 bytes:	format ID
32 bytes:	decoder ID
4 bytes:	check sum
......:	dataChunk

Example: plain audio format data Chunk:

4 bytes:	sampling frequency
4 bytes:	number of channels
4 bytes:	sample precision
4 bytes:	number of samples
.....:	sample data

Example: plain video format data Chunk:

4 bytes:	width
4 bytes:	height
3 bytes:	rgb pixel data
3 bytes:	rgb pixel data
.....	"

The system will provide support for the use of multiple formats through its plug-in architecture. Alternatively, the program will control directly the audio/video-grabbing device for increased operation speed (the corresponding devices will be specified using the options for the audio video streams with a special syntax). Note that in this case, timing information is derived from the number of frames and system time, not from time-code information embedded in the audio/visual stream that will not be decoded. The program will provide on the output socket/file a binary stream whose exact format cannot be completely defined now. It will be based on a key-value format and the following format:

S000000000X|dd:mm:yy:hh:mm:ss:cc|Lsize|Label|Asize|Adata|Bsize|Bdata|

Where the first field (starting with **S**) will contain the size (zero padded to nine digits) of the remaining data plus a final checksum character, the second will represent a timestamp identifying the time when the information was produced. The third field represents the kind of information logged and the remaining fields contain the corresponding ASCII and binary data.

The structure of the VQ profile stream can then be parsed easily and extended for future necessities. ID information and further relevant metadata will be included in the stream using appropriate labels.

The module requires as input an uncompressed digital video such as RGB or YUV format and uncompressed digital audio. This program (and those providing the video stream) may necessitate extensive buffering capabilities and the program itself will be very computationally intensive. 30-MB hard disk space (expected).

For references, see <Video Library above>.

Video Quality Interpreter

An executable function (with corresponding source code) and usage documentation. This item analyses a low level VQ profile to provide a final ASCII log of the detected picture impairments and an overall quality score for the video item considered.

The module will receive as input a binary (low level) VQ profile and will provide an ASCII log of the detected picture impairments together with a global video quality score (including audio quality data) to supplement available metadata information on the corresponding item.

This module depends on <**Video Quality Library**> and its extension plug-ins.

Modules interact with this item providing on a specified socket/file a VQ profile stream and the name of a directory where the program will store frames whose interpretation can require human intervention (if required). An html file "ToClassify.html" with timestamp information and links to the frames will also be produced. The program will output (on its output socket/file) a new VQ profile whose overall syntax is similar to the incoming one but will contain only ASCII information and is expected to represent the final quality log file. The control-input stream will be used as a remote controller triggering appropriate actions from the <**Video Quality Interpreter>.**

Video Quality Interpreter

–i <input VQ log stream>

-d <dump dir>

-o <output VQ log stream>

-c <configuration file>

-x <control input stream>

The program will provide the following information on audio-visual quality:

- signal to noise ratio for video
- recorded dropouts qualified by the corresponding number of lines
- recorded saturation
- recorded offlocks
- presence of magenta dominance (SECAM specificity)
- signal to noise, saturation, silence, and bandwidth of the audio signal
- frequency content characterisation for image quality assessment and to signal possible mpeg encoding problems

As well as basic statistics derived from them, such as average and peak values over the duration of the analyzed signal and alarm information (in textual form) whenever these values fall outside a predefined range given by user selectable profiles.

This program may necessitate extensive buffering capabilities and the program itself will be very computationally intensive. 30-MB hard disk space (expected).

For information on Processing and references, see <video quality library> above.

4.2 Time base corrector with drop out detection and compensation (VT2)

4.2.1 Hardware architecture

This hardware will be produced as stand alone equipment and will be transparent to the picture content in the preservation chain. It will take the form of a dual standard PAL/SECAM decoder. The functionality will be to replace unstable synchronising pulses from the output of the U-matic tape machine used to replay the recorded archive. It will also detect and compensate for drop outs.

The sync time base instability may be either inherent in the recorded material or introduced as part of the transport mechanism of the U-matic player. The replaced synchronising pulses at the TBC output will be stable and may be locked either, to an internal crystal, or to the local studio network The decoder will function with a wide range of U-matic machines including Low-band, High band and SP (superior performance) formats.

The TBC will normally sit next to the U-matic source in the preservation chain.

4.2.2 Time base corrector with drop-out detection and compensation

The following specification defines the technical performance and interfaces of the proposed dual standard PAL/SECAM decoder containing the functions of Time Base Correction (TBC) and Drop-Out Compensation (DOC). This hardware will be suitable for general applications but is specifically designed to overcome as many archive deficiencies as possible when working with material originated or stored in U-matic low band, high band or SP (superior performance) formats.

Decoder for time-base corrector with drop out compensation - technical profile

General Description

The decoder will lock to noisy and unstable signal sources including U-matic or VHS sources without TBC, Hi/Low/SP U-matic PAL and SECAM sources.

Improved Drop-out Correction via FM (RF) signal level analysis without full demodulation will process SECAM chroma drop outs (“Silver fishes”).

Input formats are Composite, Composite + Y Dub and Y/C (S-video).

Output format is 10 bit serial digital. A full frame synchroniser with horizontal and vertical phasing controls will lock the output to a studio reference.

The decoder will be contained in a 1U box with ergonomically designed front panel for rapid access to key functions. Remote control using Snell and Wilcox RollCall technology will be available.

Summary of Features.

- Multi-standard PAL/SECAM decoding of composite U-matic Hi/Low/SP video sources
- 10 bit data path throughout
- Lock to noisy and unstable signal sources
- Composite, Composite + Y Dub and Y/C (S-video) inputs

- Auxiliary RF input for advanced drop-out compensation control
- 2 x SDI outputs

Decoder- Synchroniser, Proposed Technical Profile
Feature........................Specification

Inputs
Composite.....................Connector format 2x BNC; PAL/SECAM
Return Loss better than –30db to 6MHz
......................Connector format 1 x BNC
Return Loss better than –30db to 6MHz
S-Video.........................Connector format 2 x BNC; PAL/SECAM
Return Loss better than –3db to 6 MHz
RF................................Connector Format 1 x BNC; U-matic RF
Return Loss better than –30db to 6MHz
Video Reference.............Connector format BNC; Composite Video (PAL)
Sync – Burst Level 0.3V + 3db

Outputs
SDI..............................Connector format 2 x BNC; 270Mbits
Return Loss better than –15db to 270MHz

Control Interface
RollCall..........................Connector format BNC; S&W Rollnet
Remote.........................Connector format 9 way D; S&W RollCall
RS485 or RS422 @ 38k Baud
GPI...............................Connector format 2 x BNC; Closing contact inputs

Direct Controls
Input Select Composite, Composite + Y(Dub), Y/C (S-video)
D.O.C. ....Off/On
Freeze ....Off/On
Genlock.........................Off/On
Pattern Black.................Off/On
Auto Chrominance Gain....Off/On (PAL only)

Proc Amp controls
...................................Video Gain + 6db
...................................Chroma Gain + 6db
...................................Black Level + 100mV

Enhance (optional)
...................................Horizontal Off, low, medium, high for mid and high frequency
...................................Vertical Off, low, medium, high

Set Up Controls
Pattern Select ....Black, 75%Bars,100% bars, Ramp, Multiburst
Genlock H and V Phase .... + 1 line in steps of 148ns; Full frame in 1H steps
Default Output ....Freeze, Black, Test Pattern
User Memories ....Store/Name x 8
GPI function...................GPI – Separate open and closed memory trigger
Preset Unit.....................Restores all factory settings

4.3 Multi-level encoding (VT4)

4.3.1 Place and requirements

- In the whole process, multilevel encoding represents the phase of input during which video formats are converted to multiple files formats.

- This phase takes place into 2 operations:
  - entry of the tapes: video quality checked tapes are delivered to the digitising plant from the central tape management service; associated data necessary for digitising operations are available and are imported through network from the central tape database;
  - output of the files: multilevel encoding is completed after encoding quality control, transfer of the files to the archiving library and return of the tapes to the central tape management service have been achieved

- Targeted requirements to be satisfied are the following:
  - encoding process should be - as much as possible - automated (including quality control) and in real time for all levels
  - valuable files are to be protected when they are delivered (master but also all intermediate quality files for which commercial usage can be forecasted)
  - automated re encoding must be possible in order to follow the evolving intermediate quality files

4.3.2 Hardware architecture

Introduction

The architecture to be adopted depends on the choices that will be finally made by the end user according to the way he will prioritise the previous requirements.

Therefore, the solution will be designed to provide different possibilities of implementation of those requirements (real time, protection, re encoding) and should lead to further comparative evaluation, which could be conducted by the reference process to be settled.

The staple architecture for an automated multilevel encoding process in which productivity (automation and real time encoding) has been prioritised is recalled hereafter.

A list of main hardware described in an operational view is presented and then comments on possibilities to handle previous requirements in terms of process organisation are made.

Functions

- List of main equipment in terms of commented functions is the following:

  VTR library: a robot able to handle all video broadcast video formats using cassettes (beta, ¾ inch and more recent physical formats like DV if necessary) is loaded manually: the main point to be solved, according to our knowledge of state of the art in this matter, is the physical adaptation of current robotics devices (cassettes bins and VTR rack) for the ¾" inch tape; if a previous transfer, for instance from ¾" to digital beta has been done, this point is of course non relevant

Switcher: as the signal must be distributed to all encoders a switcher is required with associated conversion function (A to D and D to A) and must be dimensioned according to the number of input flows: small robotics systems can currently handle 4 VTR and then 4 encoding processes can be treated in parallel, then at least 16 video input/output switcher is required

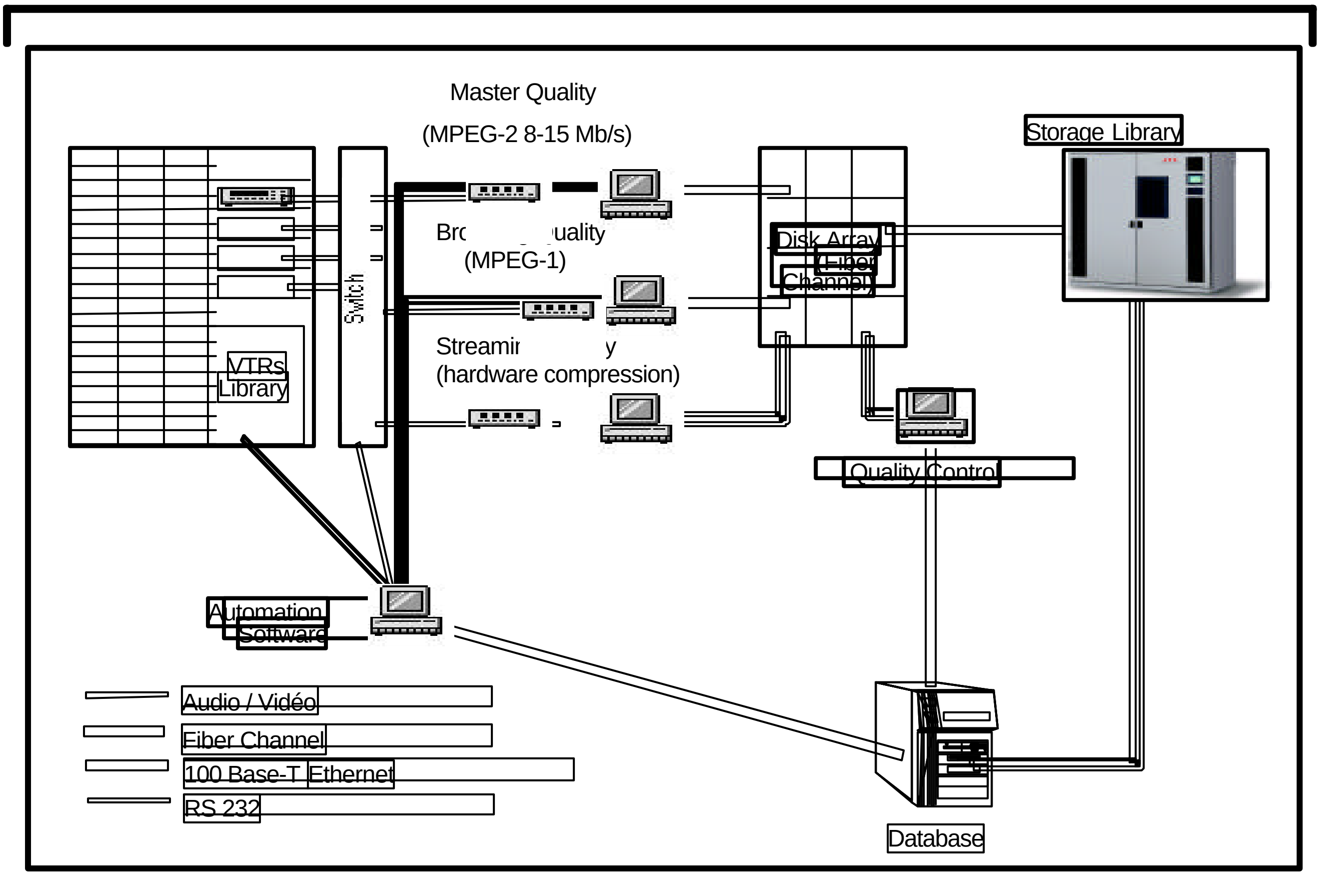

Figure 4.1: VT4-Hardware architecture

Disk Array: it is used for local storage after encoding in order to perform quality checking controls before transfer to the library; conversely it is used to load files from the library for all necessary use (transfer of the streaming files to a streaming server for internet usages, transfer of the browsing files to the browsing server for on line viewing or VHS/CD/DVD copy, transfer of the Mpeg 2 file for Beta tape copy or re encoding of sub level files at new internet standards); the disk array capacity and its cost directly depends from the volume to be transferred to and from the library but also on the overall process, the level of automation and the capacity to manage the process without intermediate stocks: for instance, if stocks of files under processing (due to non real time solutions encoding) or for quality control load permanently the disk array with high volumes, then the capacity (and the cost) of disk array must dimensioned consequently

Library: it is used for near on line storage, this type of storage being justified in the case of active programmes that is to say programmes for which the number of output for broadcast quality delivery is high; shelf storage of the Mpeg 2 file for non active programmes is another possibility that can save room in the library; types or supports and robotics device have been analysed by INA with advantages and constraints in very thorough reports to which the reader can refer

High data rate link between the encoders, disk array and library is required for fast transfer: the typical solution is to use fibre channel network

Quality control station is a PC equipped with a fiber channel board, all necessary video decoders (Mpeg 2, 1 and internet formats), a video board able to display multiple signals at the same time (4) and the quality control software described hereafter

Automation station: the hardware required is a current PC with Ethernet board addressing orders to a control box (black box equipped with TC boards) that will drive synchronously all equipment through RS 232 links: according to the size of the installation and the number of operators the number of automation stations should be adapted; it is noticeable that due to the organisation of the process and the software solutions all operations (application software for automation, data base access and global process supervision) can be achieved from a single station in a basic configuration and functions can be distributed on several stations if the volume of operation is high of the work organisation requires it

Database: a PC or workstation hosting all the information system from management of data at the tape entry level up to the archiving level

Quality devices described in the Quality Key Link must be added: the <HARDWARE SIGNALS CAPTURE AND PROCESSING SYSTEM> a real time, micro-controller subsystem dedicated to VTR electrical signals capture and processing has to be added at the output of the VTR; the <MPEG FILES CHECKING SYSTEM>, which is a computer sub-system to check synchronisation of formats, integrity of the files and conformity to standard, has to be added at the SAN level or at the library level

Requirements handling

From this architecture, it is necessary to appraise how to handle the initial listed requirements in the process.

- **Real Time**

Formats synchronisation

The request here is directly driven by the extracts activity of the archiving domain, which requires – in an inter frame digitisation system – GOP accuracy from the screening stage to the master delivery stage.

Synchronization of time codes is available for Mpeg 2 and Mpeg 1.

Synchronization with the streaming formats is not available due to the limitations of those formats in terms of time code management. This point could evolve in the future with the constant improvement of internet quality formats and the necessity to integrate broadcast technical requirements.

The solution then is to generate streaming formats from the Mpeg 1 file (or from the Mpeg 2 file as an option).

This drives to modify the staple architecture as described hereafter.

Streaming encoding is done after Mpeg 1 and 2 encoding and from the Mpeg 1 file (or from the Mpeg 2 file) stored in the Disk Array.

It is achieved with a current software solutions designed to manage video formats conversion (like Cleaner 5 from Terran).

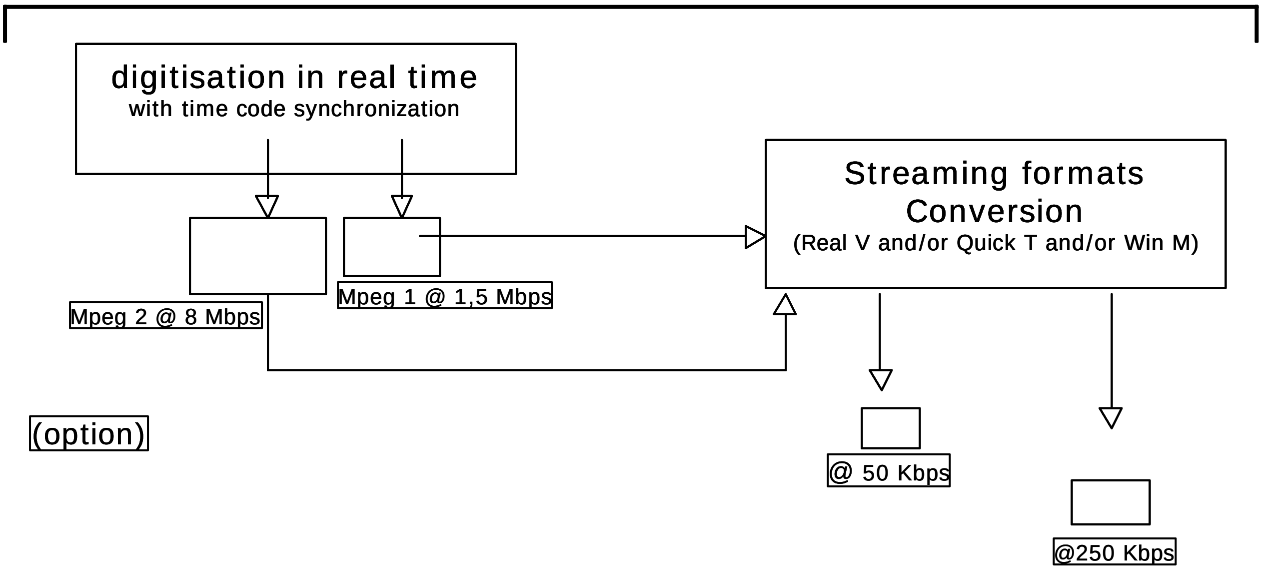

Figure 4.1: VT4-Synchronized formats

New constraints must be then taken into consideration with this type of process:

- Quality of the streaming files needs to be checked to appraise their acceptability: the Mpeg 1 source should degrade quality of those files compared with a Mpeg 2 source and a direct Pal source (in the staple process) but this could be acceptable for archiving practices;

- the streaming files are not generated in real time: a batch process using multiple PC and/or hardware accelerator board must be installed; non real time process must be managed by the automation software (and the operator) but the time required is smaller with an Mpeg 1 source rather than with a Mpeg 2 source

- due to the characteristics of frame rates (half of the Pal and NTSC rates) used by streaming formats, synchronisation must be precisely checked

If those new constraints appear to be unacceptable for archiving institutions, alternatives approaches are to be appraised, like time code insertion into the streaming files after the real time encoding process has been achieved.

Automated encoding quality control:

Requirements are derived from the Quality Key Link and are related into 2 main types of operations that can slow down the requirement of real time:

- VTR source quality control: although, as mentioned previously it is expected that the quality of the input tapes has been checked and adjusted and that solutions like previous transfer to digital beta has been done as much as possible, one must take into consideration the fact that encoding from an obsolete VTR format typically ¾" requires fre-

frequent adjustments of the VTR at the encoding process level: in order to meet the real time requirement thorough control of the VTR parameters should be handle by the automated digitisation software

- Encoding result: the consequence of the preservation of the time code synchro on real time has been mentioned previously; conformance to the Mpeg standard and integrity of the file should be an automated output of the <MPEG FILES CHECKING SYSTEM> as described in the Quality Key Link; the aided screening control should offer the possibility to the operator to check quickly the beginning and end of files (monitoring the whole encoded programme is dissuasive)with multiple windows of different programmes and/or different levels of digitisation;

- **Viewing copy protection:**

This protection is required for any valuable file which is to be made accessible to the end user.

Apart from usual protection system (like access control and scrambling) the protection is watermarking, although this possibility is not totally matured for the moment.

Watermarking is available to day for insertion of an identification information in a 4:2:2 stream.

The first possibility is to integrate this identification in the 4:2:2 flow directly at the output of the player VTR located in the robot.

The advantage is that all the files generated after should integrate this information although, for the moment, tests have not been conducted to check the efficiency and acceptance of watermarking into the encoded files.

2 limitations are to be considered in such a process:

- the key information which is the identification of the end user is not, by definition, taken into consideration (only the reference of the source, that is to say the original rights owner on the programme can be introduced at this stage)

- quality of the video files can be affected by watermarking.

This is why - if the target of the archive owner is to have a master reference file without any possible source of artefact - the preference should be to introduce watermarking with identification of the rights of the end user at the stage of Beta tape delivery as described hereafter.

The direct consequence is that none encoded file to be distributed on line is watermarked: this could be a limitation in the case of loading of quality files, like 250 Kbps streaming formats.

In a second stage, when watermarking in a Mpeg 2 flow (and then in an Mpeg 4 flow) is available and when delivery at those formats are current, watermarking can be introduced in the same way, that is to say at the delivery level (location on a streaming server and delivery of the Mpeg 2 file)

But these possibilities will not be available before the mid of the year 2003.

- **Automated re encoding:**

If the Mpeg 2 level at 8 Mbps appear to be the “reference format for operations” able to regenerate a broadcast quality flow and is perceived to day as a rather stable format, this is not the case for infra quality formats.

New standard like Mpeg 4 are to be finalized and widely used.

Moreover, constant improvement of encoders and decoders in the field of video quality as well as embedded information (for retrieval, digital rights management and security) will tend to regularly require redigitisation or modification (for instance by insertion of associated information) of the existing files.

This requirement bears consequences:

- the quality of the original format to be chosen for conversion: Mpeg 2 file to generate infra video rate formats is a guarantee of quality
- the replacement in the long run of the Mpeg 1 format at 1,5 Mbps by another one (Mpeg 4 most probably at a lower data rate)
- the consequence of this requirement on the productivity of the initial investment
- the anticipation of this requirement in the process: on the automation and information system software, in the sizing of network (here fiber channel), workstations and libraries.

Redigitisation is then comparable to the initial process after the Mpeg 2 encoding has been done.

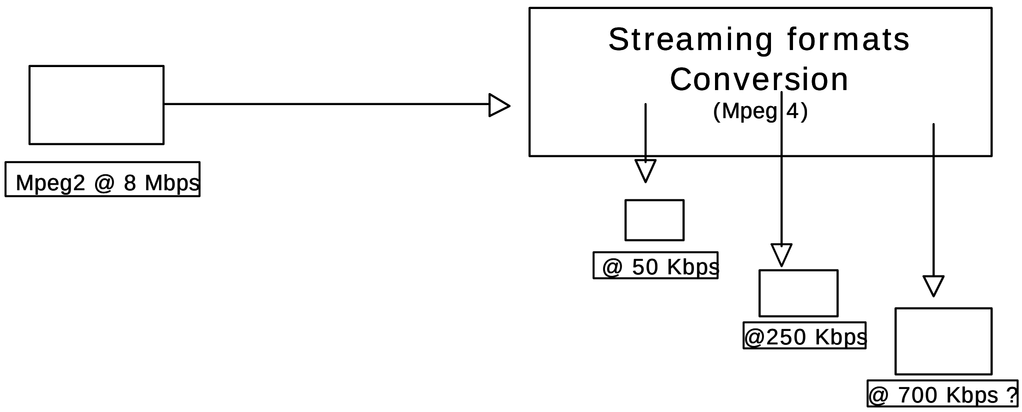

Figure 4.2: VT4-Automated re-encoding

4.3.3 Software architecture

The process software to be defined and implemented requires acquisition and/or development of the following elementary software and their interfacing into the process:

- media management software (1):
- multilevel digitising software (2)
- digitisation quality software (extension module of 2 or specific)
- files transfer software (extension module of 2)
- archiving software (3)

4.3.4 Media management software

Type and purpose

A data base able to import, gather and export all necessary information associated to the programme and which are generated all along the digitising process.

Functions

- Management of the following information at the entry stage:
  title at the programme level and sub programme levels if necessary (serial, topic…) according to the rules defined in the archiving institution
  duration
  TC in and out
  input quality control status: tapes parameters checked with date and origin
  reference number associated with bar coded tape
  physical location of the tape: entry, robot, exit…
  comments
  other data: to be defined according to the spec of the metadata key links

- Management of the following information at the digitising stage:
  Digitising characteristics to be used must be integrated into the media manager and added to those of the previous stage:
  characteristics of encoders used
  parameters of encoding used: codec, bit rate, frame rate, process parameters (video filters, audio/video levels) and settings (assignment of parameters to a group of video programmes to be encoded);
  comments
  other data: to be defined according to the spec of the metadata key links

- Management of the following information at the quality control stage:
  Video quality control characteristics to be used must be integrated into the media manager and added to those of the previous stage:
  Mpeg 2: mention of the Mpeg 2 analysis : done or not by physical display with date and operator identification; conformity to the Mpeg 2 standard (import of the Output ASCII stream to be defined that will report status on MPEG standard conformance, file integrity, and synchronism checking (see Quality control Key Link 4.1.10.2)
  Other formats: TBD;

- Management of the following information at the transfer stage:
  Transfer characteristics to be used must be integrated into the media manager and added to those of the previous stage:
  archiving ID (common to all files of the same programme but with a suffix indicating the format, the data rate and the date of digitisation)
  size of each unitary file
  physical destination: tape, library/shelf; copy to the streaming server or the browsing server
  comments
  other data: to be defined according to the spec of the metadata key links

- With other software:
  multilevel automated digitisation software, transfer software and quality control software: export and import of described data at each stage of the process

central tape management (or customer) facility: in order to follow the process (if required) and to enrich the central tape management data base with information about file copies

4.3.5 Multilevel digitising software

Type and purpose

A software specialised in industry process, able to control and synchronise, in real video time (at least 1/50 sec), through local network, all necessary equipment of the digitising plant.

Easiness of customisation, evolution and maintenance will be appreciated.

Functions

- **Editing**
  A list of orders to be executed and named multilevel digitising list is edited with information available in the media manager at this stage of the process.
  This list can be executed through manual take over, at fixed hour or as soon as possible (after analysis of the availability of resources) by session or continuously.
  An as run log of this list is edited after completion.

- **Control**
  Equipment are controlled and synchronised in real time by the multilevel encoding software (as described in the staple schematics, through Ethernet 100 base T and RS 232).
  Topology of equipment used is available so that in case of automated chains working in parallel and sharing common capacities, available equipment or capacities can be reallocated through drag and drop of equipment.
  - Equipment to be controlled is: input robotics (Beta, D1/D2, ¾ inch; other formats to be discussed) with adequate VTR, switcher, converters, encoders (Mpeg 2, Mpeg1, QT, RV, WM), SAN for buffer storage before quality control.

  - Special control for ¾"VVTR fine tuning must be provided according to Quality Control Key Link: the automation software should be able to exchange information with the <AUDIO/VIDEO ANALYSIS SYSTEM> either directly or through the <OVERALL CONTROL SYSTEM>. and control the <HARDWARE SIGNALS CAPTURE AND PROCESSING SYSTEM> for modifying as quickly as possible VTR settings in order to guarantee the best quality for tape reproduction either at the operator's initiative or at the request of the chief operator in charge of the <OVERALL CONTROL SYSTEM>

  Alarms generation and displays, remote control of the operations (in case of 24 hours digitising) and tele-maintenance must be available in real time.

  Detailed functionality shall be discussed during the phase 2 among which: Manual take over at any time on any equipment, multi encoding list editing and execution etc. according to archiving institutions end users.

Interfaces

- **With other software:**
  Media management software (1): described above
  Digitisation quality software (extension module of present software or specific software): interactions between multilevel digitising software and quality control software

software depends on definition of parameters with tolerance levels, manual /aided or automated quality control (to be provided and completed later on); an interactive process must be defined afterwards

4.3.6 Digitisation quality software

Type and purpose

A software module able to manage quality control requests (video and process) which is either a sub-module of the automation software or an independent module.

The quality control phase described above at the output of the VTR reading the tape (<AUDIO/VIDEO ANALYSIS SYSTEM> and <HARDWARE SIGNALS CAPTURE AND PROCESSING SYSTEM>) is not taken into consideration at this stage, which is dedicated to encoding result quality.

Functions

- **Video quality control**

  The <MPEG FILES CHECKING SYSTEM> described in the Quality Key Link is to analyse each unitary file through the automated Output ASCII stream (to be defined) that will report status on MPEG standard conformance, file integrity, and synchronism checking

  The overall control quality system described as well in the Quality Key Link will define a set of parameters to be monitored during transfer with value, and alarm conditions etc so that if the MPEG file checking is not successful an alarm and flag to the final report is produced and sent to operator or chief operator in charge of the global supervision of the process

  Screening quality control described above should be done on the Mpeg 1 and 2 files as well as internet formats especially the 250 Kbps ¼ screen size which is a valuable file (beginning and end of files with results indicated in the data base including date and name of the operator);

- **Process quality control**
  - **equipment control:**
    - gathering and displaying of alarms of each equipment: VTR, entry robot, encoder, SAN, Libraries Robot… any hardware equipment driven by encoding automation software: log of error origin
    - gathering and displaying of overall resources: networks, data servers
  - **workflow performance**
    - displaying of occupied/available capacities of each equipment
    - non interruption degree (for instance in case of rejected file(s))
    - statistics for reporting per reference period: input and output volumes with characteristics, rejected files, process failures (interruptions with cause and frequency)
  - **other controls**

Interfaces

- **With other software:**
  - Multilevel digitising software and file transfer module
  - Media management software in order to archive quality encoding control sheet as metadata associated to the programme

4.3.7 Files transfer software

Type and purpose

A module of the automated digitising software having the same staple characteristics (although at this stage there is no constraint due to the characteristics of the video signal) and able to control and synchronise, through local network, all necessary equipment of the archiving phase.

Functions

- **Editing**

A list of orders to be executed and named transfer list is edited with information available in the media manager at this stage of the process.
Reorganisation and adjunction of data requires efficient cut and paste function.
The transfer list can be executed through manual take over, at fixed hour or as soon as possible (after analysis of the availability of resources) by session or continuously.
An as run log of this list is edited after completion.
In case of redigitising a transfer from file to file (for instance for conversion from Mpeg 2@… to) is to be managed by the transfer software with a phase of quality control.

- **Control**

Main characteristics are comparable to the digitising software.
Equipment are controlled and synchronised by the transfer software (as described in the staple schematics, through Ethernet 100 base T and RS 232).
Topology of the equipment used is available so that in case of automated chains working in parallel and sharing common capacities, available equipment or capacities can be reallocated through drag and drop of equipment.
Main equipment to be controlled is: the tape drive (DLT, LTO, other) and its library; multiple drives and libraries can be driven at the same time.
Alarms generation and displays, remote control of the operations (in case of 24 hours transfer) and tele-maintenance must be available in real time.
Detailed functionality shall be discussed during the phase 2 among which: Manual take over at any time on any equipment, multi transfer list editing and execution etc. according to archiving institutions end users.

Interfaces

- **With other software:**
  Media management software (1): described above
  Digitisation Quality software: operator should check that quality requirements are filled before any transfer operation starts

4.3.8 Archiving software

Type and purpose

Archiving software is a standard software module designed to archive the video files.

Functions

File management of the archiving software must be able to handle the size of video files which is not always the case of current archiving software: 1 set of files of 1 hours of video programmes at 4 levels of digitations is: 4.4 GB

Usual files management for archiving purposes must be performed, mainly:

- file ID of the video programme with sub ID for each video file tape ID: technical data regarding encoding characteristics (origin, parameters) should be embedded into each video file of the programme so that checking can be undertaken easily during future usages
- location on the tape
- location of the tape
- library ID
- location of the library

Other functions are to be evaluated according to archiving institutions needs, among which:

- cancelling and replacing obsolete files, due to re-encoding process or any other reason; in case of massive re-encoding a specific procedure should be defined

- usual archiving operations: restore, defragmentation etc.

Interfaces

- **With other software:**
  Media management: this software keeps all information generated for and by the digitising, transfer, re-digitising processes and quality controls; the archiving software is a subsystem of this software

- **With library:**
  Most manufacturers propose archiving software adapted to the characteristics of their library; this point must be taken into consideration before any choice.

4.4 Lossless compression for video (VT5)

Proposal: Investigate the feasibility of the use of lossless compression in archive master material -- for the preservation of content originally held on videotape.

Purpose:

1) reduce the overall cost of storage of video on data tape
2) allow full quality storage of video on standard videotape formats
3) reduce the bandwidth requirement for movement of full-quality video

Background: There is no single data rate for the digital representation of an analogue video signal, because it depends upon the sampling rate and sampling resolution (word length). However there is a standard, IEC Recommendation 601, which works out to 270 M b/s for European 625 line colour video, at standard definition (not widescreen or high definition). With the elimination of the off-screen portions of the signal (line and field flyback), the required data rate reduces to about 200 M b/s.

If this signal could be compressed in a lossless fashion by 2:1, the resultant signal could be stored on 'top-end' professional standard videotape formats, such as 'digibeta', a Sony format widely used in broadcasting. If the video could be compressed in a lossless fashion by 4:1, the signal could be stored on the DV-PRO 50 format, which is a very small and economical format, originally developed for the high-end consumer market.

The main requirements for a lossless video compression tool are similar to those for audio:

- The decompressed signal must be bit-for-bit equal to the original
- It must be possible to implement the compression method in real time
- The compression algorithm must be fully specified and possibly standardized by a proper International Body
- The decompression tool should preferably be implemented in software and run faster than real time on a conventional PC
- The decompression tool should be platform independent
- Partial file access should be possible
- A proper encapsulation must be defined; this is expected to be a part of the XMF format, currently being developed by EC project G-FORS.

Work to be performed:

Technology Assessment: Understanding what lossless compression offers.

- Assessment of state-of-the-art in lossless compression
- Implementation of state-of-the-art software models for encode / decode
- Use of this software on standard test signals

Workflow assessment: An important consideration is the **practical** role of a new kind of compression in existing broadcast infrastructure. The study will need to look at the standard production and distribution methods in use, and being adopted. Key issues are:

- Ability of conventional distribution circuits to handle the lossless signal
- Ability of conventional digibeta or DV-PRO 50 videotape recorders to be modified to handle the lossless signal
- Relationship of the proposed lossless signal to the production and distribution requirement for DVB (digital video broadcasting; originally developed as a detailed specification by EC Project Eureka, headed by BBC R&D)

Outcome:

1) A statement of state-of-the-art in lossless video compression, including:

- Degree of compression obtainable
- Process power required for encode and decode
- Implementation requirements

2) A workflow analysis of the potential contribution of lossless video to broadcasting

- Analysis of production workflow based on videotape recorders and on servers, and how lossless compression could be used
- Analysis of transmission workflow based on MPEG-2 (DVB), and how lossless compression could be used
- Interoperability: Relationship of lossless video to other forms of video representation
- Transfer: Options for use of streamed and file transfers for lossless video

3) A set of software routines for lossless encode and decode

4) An example tape, showing what compression rates can be achieved on typical and on demanding material; such material has already been prepared by the BBC and others for tests of lossy compression, in particular MPEG-2

4.4.1 Hardware architecture

Encoding and decoding can in principle be performed either in hardware or software. For the purposes of a feasibility study, it well be assumed that hardware to implement encoding in real time could be produced if warranted, and a price estimate for this hardware will be made if needed. There will be no actual construction or purchase of hardware.

4.4.2 Software architecture

Encoder and decoder tools are software modules that can be launched by other programs. Depending upon their performance, they could run either in real time or in batch mode. The tools will be made available both for Windows and UNIX in the form of command line executables, for ease of integration in the control environment.

There are two main possibilities for implementation of the encoding process.

- the encoding is performed in real time while digitising

- the encoding is subsequently applied to the digitised files

In the first case the encoding software can probably be hosted by the acquisition hardware; in the second case there is a need for a set of encoding servers for batch processing.

Decoding: For audio file transfers, an evaluation is to be made of decoding at source vs. decoding upon receipt. **For video, the use of lossless compression will be of little interest**

interest unless the decoding can be done upon receipt, to allow reduced-bandwidth for the transfer. Therefore the standard architecture will require:

- the decoder to be hosted on the archive user workstation

While hardware decoding may be needed for real-time performance, for file transfers there is no real time constraint and therefore the decoding is envisaged as being purely in software.

Chapter 5 Key Link Technology in Audio Preservation

5.1 Audio playback devices improvement (AT1)

Playback devices used in the massive transfer chains should be optimised for maximum speed of operation of the overall system. This activity consists of

- selecting on the market the equipment most suitable to massive transfer operations,
- identifying areas where the ergonomics of the equipment can be improved
- defining appropriate modifications or tools where needed.

The specific requirements for this link are:

- Apply automation, where technically possible, to the playback devices.
- Reduce the need for manual trimming and conditioning of the playback devices to follow the media peculiarities, like azimuth alignment, selection of recording format (mono, stereo, quad) for tape players and pickup type, rotating speed, equalization for turntables.
- Automatically notify when maintenance or consumables replacement is required.
- Select state of the art equipment, considering also future needs for maintenance, spare parts supply and ease-of-manufacture of the solutions

5.1.1 Hardware architecture

The hardware involved is mainly constituted by the analogue part of the transcription chain, that is tape players, turntables, preamplifiers and A/D converters.

5.1.2 Software architecture

No specific software architecture is expected for this key link even if there could be the need to include some software procedures under control of the transcription manager to support the hardware automation.

5.1.3 Supporting tools

Type

Software procedures could be needed to support the hardware automation. No particular criticality is expected from these procedures.

Purpose

The purpose of these procedures is that of issuing commands to the hardware, by driving serial or parallel ports, and of monitoring the hardware activity by analyzing the reports of the quality control module.

5.1.4 Double arm turntable

Type

The activity required by this hardware item consists of defining proper mechanical modifications to commercial turntables to improve the efficiency of 78 RPM records transcription chains.

Purpose

78 RPM records were produced in absence of standard. Several different parameters could have been employed that require a proper matching in the reproduction equipment. The main implications are on the geometry of the pick up, on the reproduction speed and on the equalization applied. While errors on speed and equalization can be effectively compensated with a suitable digital post-processing, and therefore, be moved off-line with respect to the main transcription chain, reading the record with an inappropriate pickup results in the loss of signal details that cannot be recovered at a later stage.

Table 12: AT1-Most widely used styli vs. time

period	Conical truncated stylus	Elliptical truncated stylus
Pre 1920	.0040’’	.0040’’ x. 0012’’
1920-1939	.0035’’	.0035’’ x. 0012’’
1939-1966	.0028’’	.0028’’ x. 0009’

The table above enumerates the most widely used styli and an approximate period for which they are appropriate. It can be noted that there are two variant for each stylus width, a conical profile and an elliptical one. Generally the latter is to be preferred as it is known to reduce the distortion but in the presence of records in poor condition, conical truncated styli can produce better results as they are less sensitive to surface damages.

Provided that the items to be digitised will be grouped into transcription sets taking into account the production period, still there are at least two possible styli to choose between. To avoid the need for switching stylus or cartridge between the transcription of an item and the next, it is convenient to fit two tone arms on a single turntable, so that two styli configurations can be tried on the fly without any reconditioning of the chain.

Function

By installing two tone arms on a single turntable, we have available up to 4 versions of the reproduced item, in the hypothesis that stereo cartridges be used. The rationale for using stereo cartridges resides in the fact that the two sides of the record groove could "sound" slightly differently due to possible damages of the surface or uneven distribution of dirt particles. Furthermore, the alignment of the stylus might not be perfect due to several reasons, e.g. the arm is not perfectly calibrated or the record is bowed. Then, if the operator feels uncertain about the best version, it is possible to digitize all the 4 versions at the same time, letting therefore the chain work at its maximum speed, and to postpone the decision about the version that will be eventually stored permanently at a later and more convenient stage.

Interfaces

The use of 2 tone arms requires that the transcription manager software be aware that the various versions are alternatives for the same item, so that they will be properly handled and classified.

Processing

Transcription sets: it is convenient that the sets be composed of homogeneous items with respect to Table 12 , so that the turntables can be equipped with the two styli that are most likely appropriate, probably a conical and its equivalent elliptical.

Transcription process: a single pass transcription can be performed by recording the arms outputs with a 4 channel system. The selection of the best transcription can be done on the digital signal, either automatically (driven by the quality check key link), or manually by listening. If performed manually, this operation can be demanded to a skilled person, not necessarily coincident with the transcription operator.

5.1.5 <A/D conversion technology selection

Type

All the modern implementations of audio Analogue to Digital converters use devices based on the Sigma-Delta modulation technique that solves several problems generally found on traditional parallel converters. Sigma-Delta converters are in fact intrinsically linear over all the usable dynamic range and several devices on the market show that with this technology true 20 bit resolution can be obtained, giving a Total Harmonic Distortion + Noise (THD+N) figure of -120 dB. Recently, a new approach to A/D conversion has been proposed, based on the use of a parallel of up to 4 Sigma-Delta converters, appropriately scaled to cover distinct dynamic ranges. The output of the single converters is then combined to obtain the digital representation of the analogue signal. The philosophy of this approach is that of extending the dynamic range of the converter up to 150 dB, while maintaining the THD+N constant at about -100 dB in a range of 50 dB of the input signal. Having diverging measurement results, namely higher dynamic range and lower THD+N, it is not trivial to instrumentally determine if this approach brings any benefit to the quality of the conversion compared to that employing a single Sigma-Delta converter, but further investigations, including listening tests are needed to understand how the perceptibility of the quantization noise/distortion produced by this equipment compares to that of state of the art single converter equipment.

Purpose

The purpose of the A/D conversion is that of digitising the signal contained on the analogue archived media to store it in the new digital formats. The quality of this operation is crucial to the achievement of the goal of an optimal preservation of the archive content. The best approach to A/D conversion, taking into account the peculiarity of the material to be digitised, must therefore be found via extensive testing of state of the art commercial equipment.

Interfaces

Analogue to Digital converters are connected to the transcription chain via standard audio interfaces. The technology used to implement the conversion device has no impact on other links of the chain.

5.1.6 Turntable automation

The activity required by this item consists of defining proper mechanical modifications to commercial turntables to improve the efficiency of vinyl transcription chains, by automating start and stop operations.

The choice of commercial turntables is drastically reduced to a scant number of semi-professional products, if start/stop automation is required. The expected advantage of turntable automation consists of the possibility of automatically starting the transcription with a single key press on the transcription control computer without having to manually position the pickup on the record. The end of the transcription can be automatically detected by a sensor, relieving the operator from the need of continuously monitor the position of the pickup on the record, operation that would make rather difficult the parallel management of several transcription lines by a single operator.

The functionality required is as follows: before starting the transcription, the arm must be manually positioned over the record in its lifted position, then, when the operator starts the acquisition on the control computer, the turntable motor is started, the arm is automatically lowered on the record and the digitisation starts as soon as a signal is detected from the pickup. When the pickup reaches the end of the record, a sensor captures the event and notifies it to the control computer, that stops the acquisition, lifts the arm and stops the turntable motor. The arm must then be manually put in its rest position to remove the record.

The automation of turntables must be driven by the transcription control software.

5.2 Audio quality control (AT2)

5.2.1 Hardware architecture

PC Processor PIII-1GHz, 256Mb RAM, HD 10Gb UltraScsi, Sound Blaster, Linux RedHat6.2.

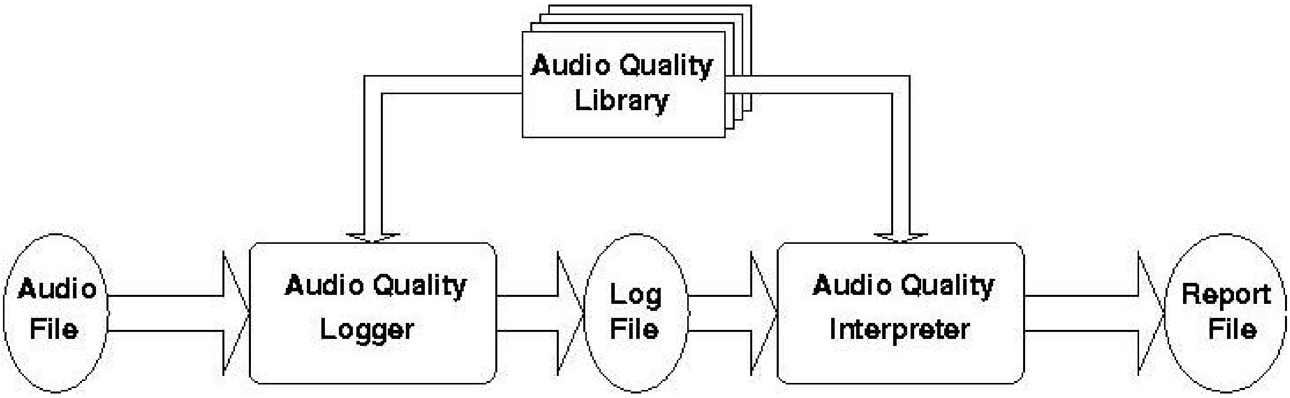

Figure 5.1: AT2-Hardware architecture of the Audio Quality Control

5.2.2 Software architecture

5.2.3 AudioQualityLibrary

The module is a library of subprograms for Linux/Unix OS including header files and documentation.It will contain all functions implementing algorithms for measuring parameters and detecting artifacts needed to be logged and successively used for determining the overall quality of the input audio file.

For each of the following parameters there is a specific function computing it (for example see reference [1]):

- energy
- bandwidth
- saturation
- peak signal level
- silence duration
- noise duration
- phase correlation

Moreover, the library will contain functions to detect particular artifacts (see for example [2] and [3]):

- drop-outs

- clicks and scratches

Finally, functions for computing meta-data related to acoustic contents will be into the library too:

- segment boundary detector ([4,5,7])
- segment classifier ([6,7])

This module depends on the operating system for which the library will be compiled.

It is required by the AudioQualityLogger and by the AudioQualityInterpreter.

The module is a library and then it will be used by the items requiring it through appropriate linking.

The software item is a library of object codes, to be linked to a main source. The sources of subprograms, written in C/C++, will be compiled for Unix/Linux OS. Then, in order to use the library, a standard Unix/Linux workstation/PC with a C/C++ compiler and a main program are needed. For obtaining responses in less than real-time, hardware equivalent to low-cost modern PC is enough, e.g. a PC equipped with a PIII-1Ghz, 256Mb of RAM and 10Gb of hard disk. Strictly speaking, an audio device is not necessary since the acquisition of samples is outside the scope of the module, but it is recommendable for checking purposes, as it allows the playing of audio signal.

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Processing

- energy: this will be computed directly from the samples of the digital representation of the signal;

- bandwidth: for each analysis window, the power spectrum can be estimated by using the FFT; the bandwidth can be given as the frequency interval where a given percentage of the overall energy falls;

- saturation: if it is due to the original media (e.g. Fully magnetization of the tape), it can be detected by selecting a proper threshold on the energy level. If it is due to the digitising process, it can be detected depending on what the converter put into the audio file. Often, audio

audio samples of a saturated interval assume the highest value of the numeric representation employed; in this case, a saturation condition can be detected counting the number of such samples in an analysis window with respect to the total number of samples;

- peak signal level: it can be given as the highest energy value among all the analysis windows of the overall signal;

- silence duration: it can be assumed that silence occurs where the signal energy is below a certain threshold. Given that, the total amount of silence in the audio file is easy to compute. The problem is to establish the threshold. It could be set by hand, but an interesting issues is how to automatically set it to a proper value. This will be investigated during experiments;

- noise duration: comments are similar to those on silence duration, given a definition of the noise level; this issue will be experimentally investigated too;

- phase correlation: especially in vinyl disk digitisation, a wrong azimuth of the cartridge can be detected by measuring the phase correlation between the right and left channels;

- drop-outs: should be detected on the basis of the signal energy and power spectrum curves;

- clicks and scratches: algorithms have been recently presented in specialized literature ([2,3]);

- segment boundary detector: the Bayesian Information Criterion (BIC) was defined in [4] and used for the first time for audio segmentation in [5]; its implementation will be included in this library;

- segment classifier: a classifier based on Gaussian Mixture Models (GMMs) [6] will be added in this library, including also acoustic models trained on sample data.

5.2.4 Audio Quality Logger

The source and the executable form of a main program and documentation. The system represents a stand-alone program for the production of a log file containing information regarding the quality of the input audio file.

It performs the computation of the parameters and the detection of artefacts of an input audio file. Moreover, it segments the audio contents into acoustically homogeneous chunks, and classifies them into broad acoustic classes.

It requires the linking of the Audio Quality Library and its output will be processed by the Audio Quality Interpreter.

Input: an audio file containing PCM samples (24bit, 48KHz, stereo). The header (if any) has to be defined.

Moreover, it needs some other input parameters; at least:

- a threshold for the silence/no-silence discrimination
- a threshold for the noise/no-silence discrimination
- a threshold for the saturation/no-saturation discrimination
- a threshold for the BIC algorithm
- a set of trained GMMs, representing acoustic classes

Output: a text file containing the log of all the features computed on the input audio file by using the functions of the library AudioQualityLibrary.

The software item is an executable main program, compiled for Unix/Linux OS. The source code will be released as well, written in C/C++. A standard Unix/Linux workstation/PC is needed to run the executable; a C/C++ compiler is also required if there is the need to compile the source program. For obtaining responses in less than real-time, hardware equivalent to low-cost modern PC is enough, e.g. A PC equipped with a PIII-1Ghz, 256Mb of RAM and 1Gb of hard disk.

For references and details of processing, please see Audio Quality Library.

5.2.5 Audio Quality Interpreter

The source and the executable form of a main program and documentation. It represents a stand-alone program for the assessment of the overall quality of an audio file. It analyses the log file output by the Audio Quality Logger for individuating intervals of the audio file needing to be restored and for giving a global audio quality score. It requires the linking of the *Audio Quality Library*.

Input: a log file containing information regarding the acoustic contents of an audio file; the log file is produced by the AudioQualityLogger.

Output: in some format to be defined, the program will output timestamp information about interval needing restoration and the global audio quality score.

The software item is an executable main program, compiled for Unix/Linux OS. The source code will be released as well, written in C/C++. A standard Unix/Linux workstation/PC is needed to run the executable; a C/C++ compiler is also required if there is the need to compile the source program. The computational load is low and then no particular hardware is required (see *Audio Quality Logger*).

For references and details of processing, please see Audio Quality Library.

5.3 Lossless compression for audio (AT3)

5.3.1 Introduction

To avoid that important signal details be removed during the transcription process, the audio signal is digitized at 48 kHz/ 24 bit stereo, resulting in about 1 Gbyte per hour. Therefore, an average European radio archive will require several hundred Tbyte of storage. It is clear then that storage represents a relevant part of the overall archive cost and that reducing the size of the archived materials consistent savings can be expected. Unfortunately, in order to preserve the highest possible signal quality, no lossy bit rate reduction processing, like MPEG coding, can be applied. It is known that perceptual coding techniques may introduce the following kind of problems:

- coded signals are prone to multigeneration coding noise amplification, that is, cascading several coding-decoding passes (when reusing the material) causes an uncontrolled increase in the coding noise that could eventually make the signal unusable
- post-processing can change the time-frequency distribution of the signal components and reveal coding artifacts not otherwise audible (the so called unmasking effect)

In the computer environment utilities like **ZIP**, **gzip**, or **compress** are effectively used to reduce the size of generic data files without impairing in any way the actual information content, that is the cascading of the compression and decompression processes produce a file that is bit to bit identical to the source. These programs are generally variants of the well known Lempel-Ziv algorithm [7], that exploits the statistical correlation of a generic binary source. The obtainable compression ratio is very dependent on the source itself and this approach cannot be applied to fixed channel applications, like broadcasting. In the case of archiving, this is not a problem as the purpose here is that of reducing the storage space, that can be freely allocated by the system where needed. Unfortunately, the audio signal presents a statistics that is not very suitable to the application of these algorithms, and a rather low compression ratio is to be expected.

5.3.2 Technical background

A few methods have been proposed in the literature specifically studied to compress the audio signal without loss of information. The basic principle is that of removing the signal correlation: removing parts of the signal that can be reconstructed by the decoder without being explicitly stored.

Two basic approaches are described in the literature:

- time correlation exploitation [1]
- frequency correlation exploitation [2,3]

The first approach reduces the entropy of the signal by removing the periodic components of the signal in the time domain, while the second performs the same task but operating on the signal after transformation in the frequency domain, by means of a mathematical transformation like the Fast Fourier Transform (FFT) or the Discrete Cosine Transform (DCT). The ap-

The approach based on the transformation in the frequency domain is more complex as the process is truly reversible only if infinite precision is used to represent the transformed coefficients. This is of course not realistic and therefore practical implementations use fixed length coefficient precision at the expense of needing to take into account a residual signal error that must be coded to allow for perfect reconstruction after decoding.

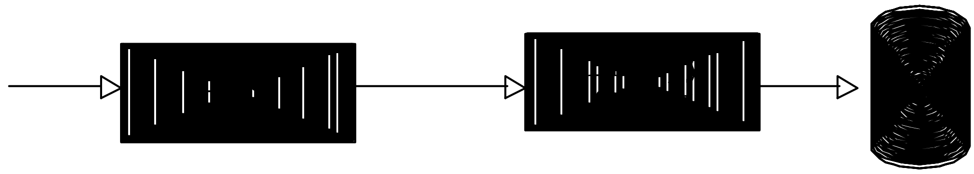

Figure 5.1: AT3-Basic scheme of temporal prediction based lossless coders

Figure 5.1shows the building blocks of the systems based on the temporal correlation exploitation. The first module, called LPC (Linear Prediction Coding), implements one or more selectable prediction strategies, where the value assumed by a sample at time *t* is predicted by the observation of the values assumed by the *n* previous samples. In the literature several LPC based algorithms have been proposed and applied to various applications in the field of audio and speech coding. The output of the LPC module is then the difference between the actual sample value and its computed prediction, often called residual signal. It is expected that the dynamic range of the residual be generally lower than that of the original signal, therefore, less bits are required to code the sample values. Looking at the distribution of the values assumed by the residual samples over time, a Laplacian curve centred on the zero value can be observed. This kind of distribution is most efficiently coded using Variable Length Codes (VLC), of which Rice codes and Huffman codes [4,5,6] are the most known and used. This is the purpose of the second module, called Entropy Coding. To summarize, the LPC module reduces the correlation of the signal by removing the "predictable" parts, while the Entropy Coding module exploits the uneven distribution of values to efficiently store the residual signal. In practical applications several variants are possible, including continuous adaptation of the coding parameters (like prediction coefficients or codebooks) to cope with the long term non stationariety of the audio signal.

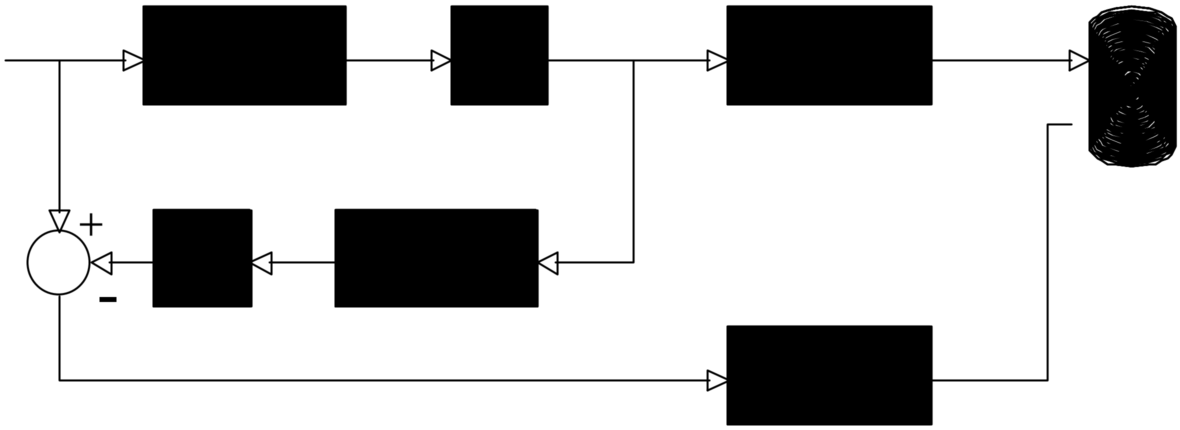

Figure 5.2: AT3-Basic scheme of transform based lossless coders

The principle scheme of transform based lossless coding is shown in Figure 5.2. The time signal is transformed to a frequency related domain via a mathematical transformation, generally the DCT. This transformation is known to reduce the correlation of the audio signal and is effectively used in several lossy coding audio schemes, including MPEG Layer 3 (also known as MP3) and MPEG4 AAC. Unfortunately, the DCT transformed signal could be losslessly reversed to the time domain via the Inverse DCT (IDCT) module only by using infinite precision arithmetic. This cannot obviously be realized in practice and therefore we cannot avoid the Quantizer (Q) module. The error introduced by this truncation operation must then be coded in a stream be coded in a stream parallel to that containing the transform coefficients. Entropy coding is used also here to efficiently store both the truncated transform coefficients and the error signal.

Regardless of the higher complexity, no major advantage has been observed up to now using the transform approach, most of the systems that can be found on the market are based on the time correlation exploitation.

5.3.3 Scope of the activity

The compression ratio obtainable from any lossless system is highly dependent from the signal, or better from its entropy: white noise cannot be compressed at all, as each sample is completely uncorrelated from any other, while a DC level requires virtually no bit, as all the samples are equal. On the internet, a few independent tests can be found that evaluate the performance of some of the available lossless coding tools [8,9]. As the compression ratio varies, we can only take into account an average value of the ratios obtained applying compression to different kinds of materials. In our case, as the amount of material to code is very high and varied, this figure can be considered representative as long as the tests included a balanced mix of items. The comparison of the results of the evaluation of the various tools shows that there is no dramatic difference of performance between the different tools, while the performance varies strongly with the coded signal, as expected. Of course we cannot say that all the systems are equivalent, as they show up different features, like real time performance, cpu usage, input and output formats handled, detailed documentation of the encoding-decoding algorithm and, of course, coding efficiency.

The best tools rate at an average compression ratio close to 2, that is, the storage required could be roughly halved, but all the readily available results are related to 16 bit signals. Thus, further tests must be organized and run using 24 bit materials coming form the digitisation of legacy materials to verify if this figure can be considered valid for archiving applications.

The main tasks of this activity are, therefore, the selection of a suitable coding strategy between those proposed in the literature, its adaptation to 24 bit signals, and the performance of tests to verify if the compression ratio achieved is such to justify the inclusion of such a tool in the reference transcription chain. In case of positive results, then, an operative scenario must be devised and experimented.

The main requirements for a lossless compression tool are:

- The decompressed signal must be bit-to-bit equal to the original
- It must be possible to implement the compression method in real time
- The compression algorithm must be fully specified and possibly standardized by a proper International Body (this activity extends beyond the PRESTO scope, and can only be promoted here)
- The decompression tool should preferably be implemented in software and run faster than real time on a conventional PC
- The decompression tool should be platform independent
- In no case the compressed signal should result in a file larger than the original
- Partial file access should be possible
- A proper encapsulation in the BWF container must be defined

5.3.4 References

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[9] **Monkey's Audio – Comparison**
http://www.monkeysaudio.com/comparison.html

5.3.5 Hardware architecture

Depending on the time performance of the encoding and decoding tools the hardware platform required to sustain the digitisation process will be estimated. Two possible architectures for decoding are envisaged:

- the decoder is hosted on the archive user workstation
- the file is decoded before being sent to the user

In the first case no dedicated hardware has to be provided, while in the latter a set of decoding servers must be provided in the archiving system.

A similar situation is envisaged at the encoder side:

- the encoding is performed in real time while digitising
- the encoding is subsequently applied to the digitised files

In the first case the encoding software can probably be hosted by the acquisition hardware, in the second case there is a need for a set of encoding servers for batch processing.

In any case, no custom hardware implementation of either the encoder or the decoder is foreseen, as it is expected that software based implementations, running on standard PCs, can be cheaper and more easily integrated in the transcription and archive systems.

5.3.6 Software architecture

Encoder and decoder tools are software modules that can be launched by other programs. Depending on the performance, they could be run either in real time or in batch mode. The tools will be made available both for Windows and UNIX in the form of command line executables, for ease of integration in the control environment. A few open implementations are already available on the internet. They will be evaluated to verify if any of them can be used as a base for further enhancement and adapted to our envisaged application. In particular, extension to 24 bit resolution (if not yet available), management of BWF files and coding enhancements are expected activities. In case that no appropriate freeware software be available, a custom implementation will be done.

5.3.7 Encoding_tool

The encoding tool is a command line program that performs the compression of an audio signal. The linear audio to be encoded must be contained in a BWF file. The program produces at the output a BWF-like file containing the same metadata as the input file and the compressed audio.

The item solves the function of reducing the storage space, as well as the transmission bandwidth, of the digitised audio materials without impairing the objective sound quality.

This tools performs lossless compression of the audio signal by exploiting the time correlation of adjacent samples. The result of the operation will be a sequence of packets of bits corresponding to groups of adjacent audio samples. The original audio samples can be reconstructed by applying the decoding tool.

This tool will be used by the storage manager after digitization to reduce the physical space of audio files.

To simplify the integration of this tool in the transcription chain, the tool will be made available as a command line executable with the following syntax:

prog_name input_file output_file

where *input_file* is a BWF file, as produced by the transcription chain and *output_file* is a BWF-like file containing the compressed audio samples and the metadata extracted from the *input_file.*

The resources required are mainly constituted by CPU cycles and disk space.

This tool runs asynchronously from the main transcription chain. The only prerequisite is that the source BWF file must have already been generated and closed.

5.3.8 Decoding_tool

The decoding tool is a command line program that performs the decompression of an audio signal. The compressed audio to be decoded must be contained in a BWF-like file. The program produces at the output a BWF file containing the same metadata as the input file and the decompressed linear audio.

The purpose of the item is that of restoring the original PCM signal format for playback and editing. The decompression tool can be required in several places in the archive infrastructure, the most common of which are the user terminal and the export module of the archive.

To simplify the integration of this tool in the transcription chain, the tool will be made available as a command line executable with the following syntax:

prog_name input_file output_file

where *input_file* is a BWF-like file containing the compressed audio samples and *output_file* is the BWF file containing the decoded audio samples.

The resources required are mainly constituted by CPU cycles and disk space.This tool will be executed each time a user needs to recover the PCM form of an audio item for auditioning or editing. The running environment can be either the archive system or a user workstation.

Chapter 6 Key Link Technology in Film Preservation

6.1 Auto-re-splice (FT1)

Specifications for an automated device for mechanical restoration of 16mm films edited with adhesive tape

6.1.1 Position and definition of the problem

Introduction

Until 1975, a considerable amount of news and television dramas was produced in reversal (positive) 16mm film, and edited directly on the original using scotch tape splices. A great number of these pieces of film are original and unique samples. After ten years or more, some of these tape splices have become dry, dirty, opaque, and so fragile that they break immediately when processed trough a telecine without preparation, and the glue of other splices has become sticky, has shifted, and spread on the adjoining film spirals. Therefore the massive archived edited reversal 16mm film digitisation process begins with a preparation phase of the material including an important manual step, mainly cleaning and repairing tape splices.

This step requires considerable manpower hours. An average time of 20 hours per hour of programme is required, with a mean cost of 1000 Euros per hour of programme. It is worth noting that this corresponds to 10 times the cost of analogue video digitisation, and that the telecine transfer represents only 25% of the total cost of the process.

INA, RAI, and BBC are among the largest broadcast archives in Europe. A reduced survey, has given the following table, which demonstrate the European dimension of the problem.

Project stakes

INA and other major archive owners are currently launching important digitisation plans for their archive preservation. To achieve such an objective, BBC, RAI and INA are leading a European project with the purpose of reducing the very high costs of their preservation programs in which film preservation is one of the major parts.

As volumes and costs are very important, it is crucial to reduce as much as possible times and costs involved in the film preparation phase before transfer. If this cannot be achieved, preservation might not be done properly and finished on time and consequently many archived films could be lost.

Purposes of the project

Improving this preparation step by automating a major part of the task can reduce significantly the cost of the overall film process.

The target of the project is to cut down costs of the film repair step by 30 to 50%

To achieve this the project will develop a highly automated prototype device that is able to perform:

- guided and controlled film pieces transportation through the device
- old scotch tape detection and removal at splices
- film ends cleaning at the splices
- Re-joining ends in order to produce a new reel that can be handled by standard telecine or scanner

INA intends to lead the project and wishes to do that with the help of a subcontractor with suitable skills and knowledge.

Analysis of the feasibility studies

INA has evaluated the proposals of three pretenders that declared themselves interested in the development of an automated device. PHOTOMEC, MINAUS and BERTIN TECNOLOGIES.

The three candidates have prepared a feasibility study report, and have presented their conclusion to INA who were supported by an external consultant, from a company specialised in film handling equipment.

First the evaluation pointed out that the splice removal will be the critical phase of the project and the overall speed of the process will depend on the rapidity and efficiency of the techniques to develop. This is the “key point” that will require a deep research of processes.

The second point to consider is how an automated device will manage “incidents” like “will dry splices often break in the transport?” or will greasy splices stick in the transport”...All these parameters are to study and tests must be led with the subcontractor in the early steps, in order to decide the achievable degree of automation.

Results of evaluation

PHOTOMEC provided a non-convincing small report, which, as shown during their presentation on 27/11/2000, in Bry Sur Marne, demonstrated that they were not fully confident they could undertake the development of such a prototype. They however demonstrated their knowledge about film

They pointed out (like others) that removal of old splices is the critical point they concluded that they could not give a fully estimated cost of study and implementation at this stage before their study of processes is finished.

MINAUS provided a more complete report, with some good ideas on the removing of adhesive tapes, but they clearly underestimated the film handling difficulty. The main problem was that they were considering a film transport system and a splice detection system that were not realistic. Further discussions with the Chief Engineer, during the review on 13/10/2000, demonstrated that their knowledge of the film handling difficulties was not sufficient for the considered development, and that it would be difficult to convince them of the problems in their approach.

BERTIN Technologies provided a detailed report, with detailed schematics, on one of the possible ways for removing the adhesive tape, which they clearly identified as the only hazardous part of the work. They presented a complete design in which this option was integrated. In their proposal they pointed out that important time and costs-savings, for very large volumes (10000 hours and more), directly, should be achieved with an approach where the final system would consist of a pool of several devices (groups of 4 devices supervised by one operator, raising the operator's productivity. During the review, on 5/10/2000 they insisted that the first step was to test several approaches for tape removal, including high-pressure solvent jets, ultrasonic processing, dry ice, or a combination of these. A full study of all these processes was all these processes was part of the work, combined with an evaluation of the other sub-processes performance (detection, cutting devices), and would conclude on a detailed evaluation on the time required for each operation.

Further exchanges with BERTIN helped modifying their proposal so that it could fit into Presto work plan (time scale, costs), and that there would be a working single-unit prototype at the end of the development very close to a future industrial standalone device (Their initial plan included only a non-functional prototype). Recent exchanges, have led to the demonstration that even a single device on small volumes (from 500 hours to 2000 hours) can be time and cost saving. We have clarified expected performances, in a realistic way, and estimate return on investments.

These considerations demonstrate that the only eligible proposal was Bertin's one. (3380 kF-15 months, with checkpoint at 1310 kF-5 months). The First 6 months, a phase of research of processes should help to decide to make (or not) the prototype, provided the research has been successful (or not).

For an estimation of costs savings with BERTIN' s proposition, see appendix.

6.1.2 User requirements

Film condition and current practices

Film condition

- Film is reversal (positive) 16 mm edited with adhesive tape.
- Documents are often original and unique (there are no archived copies).
- The number of splices greatly varies from one document to another. For example, on a 3' news reel, 10 splices can be found, and in another reel more than 50 splices with sometimes a very short duration between 2 splices (shots of 2 to 3 seconds).
- Splices were made with adhesive tapes of different quality. In the same reel splices made at different times from previous restorations may be found.
- There are dry splices and sticky splices according to the type of tape used.
- Dry splices can be easily removed but sticky splices are still holding and dirt and glue are difficult to remove.
- Time spent on a sticky splice is 2 to 4 times as much as time spent on dry splices.
- Time spent by an operator on a dry splice: from 1 to 2 minutes.
- Time spent on a dirty splice: From 4 to 8 minutes
- Average time spent by an operator per splice to redo: 3 minutes
- Splices in the same film roll are generally homogeneous (all dry or all dirty and greasy)
- It may rarely happen that a film roll has the two types of splices.
- There are documents films with perforations on one side as well as on both sides.

- Films generally have separate magnetic recordings.
- Film may have retraction until 3% of its length.

Audio separate magnetic tracks (sep mag) are also sometimes edited with adhesive tape

Current Practices

- A film operator does the work.
- Dangerous chemical products are used. The operator works on a table with a film supply reel and a take up reel to wind film, under a vacuum hood under strict and ruled conditions of ventilation.

Work is done as follows:

- To remove the adhesive tape, the film is dipped in a trichlorethylene (TCE) solution (this product is authorised only under strict using conditions; other products exist and can be used but are much less efficient). The immersion duration varies according the nature of adhesive tape (dry or sticky).
- Once the old adhesive tape is removed, sticky traces and dirt on the two ends of film are very carefully eliminated with a soft towel and TCE. It requires care not to damage film at this step.
- Most of time, it is necessary to clean the whole film because damaged adhesive tape has left sticky traces on the rolled film.
- A new adhesive tape splice is made on both sides of the film. Tape in excess on the edges of film is removed with a razor blade.
- Quality of the new splice is visually tested.
- Film is wound on a new reel.
- If necessary, the work previously described is also done on audio reels. It differs only in that separate audio track is not dipped in TCE but TCE is spread softly on tape with a towel.
- It is important to preserve the initial duration of the film in order to keep synchronisation of audio and picture all along the document. Sometimes the film is too much damaged at a splice and it is necessary to remove one frame. The film operator compensates this by adding a black frame in the movie or removes one frame on the audio track.
- Finally, the operator verifies the film and checks synchronisation on a film-editing table, and corrects losses of synchronisation.

6.1.3 Functional requirements

The future device will be as automated as possible. All functions performed by the device will be «on line» and chained without interruption. The operator should only intervene:

- To introduce the film segments at the input of the device
- To control the quality of work
- To control the good operation of the device
- To maintain regularly the device
- To stop the automated process in case of an incident

The device will combine in one entity:

- Mechanical and electromechanical components
- Chemical components

It is required that the future device will:

- Transport and guide the film or pieces of film (if they are already disassembled), inserted by an operator, from a supply reel to a take up reel.
- Automatically detect the splices.
- Automatically detect damaged perforations and alert the operator.
- Remove old tape splices.
- Clean more deeply the two ends at each splice.
- Apply a differentiated process on film and audiotape and, if required, according the to type of film (it is not required that film and audio are processed simultaneously, but the proposed method should minimise time spent by the operator working on the device).
- Wipe the film off.
- Precisely position of the 2 ends of the film to be spliced again.
- Re-assemble the film with a new piece of adhesive tape or whatever method achieving the requirements described in VIII.
- Make new perforations in the tape, and remove tape in excess on edges, if necessary
- Allow a visual control of the quality of the new splice.
- Inform the operator of every kind of dysfunction.
- Allow the operator to stop the automated process and easily proceed manually on difficult points.

Critical requirements and expected results

Integrity requirements

The future device will never damage the film or picture.

The device will not modify the duration of each shot of the film thereby maintaining the synchronisation between picture and sound.

Security requirements

The device will safely transport and handle the film

New splices will be well accepted by all kind of telecine transport mechanisms, thereby avoiding unexpected damages

The future system will operate in a safe environment (ventilation.) but will also provide safe operating conditions for the operator (for example for the use of dangerous products).

Accessibility requirements

The device will allow a quick intervention in case of emergency. The operator should be able to unload the film easily and without damage.

All mechanical parts of the system will be easily accessible to allow a regular and easy maintenance.

Performances requirements

- A standalone device should at least allow 50% timesaving on each splice to redo.
- That means that the 3 ‘ average time spent by an operator manually should drop at least

- That means that the average 20 hours per hour of programme should drop to 10 hours.

Man machine interface

The operator will have adapted controls to drive the device.

He will get from the machine:
- The status of the device and information about failures.
- Information about damaged perforations
- A production histogram (numbers of repaired splices per minute, processed film length per hour, per day, cumulated.

6.1.4 Specifications

These specifications are based on preliminary BERTIN’ s studies and may vary according the results of the research of the best processes in the first phases of the project.

Processes

The heart and main difficult part of the device is the process for old splices removal and film tips cleaning. This step is the most critical because the original status of splices may strongly vary:

There should be only one process either the splice is dry or greasy and somewhat elastic. The system should be compliant with these two extreme situations and all intermediary situations.

The device will be able to detect the presence and position splice but not detect its physical and chemical status and cannot adapt the process to its status.

To achieve this, in the first phases of the project, different processes will be evaluated: thermic, cryogenic, mechanical, chemical…

For a detailed architecture of the device, see appendix.

6.2 Alternative handling, specific scanner & procedures (FT2)

Scanning devices currently in use can handle 4 to 8 frames per second at resolutions of 4K and require operator intervention throughout the process. A transfer process that can take six times programme duration, or longer, and requires highly skilled operators is simply too costly and time consuming to be considered as a realistic preservation transfer chain.

Scanning devices used in the large preservation transfer chains should be optimised for maximum speed of operation of the overall system. This activity consists of

- the development of a new and novel approach to film transport
- identifying areas where the operability of the equipment can improve on conventional scanning techniques
- defining appropriate modifications or tools to currently available components on the market where appropriate.

The specific requirements for this link are:

- Apply automation, where technically possible to scanning device.
- Achieve real time or real time plus scanning at 4K resolution.
- Develop minimal contact film transport to reduce the risk of damage and attain a steadier transfers than fixed light source and scanning head devices.

6.2.1 Hardware architecture

The hardware involved will take a different approach to film scanning than currently utilized by conventional methods of fixed light source and scanning head tele/datacines. To achieve real time or real time plus transfers at high resolution it is proposed that multiple scanning heads/cameras are mounted on a moving sledge which will travel over static film held in place with either negative or positive air pressure.

6.2.2 Software architecture

New software will have to be developed to deal with multiple simultaneous scanning head data and film to scanning sled synchronous movement.

6.2.3 Supporting_tools

Software that is being developed in the lossless compression key link will be employed as a supporting element to transport and data management software specific to the scanning device operation

6.3 Film Format Converter (FT3)

In order to improve workflow efficiency and decrease exploitation cost in digital film archives the concept of a single digital film master (within an archive) is introduced. The main target is to avoid multiple scans of film material, which is a time- and cost-consuming process. The specifications of a digital film master (scanning properties) for a given archive have to be defined according to the specific needs of that archive.

A second goal is to ensure the highest possible quality of the digital film/video material. The initial scan is made in the highest possible/necessary resolution. When any other (i.e. lower) resolution is needed this resolution is generated with a format conversion tool.

This conversion tool takes as input the output of a digital film scanner, usually a 24p film master. The output of the conversion tool is any desired video format (SDTV, HDTV, 50i, 4:3, 16:9) Hence the format conversion tool has to be able to support conversions in frame resolution and frame rate.

A software solution is suggested for this key link as it offers several advantages:

- Resolution independence: As hardware solutions are restricted by “real-time requirements” this limiting factor does not apply to software solutions. Therefore a software tool can support a higher resolution than a hardware box.

- Easy adaptation to emerging standards and new algorithms.

- Software solutions can serve as a basis for future hardware developments.

6.3.1 Hardware Architecture

This software format converter will be developed for running on a commercial of-the-shelf execution platform (PC). It is assumed that the execution platform can access mass storage (a disk system) for reading/writing image files from/to it, this means that the storage solution (e.g. local disk system, network file system, storage area network) the requirement is that this software module running on the execution platform can do standard file I/O operations for accessing image data stored on the mass storage solution.

6.3.2 Software architecture

An overview of the main software subunits is given in the following figure.

- Motion estimation is the first main step inside the format converter. It is essential for high quality format conversion.
- The general Image Sequence Sampling Format (ISSF) allows the handling of progressive and interlaced sampled input material.
- The Resampling unit is responsible for deriving the desired output format from the ISSF format.

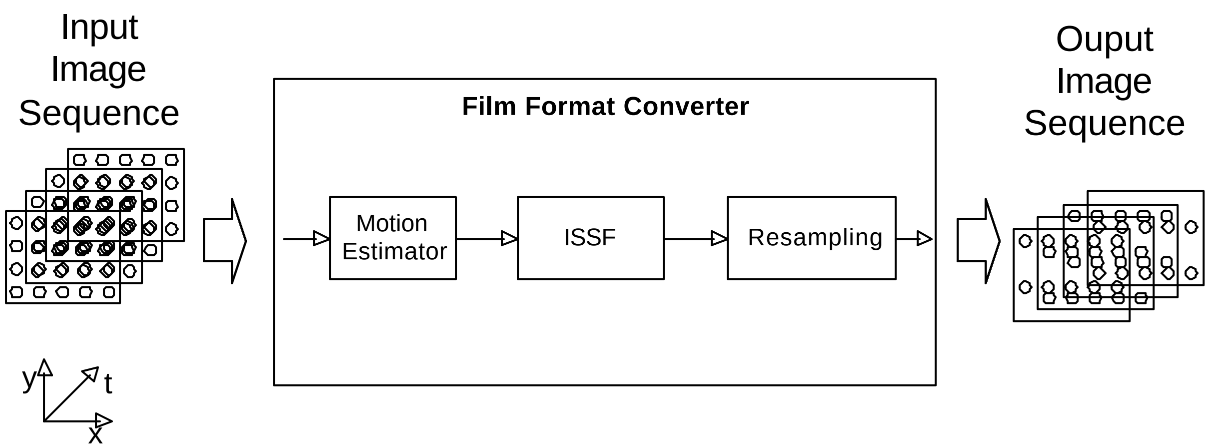

Figure 6.1: FT3-Software architecture

6.3.3 Film Format Converter

Type and Purpose

The film format converter is a software component, which allows the generation of different film, and video formats from one master film format. It bridges the gap between sampling structures of different progressive (p) film and interlaced (i) video standards. Accurate motion compensation enables highest standard converter quality.

Function

An input image sequence is transformed by the Film Format Converter to an output image sequence, see also Figure 6.1. Different properties of film and video standards have to be taken into account:

- different frame rates for film (e.g. 24p, 18p) and field rates for video (e.g. 50i, 60i),
- different system resolutions (e.g. SDTV-720x576, HDTV-1920x1080, 2k, 4k) and
- different aspect ratios for film (between 1.33:1 and 2.35:1) and video (4:3 and 16:9).

The following table distinguishes between base functionality, which will be developed inside the PRESTO project, and extended functionality which makes the module more versatile, but effort would exceed planned project budget, and thus cannot be implemented during the project.

Table 13: FT3-Base vs. extended functionalities

	Base Functionality	**Extended Functionality**
Frame/Field Rate	conversion from standard film frame rate (24p) to PAL interlaced video field rate (50i)	support of conversion from other progressive frame rates (18p, 24p) to other interlaced field rates (e.g. 50i, 60i) support of conversion from other interlaced field rates (e.g. 50i, 60i) to other progressive frame rates (18p, 24p, 60p)
Spatial Resolution	support of all different system resolutions (SDTV, HDTV, 2k, 4k) by a resolution independent approach	

Table 13: FT3-Base vs. extended functionalities

	Base Functionality	Extended Functionality
Densitometric Resolution	support of different densitometric quantisation accuracies (support of 8 and 16 bit colour depth per channel)	
Aspect Ratios	support of all standard aspect ratios of film and video. The definition of a rectangular region of interest (ROI) is possible for the case that not all the image area is used.	

Main development tasks are:

- A general Image Sequence Sampling Format (ISSF) has to be developed which is able to hold progressive and interlaced image sequence data. Parameters are the sampling structure, the frame/field rate and the spatial resolution.
- Development of ISSF conversion methods. They allow the conversion between ISSF's with different parameters, e.g. conversion from 24 progressive frames to 50 interlaced fields.

Interfaces

The film format converter will be developed as a standalone executable, where the following information is required for execution: input and output image sequence specification, input and output frame/field rate, region of interest definition on input image sequence, spatial resolution, densitometric resolution and aspect ratio of output image sequence.

This information can be transferred to the module in two ways. First, by command line parameters and second, by supporting the API/interface of an existing film manipulation system.

Resources/Compatibility

This software component will be developed in an operating system independent way, Win2000 and Linux are used for development and evaluation. Main memory consumption has to match the capabilities of current PC technology and should not exceed 512MB.

6.4 Lossless compression for film (FT4)

Proposal: Investigate the feasibility of the use of lossless compression in archive master material -- for the preservation of content originally held on film.

Purpose:

- reduce the overall cost of storage of film content on data tape
- reduce the bandwidth requirement for movement of full-quality film data

Background:

Data files of film content can be ten times larger than those of video with one hour of content scanned at 4K(4096 X 3112 lines) requiring a file size of 4.13 TB, with scanning time s of four to eight times programme duration. If grading or image manipulation is required this transfer time and cost (typically 40–60,000€ per hour) increases still further The benefit of transferring film at high resolution is capturing the quality of the original film material and the creation of a digital master, which can graded or re-versioned for purpose at a later stage using virtual telecine at the users desktop.

The current methods used for scanning film are simply just too time consuming and expensive to perform large scale preservation projects so we will have to look into developing tools that bring transfer times down to at least real time and reduce the data file size, whilst retaining the quality of the original film. Current professional broadcast tape formats do not possess the ability to record and store the size of file required for high resolution film data, therefore a data tape solution that fits in with both archive and broadcast systems will have to be explored employing a lossless compression solution that retains film quality and significantly reduces file sizes.

The main requirements for a lossless film compression tool are similar to those for audio and video:

- The decompressed signal must be bit-for-bit equal to the original
- It must be possible to implement the compression method in real time
- The compression algorithm must be fully specified and possibly standardized by a proper International Body
- The decompression tool should preferably be implemented in software and ideally run faster than real time on a conventional PC
- The decompression tool should be platform independent
- Partial file access should be possible
- A proper encapsulation must be defined; this is expected to be a part of the XMF format, currently being developed by EC project G-FORS.

Work to be performed:

Technology Assessment: Understanding what lossless compression offers.

- Assessment of state-of-the-art in lossless compression

- Implementation of state-of-the-art software models for encode / decode
- Use of this software on a variety of test material

Workflow assessment: An important consideration is the **practical** role of a new kind of compression in existing broadcast infrastructure. The study will need to look at the standard production and distribution methods in use, and being adopted. Key issues are:

- Ability of conventional distribution circuits to handle the lossless signal
- Relationship of the proposed lossless signal to the production and distribution requirement for DVB (digital video broadcasting; originally developed as a detailed specification by EC Project Eureka, headed by BBC R&D)

Outcome:

5) A statement of state-of-the-art in lossless film compression, including:

- Degree of lossless compression obtainable
- Process power required for encode and decode
- Implementation requirements

6) A workflow analysis of the potential contribution of lossless film to broadcasting

- Analysis of production workflow based on current film handling technology and on servers, and how lossless compression could be used
- Analysis of transmission workflow based on MPEG-2 (DVB), and how lossless compression could be used
- Interoperability: Relationship of lossless film to other forms of representation
- Transfer: Options for use of streamed and file transfers for lossless film

7) A set of software routines for lossless encode and decode

8) An example tape, showing what compression rates can be achieved on typical and on demanding material; such material has already been prepared by the BBC and others for tests of lossy compression, in particular MPEG-2

6.4.1 Hardware architecture

Encoding and decoding can in principle be performed either in hardware or software. For the purposes of a feasibility study, it well is assumed that hardware to implement encoding in real time could be produced if warranted, and a price estimate for this hardware will be made if needed. There will be no actual construction or purchase of hardware.

6.4.2 Software architecture

Encoder and decoder tools are software modules that can be launched by other programs. Depending upon their performance, they could run either in real time or in batch mode. The tools will be made available both for Windows and UNIX in the form of command line executables, for ease of integration in the control environment.

There are two main possibilities for implementation of the encoding process.

- the encoding is performed in real time while digitising
- the encoding is subsequently applied to the digitised files

In the first case the encoding software can probably be hosted by the acquisition hardware; in the second case there is a need for a set of encoding servers for batch processing.

Decoding: For audio file transfers, an evaluation is to be made of decoding at source vs. de-coding upon receipt. **For film and video, the use of lossless compression will be of little interest unless the decoding can be done upon receipt**, to allow reduced-bandwidth for the transfer. Therefore the standard architecture will require:

- the decoder to be hosted on the archive user workstation

While hardware decoding may be needed for real-time performance, for file transfers there is no real time constraint and therefore the decoding is envisaged as being purely in software.

6.4.3 References

A lossless compression algorithm, which can handle video and film, has been developed by Dr Xiaolin Wu of the University of Western Ontario, and is used in NTEC Medias CineCodec.

1998 International Conference on Image Processing, October 4-7, 1998 Chicago, Illinois, USA

PIECEWISE 2D AUTOREGRESSION FOR PREDICTIVE IMAGE CODING. Xiaolin Wu, University of Western Ontario; Kai Uwe Barthel, NTEC MEDIA GmbH; Wenhan Zhang, University of Western Ontario

ADAPTATION TO NONSTATIONARITY OF EMBEDDED WAVELET CODE STREAM. *08.09.Xiaolin Wu, University of Western Ontario; Kai Uwe Barthel, NTEC MEDIA GmbH; Gerhard Ruhl, NTEC MEDIA GmbH*

IMPROVED TECHNIQUES FOR LOSSLESS IMAGE COMPRESSION WITH REVERSIBLE INTEGER WAVELET TRANSFORMS. *WP09.08 N. Memon, Polytechnic University; X. Wu, University of Western Ontario; B.-L. Yeo, Intel Corporation*

LOSSLESS COMPRESSION OF CONTINUOUS-TONE IMAGES VIA CONTEXT SELECTION, QUANTIZATION, AND MODELING < ftp://ftp.csd.uwo.ca/pub/from_wu/papers/paper0.ps >by Xiaolin Wu, University of Western Ontario. A paper on high performance lossless image compression.

Chapter 7 Key links technology in Metadata

7.1 Common access to broadcast archives (Broadcast OPAC) (MT1)

One of the objectives of the PRESTO project is to develop an end-to-end chain of technologies in audio-visual content archiving for indexing, search & retrieval and exchanging of digital audio-visual content. The targeted application domain for this chain of technologies is the current public audio-visual archiving industry (public broadcasters) and its professional end users.

In their current situation this industry faces the problem of not being able to integrate or to access the available content with the demand for required content access or the exchange of relevant cultural heritage content in public audio-visual archiving industry. In order to bridge the gap between public content providers and professional end users standard access interfaces will be offered to enhance the access of relevant cultural heritage. As most archives are funded by public money, there will be a web interface for the broader public to access the archive holdings.

The problem obviously cannot be solved by tackling only one single technological aspect of the chain. The solutions require an impact on the complete workflow from content provision to content access and retrieval and should be built on open standards (e.g. MPEG-7) to allow easy integration into up-to-date middleware solutions which are used in the content providing archives.

7.1.1 General architecture

To get a better view of the planned architecture the following schematic figure is provided.

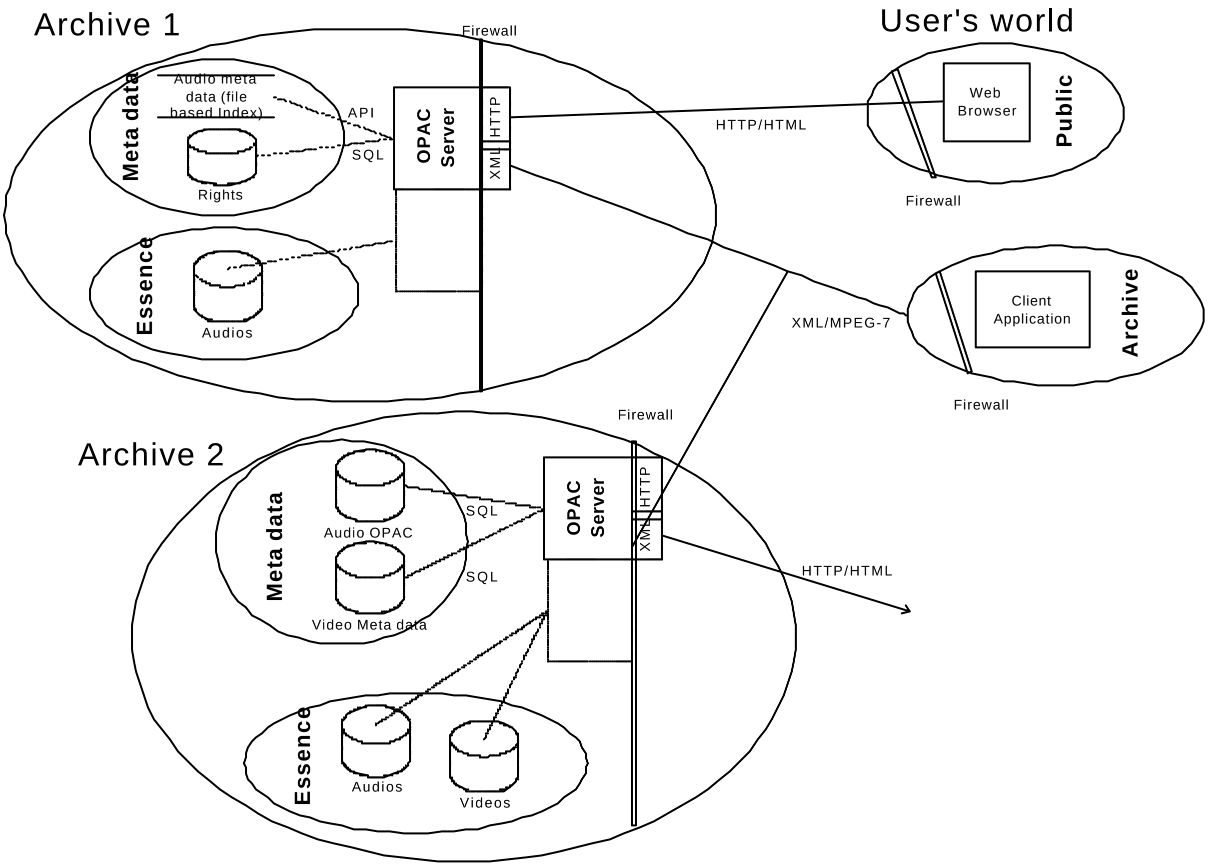

Figure 7.1: MT1-Overview of system architecture

The basic requirements are that distributed archives (e.g. Archive 1 and Archive 2) provide a standard access interface to the archive via the Internet. In the archive different audio-visual content and descriptions are available. This can be audio, video, images and text.

The user may be outside of the professional archive (typical professional end user who works in the content production community). The goal of the end user is to search and retrieve the interesting cultural heritage content in the archive in a simple way. This typically is done by a web-based interface by using a web browser. Another type of end user is an archive. The aim of the archives is to exchange the available content for reducing the production costs in the archives. In such a case the end user application is an application which should communicate via a standard interface to the archive. Moreover, technical limitations like firewalls are considered in this architecture, as well. The technical solutions have to be flexible and open to changes on the requirements of the search and retrieval clients, e.g. the search attributes will be easily configurable. For the definition of the interface standards like MPEG-7 will be considered.

The interfaces support different kind of media types (e.g. audio, video, image and text). Each supported kind of media type should be describable in one or more MPEG-7 description schemes.

7.1.2 Hardware architecture

The Broadcast OPAC software will be designed in a way that it will be running on state-of-the-art PC based server platform. This includes

- state-of-the art processor speed
- sufficient memory

- redundant disk array
- permanent internet connectivity

7.1.3 Software architecture

The Broadcast OPAC will consist of three different layers, which are described below and are shown in Figure 7.1.

- The **Access Layer:** This layer contains the public interface including the search & retrieval and browsing facilities.

- The **Aggregation Layer** contains the core of the Broadcast OPAC, which includes facilities for information integration, administration, user registration, interfaces to the information resource, etc.

- The **Data Layer** contains the stored meta-data. It provides the interface to existing data sources and legacy databases.

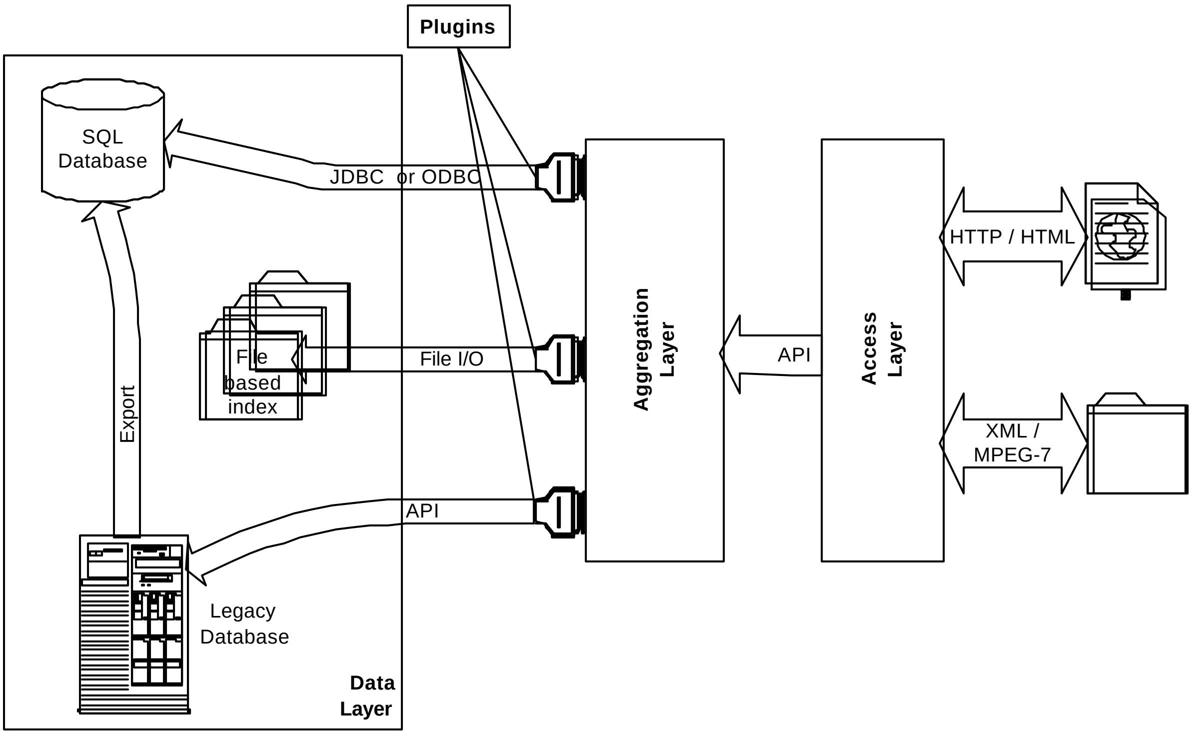

Figure 7.1: MT1-Software Layers in the Broadcast OPAC

The following list defines the required software items:

In general the implemented solution will be platform independent, however during development and testing phase one of these operating system will be used.

Environment	**Description**
Run time	• **Operation System:** - Linux Kernel 2.2 or higher - MS Windows NT 4.0 or MS Windows 2000 • **Web Server:** - Apache 1.3.x or higher - MS Internet Information Server 4.x or higher

- **Middle-ware tools:**
  SOAP Toolkit:
    - Apache, SOAP 2.1
    - Microsoft SOAP Toolkit (for Visual Studio 6.0)
- **Database:**
  - Oracle
  - MS SQL Server

7.1.4 Access Layer

Collections are accessed either though a standard Web browser or a SOAP compatible client. The interface offers search and browsing facilities. Occasional users will be provided a very simple query interface (e.g. a single line for full text) while expert or professional users want to make more sophisticated searches. An advanced search form which includes attribute search (if supported by the media types). So the system supports both user groups by providing different kinds of query interfaces.

Type

The user interfaces will be implemented in either of two modules:

- The HTML/HTTP interface are HTML pages (e.g. example templates) and dynamical web pages (based on PHP4/JSP/ASP technology) which will interface the Broadcast OPAC middle-ware.

- The XML/SOAP interface is using a SOAP implementation (e.g. Apache SOAP 2.1) which encapsulates the functionality of the Broadcast OPAC in a number of functions accessible via the SOAP protocol.

Function

In the following subsections the search user interface, the presentation user interface and the browse user interface are described.

Search User Interface

Typical searches performed by a user can be divided into the following two groups:

- Full text searches like “Find everything related to the *PRESTO project*”. The user only specifies the search terms “PRESTO” and “project” and the system scans all textual annotations in the database where these terms occur and reports all items that match the condition(s).
- Attribute searches like “Find all *documentaries* on *Bill Clinton* aired in *2000*”. The user fills out a search form where he specifies a query like “genre == documentary” AND “subject == Bill Clinton” AND “date == 2000”. This allows the user to specify a more detailed query than the first case

The user interfaces for simple searches contains one text field for entering the search (terms).

For searches the user interface will be available with several elements:

- Areas for sub queries containing attribute fields, text fields
- Operators for combining sub queries
- A range selector for the number of returned results

Attribute fields

The attribute fields allow identifying the attributes to be searched. This can be done either in form of list boxes or in a hard wired way. The full text search can be either one “special” attribute or an additional field.

Search attributes

Following is the list of searchable attributes (FIAT/IFTA Minimum Data List definitions taken from **[1]**). The “*” denotes attributes which will be also used in the browse user interface. Three areas are covered by the set of attributes: Identification, Technical and Legal.

Identification area

- **Title**
  Denomination given to a production by its producer.
- **Given title**
  Denomination given by the archivist when the proper title is missing.
- **Subtitle**
  Secondary title in the case of unique production. Title of each part of a series production. Part title of each item within a production consisting off several subjects.
- **Other titles**
  Any other title identifying a production, including original title if not given above.
- **Date of transmission***
  Date of first public transmission by air or by cable.
- **Date of shooting***
  Could include several dates covering shooting over a period of time.
- **Producer***
  Person who organises and directs the operations necessary to make a programme (see also Producer in legal area below).
- **Production number**
  Unique number given to a programme for administrative purposes.
- **Archive number***
  Unique identification number given by the archive.

Technical area

- **Content**
  Summary of the subject described in a production.
- **Keyword***
  Word or group of words, possibly in lexicographically standardized form, taken out of a title ore the text of a document characterising its content and enabling its retrieval.
- **Place of shooting***
  Place(s) of shooting of the programme.
- **Running time**
  Duration of the transmission period used for a production, under normal conditions for the medium used.
- **Language**
  Language used in a production (may include also information on e.g. different versions of a multi-track videotape).
- **Medium**
  Nature of the carrier on which the production is made (film, videotape, disc etc). Also comments on quality.
- **Format and Standard**
  Gauge of film, tape width and line standard (525, 625 line etc.), analogue or digital standard.

- **Sound recorded**
  Nature of the procedure of sound registration (eventually including mute of international sound track). Also note on analogue or digital sound.
- **Colour and / or black and white**
  System of colour for film (e.g. Technicolor, Kodachrome, Agfacolor, etc.) and for videotape (PAL, NTSC, SECAM, etc.) and for discs.

Legal area

- **Origin**
  Gives an indication of how the material is acquired and where it comes from.
- **Contract**
  Agreement concerning copyrights and other conditions for a programme (may also include a summary of conditions from the contract like, period covered by the contract, names of participants, financial arrangements and payments to participants, conditions for distribution and screening).
- **Copyright**
  Designation of the person(s) or organisation(s) holding the rights to make use of a production (specification of copyright holders). It may be specifically noted whether all rights rest with the archive or not (with or without reference to a contract) or they may be reference to the contract only.
- **Producer***
  An individual or legal entity under whose initiative and responsibility fixation of a work is first made (see also producer in identification area).
- **Other names***
  All other names of significance, if possible combined with function, and referring to the realisation of a production and bearing specific rights.

Entering search terms

The text fields are used for entering the search terms or a combination of terms. The simple search is attached to the full text search and in the advanced search each text field is attached to exactly one attribute field. The syntax that can be used in the text field is explained below. Date related information can be handled in two text fields used for the beginning and the end of a period to be used in a search.

Search grammar and syntax

Within the text fields of the search user interface not only single words can be entered but this can be also structured according to a search grammer. Some rules apply to entered search terms respectively combinations of such terms:

- Queries are automatically handled as “AND” queries (i.e. all terms must be found in one record to have that record included in the result). So no “AND” has to be included in a query.
- To get results including one of the entered search terms the word “OR” in capital letters can be used. The “OR” prevails the implicit “and” (e.g. “video scientific OR historical” means all videos being either scientific or historical).
- Common words or single characters (usually called stop words) will be excluded in standard searches. To explicitly include them in searches the “+” character is used before the stop word.
- Search terms are not handled case sensitive (e.g. “Video”, “video”, “VIDEO” will return the same results).
- Refining queries can be done in two ways. As already stated adding a new term means restricting query results to another term. The other possibility is to exclude results which contain a specific term. To refine a query in this direction the term is entered with the “-“ sign in front. A space has to appear before the “-“ sign.

- Phrase searches can be done by enclosing the phrases with double quotes. Such queries will result in records containing the whole phrases except the stop-words if included.

Combination of queries

Logical operators allow the combination of sub queries within the user interface. There are 2 ways how the operators can be done: the more technical version (set based with AND, OR and AND_NOT) or the more user oriented approach (using “Matches ALL”, “Matches ANY” and “Excluding”).

Browse User Interface

The browse user interface offers two fields: an index field and a text field.

- The index field allows the user to choose the index to be browsed. The attributes for indexes that could be candidates for browsing are included in the search attributes list and are marked with “*” in the list given in the previous chapter.
- In the text field the term is entered which denotes the entry point into the index (i.e. where the result list starts).

The terms of the index are offered in form of a list. Each item includes an occurrences count (i.e. the number of objects indexed with that term). A search with a specific item can be initiated by selecting the index item

Interfaces

Two different types of interfaces to the PRESTO Broadcast OPAC are be provided, namely an HTML/HTTP interface and an XML/SOAP interface.

The public access to broadcast archives could be seen as a special Web service for searching, retrieval and access meta data descriptions available in broadcast archives. For the purposes of this specification, the “public accessible interface” is a Web service that performs a specific task, and conforms to a specific set of technical specifications which make it interoperable with compatible components. The key architectural principles are the service-oriented architecture as described in this specification.

The goal of the PRESTO Broadcast OPAC design is to define a clear publicly available interface to the archive data by using open standards.

- **HTML, HTTP:** A simple web interface to the OPAC database will be provided. It will include search pages which will allow the user to specify queries by selecting search attributes and search terms, but will also enable him to launch full text queries. Formatted search results will be presented to the user. A browse interface enables the user to access archive holdings without necessarily knowing what to search for (a common situation for first searches on a lot of search & retrieval systems around).

- **XML / MPEG – 7:** Search results will be provided in a standardised way by delivering XML- and MPEG-7 compliant data records. The application of these standards will enable system-to-system communication and will be used by rich client applications.

- **SOAP:** The transport protocol for exchanging messages will be the RPC-style message protocol SOAP. Existing RPC-style protocols such as DCOM and IIOP (CORBA) have not proven to be adaptable to the Internet. Both of these protocols require a non-trivial amount of dedicated runtime support in order to implement the complete set of services that both protocols have to offer. Finally, the existing Internet security infrastructure has embraced HTTP to the point that trying to communicate across organizations

organizations using anything else than HTTP requires an excessive amount of organizational and engineering resources. To support rich application clients which use the functionality of the OPAC framework in their applications they should have the possibility to access the framework via XML/HTTP. One of the advantages is that firewall systems do not require reconfigurations. Advanced features for automatic search & retrieval, download request processing can be integrated in several client tools.

HTML, HTTP

A simple web interface to the OPAC database will be provided. It will include search pages which will allow the user to specify queries by selecting search attributes and search terms, but will also enable him to launch full text queries. Formatted search results will be presented to the user. A browse interface enables the user to access archive holdings without necessarily knowing what to search for (a common situation for first searches on a lot of search & retrieval systems around).

XML / MPEG–7 & SOAP

Search results will be provided in a standardised way by delivering XML- and MPEG-7 compliant data records (see **[3], [5]**). The application of these standards will enable system-to-system communication and will be used by rich client applications.

The transport protocol for exchanging messages will be the Remote Procedure Call (RPC)-style message protocol SOAP **[11]**. SOAP is a lightweight protocol for exchange of information in a decentralized, distributed environment. SOAP can be used in combination with a variety of protocols; however, the only bindings used in the Presto project will be the SOAP in combination with HTTP and HTTP Extension Framework.

SOAP consists of three parts:

- The SOAP envelope construct defines an overall framework for expressing the content of the message (e.g. definition of the search query).
- The SOAP encoding rules defines a serialization mechanism that can be used to exchange instances of application-defined datatypes.
- The SOAP RPC representation defines a convention that can be used to represent remote procedure calls and responses.

One advantage of SOAP is its text-based protocol nature by using XML. For example, it is easier to debug applications based on SOAP because it is much easier to read XML than a binary stream. The body of the request is in XML. A procedure executes on the server and the value it returns is also formatted in XML. The binary streams refer to the remote procedure call itself, not to the results set which can contain binary streams. Procedure parameters and returned values can be scalars, numbers, strings, dates, etc.; and can also be complex record, list structures and MIME types like e.g. images and Internet enabled video streams.

Existing RPC - style protocols such as DCOM and IIOP (CORBA) are binary formats and have not proven to be adaptable to the Internet. Both of these protocols require a non-trivial amount of dedicated runtime support in order to implement the complete set of services that both protocols have to offer. Finally, the existing Internet security infrastructure has embraced HTTP to the point that trying to communicate across organizations using anything else than HTTP requires an excessive amount of organizational and engineering resources. To support rich application clients which use the functionality of the OPAC framework in their applications they should have the possibility to access the framework via XML/HTTP. One of the advantages is that firewall systems do not require reconfigurations. Advanced features for automatic search & retrieval, download request processing can be integrated in several client tools.

7.1.5 Aggregation Layer

The Aggregation Layer of the Broadcast OPAC consists of several different modules, namely:

- Translator module (query translation, extension and gather the result set)
- Administration modules
- Complementary features including the session manager, user registration, logging facilities etc.
- Multiple language module

Type

The Broadcast OPAC will be a lightweight middle-ware software programme (using an application server) which consists of different modules.

Function

The main functionality of the Aggregation Layer is the transformation of incoming query requests into valid data store requests (e.g. in the moment on a relational database or in the future whatever the data store will look like). After execution of the database queries and retrieval of results from the database the found results are converted into a form which can be returned to the requesting party.

Translator module

The translator module receives the search request from the access layer. The translator module extends the search request and translates it into the format of the underlying data resources, distributes it to the various data resources and delivers the results from the search targets to the access layer later on.

Configuration database assistance

The configuration database assists in two tasks:

- Translation of the query elements and entered terms from the user interface into the XML query presentation
- Translation of returned XML formatted results into a presentable format for the user interface

XML represented query

The XML formatted query requests will contain:

- identifiers for the databases to be queried (in case that more than one database exists),
- binary tree structures using operators like AND, OR and AND NOT in the nodes of the tree and the operands (query terms) in the leafs of the tree,
- operands being combinations of attributes and search terms,
- number of records to be returned in the direct response to a search,
- sort criteria (attribute to be used for sorting of results).

The exact list of necessary query elements will be defined in an XML schema during the design phase of the software development.

XML represented records

The XML record representation contains the elements as described in the section about search attributes. Additional elements may be added to that set of search attributes as needed by the archives. More information about this will become available in the following months of the project. Normally this should not cause a serious problem to the use of the system. A proper design of the software will take care of these configuration aspects of the software. Therefore a detailed specification of an XML schema is not needed at the moment. The potential use of MPEG-7 also influences the design of the schema and the availability of appropriate parsers for MPEG-7 formatted files has to be investigated.

Beside the records another type of information may be transported within the XML records. This information provides diagnostic feedback on the query execution and evaluation. This is not an issue for the system developed in PRESTO at the moment. But in case there will be other systems interfacing the OPAC server in the future appropriate problem resolving mechanisms are a necessary part of such a system. Therefore the OPAC system's response messages will be designed in a way to also provide this type of information if necessary.

Administration module

Maintenance should be easy with the provision monitoring tools. The administration module consists of 3 modules.

The application server is equipped with an administration tool. This tool allows a systems administrator to configure and monitor the execution of the application server.

The Broadcast OPAC needs configuration data. This data is stored in a configuration database.

This database contains following data (at least):

- Data sources connected to the system,
- Information about these data sources (e.g. address, port number, database name etc.
- Search attributes and mapping rules

Facilitating the access and exchange of information between two audio-visual archives applications is a lot like translating between two people speaking different languages. In a complex environment like an audio-visual archive network, however, such differences must be resolved in a way that enables all participants to take part in the communication process. This is far more difficult challenge. However, the OPAC framework should support the user with a clear transformation tool (using the standard MPEG-7) to resolve the differences between the data models or the import/export formats. The rule-based transformation and mapping as part of the system administration console will be implemented. Basically, the system supports the mapping of one input schema to a common output schema.

Complementary features

These basic modules are building a framework which is used in the development process. Each element of this framework is described in the following subsections.

User registration / management

The access to the PRESTO facilities should be restricted by a user registration and login procedure (for a user name and a password). No user should be allowed to use the system without going through this authorization procedure. The registration module will support the definition of individuals or groups.

Session management

The Broadcast OPAC stores temporary data like user session identifiers to keep track of the state of a connection. The session identifier is needed to identify the user who submitted a request. Because more than one request (from different users) may be sent to the Broadcast OPAC concurrently the results of each request have to be managed to the corresponding user session.

Logging

Following logging services should be possible:

- session tracking
- session logging of the underlying database access and index access
- query and result set
- performance and event logs.

Multilingual support

Within the Broadcast OPAC there are several areas which requires multilingual requirements.

- Thesaurus support
- Dates / times standards
- Character sets – Unicode support

Multilingual thesaurus:

A multilingual thesaurus support for query expansion with controlled vocabulary would be useful. In fact, within the Presto Broadcast archive no multilingual thesaurus will be developed, but in an related EU-project AMICITIA **[12]** such a multilingual thesaurus tool will be developed by JOANNEUM Research. That tool will be evaluated within project PRESTO.

Dates / times standard:

A standard for dates/ times should be also considered within the aggregation of the search results of several media archives. A common standard is ISO 8601.

Character sets – Unicode support :

Unicode (namely UTF-8) will be used throughout the software, allowing different languages to be processed in a consistent manner. Several languages can be supported. Some kind of on-the-fly conversion is used to convert from Unicode to an alphabet supported by the user's application client.

The Unicode standard is the universal character encoding standard used for representation of text for computer processing. Unicode provides a consistent way of encoding

encoding multilingual plain text and brings order to a chaotic state of affairs that has made it difficult to exchange text files internationally. Computer users who deal with multilingual texts (e.g. business people, archivists, linguists, researchers, scientists) will find that the Unicode standard greatly simplifies their work **[6].**

Unicode uses 2 bytes for each character. Therefore 65536 characters are possible (only 40000 are used in practice, see also Unicode web pages). The first 256 characters of Unicode are the same as in the ISO Latin-1 standard. This makes the parsing process of Unicode data simple. The first two bytes of a file are the first character. The next two bytes are the second character, and so on.

Within the Broadcast OPAC the character encoding standards based on UTF-8 will be used. It follows a list of the Unicode support in the Broadcast OPAC software including browsers, protocols, databases and programming languages.

Table 14: MT1-Unicode support in the Broadcast OPAC

Area	Description	Supported
Web browser	MS Internet Explorer 4.0 or higher, Netscape 4.0 or higher Opera 4.0 or higher	✓
HTML & XML	Language codes are used to indicate the language of text in HTML and XML documents **[7].** For HTML 4.0, language codes are specified with the “lang” attribute. For XML, language codes are given in the xml:lang attribute. In HTML / XML information is inherited along the document hierarchy, i.e. it has to be given only once if the whole document is in one language. Language information nests, i.e. inner attributes overwrite outer attributes. HTML and XML rely on RFC 1766 to define language codes. RFC 1766 is in turn based on ISO-639 two-letter language codes, and on ISO-3166 two-letter country codes.	✓
HTTP / SOAP	HTTP 1.1: It contains the following internationalization features indicating: the used character encoding of a page (char set parameter), in the character encoding understood by the client to the server, the character encoding of a page sent from the server to the client (char set parameter) and the language(s) of a page sent from the server to the client (using the Content-Language in the response header).	✓
Databases	Oracle **[8]** SQL Server **[10]** Informix **[9]**	✓

Table 14: MT1-Unicode support in the Broadcast OPAC

Area	Description	Supported
Programming Languages	JAVA: JAVA is one of the few programming languages to explicitly address the need for non-English text. JAVA understands several dozen different character sets for a variety of languages. Internally, JAVA uses a form of UTF-8. C++: Unicode Strings are supported in Visual C++ (see also **[10]**).	✓

For improved search functionality on the server side the use of simple translation tables for frequently used terms is foreseen. These tables would allow translation between common languages. As an alternative the set-up of a (multilingual) thesaurus may assist the search processes. The creation of translation tables or appropriate thesaurus structures is a task for the end user partners of the consortium or eventually the BAUG members due to their detailed knowledge of the specific domain. From previous experiences it is known that the creation of properly working list or structures is not an easy task and it normally cannot be done within the remaining time frame of this project. Therefore the implementation and integration of such a component is not given very high priority at the moment and so it might not be included in the final software version within the project. Nevertheless the attention of the development team will be also turned to this topic during the design phase. The definition of an open interface for that purpose is planned. This will allow an easier integration at a later stage (e.g. in follow up activities to the project).

Scalability

The application should be scalable regarding two major aspects:

- Size of user group: The Broadcast OPAC should support a very large number of concurrent users accessing the database.
- Amount of accessible data: The data repositories should support very large amount of data. It is essential in the development framework that the underlying background systems (indexing, multi-media database etc.) support the scalability of the solutions.

Open Interfaces

The Broadcast OPAC's Aggregation Layer has an internal interface to the access layer and includes several wrapper interfaces which can be added to the software via plug-in techniques.

This allows interfaces to systems like

- SQL databases accessed via ODBC / JDBC
- New Zealand digital library software **[4]**

to be supported with reasonable additional effort.

7.1.6 Data layer

Within the data layer the meta data are stored. There are several possibilities to store and access the meta data:

Method 1: Access to a copy of the metadata of the production system: The data is exported of the legacy (relational) database and the Broadcast OPAC directly accesses this da-

database. This could be a possible solution, if the database schema of the copy is well designed to support fast retrieval of the data records.

Method 2: Access to a copy of the metadata which transform the used database schema to a very fast search & retrieval enabled database schema.

Method 3: Another possibility which could be supported within the Broadcast OPAC is to access the content in a file based index (e.g. New Zealand Library).

Method 4: Direct connection to production/legacy database. This is not recommended at least not for the prototype.

These are the methods that can be used implementing the system. However the actual methods used in the Broadcast OPAC will highly depend on the data sources which are currently available in the archives. Therefore a details description is not given here and is specified later in the implementation phase.

Type

The data layer is either a database schema adapted and enhanced within the PRESTO project or uses already available database schemas.

If method 2 (Access to a copy of the metadata) is used, scripts will be provided which transform the available metadata into a new database schema which is used by the Broadcast OPAC.

Purpose

Within the data layer the performance of accessing the data is an important issue. Therefore it is recommended to do performance tests with production data.

Function

The software product in this layer are based on databases or other retrieval software with a well defined client interface.

Interfaces

There are two possible interfaces:

- The interfaces to the database are provided via JDBC / ODBC drivers.
- The underlying retrieval software provides a well defined API which can be used by the adapters.

7.1.7 References

[1] Annemieke de Jong: **Metadata in the audiovisual production environment**; ©2000 Netherlands Audiovisueel Archief

European Broadcasting Union: ***PMC Project P/META (Metadata exchange standards)****; http://www.ebu.ch/pmc_meta.html*

[2] **MPEG-7 main page**; GMD - Forschungszentrum Informationstechnik GmbH; http://www.darmstadt.gmd.de/mobile/MPEG7/index.html

[3] **The New Zealand Digital Library Project**; http://www.nzdl.org/

[4] Philippe Salembier: **Status of MPEG-7: the Content Description Standard**; *International Broadcasting Conference Amsterdam, The Netherlands, September 8, 2000*; Universitat Politecnica de Catalunya, Barcelona, Spain

[5] The **Unicode Standard**, http://www.unicode.org

[6] Character sets on the web, http://www.w3.org/International/O-charset.html

[7] Oracle, http://www.oracle.com

[8] Informix, http://www.informix.com

[9] Microsoft, http://www.www.microsoft.com

[10] The **SOAP protocol**, http://www.w3.org/TR/SOAP/

[11] Amicitia, http://www.amicitia-project.net

[12] THE ISO Standard 8601, http://www.iso.ch/markete/8601.pdf

Chapter 8 Appendix 1 - Estimation of costs savings with BERTIN' s proposition for Auto-re-splice

The following tables and calculations are only a few examples to demonstrate that automated devices should be cost saving and adapted to different size of holdings.

Initial parameters:

- Estimated cost of an industrial standalone device: 122 K Euros (800KFF)
- Estimated cost of maintenance and provision: 15.2 K Euros /year (100 KFF)
- Expected life of a device: more than 12 years
- Expected performance: reduce average time spent on each splice by 50% time.
- Average time for a manual process: 3 '
- Expected time with an automated device: 1'30
- Average time per hour of programme (manual process): 20 hours
- Average time per hour of programme (automated process): 10 hours
- Average complete operator's costs(including room and environment): 45.7 Euros per working hour (300FF)

- an operator works about 1575 hours a year.(7 hours a day, 225 days a year)

NOTES: 50% is a realistic objective because automatic could be done simultaneously on the two tips and the two faces. On the other hand, an operator cannot keep its attention all day long on a boring task so the total time spent on a film is significantly longer than the average time multiplied by the number of splices.

Calculation mode

- P: number of hours of Programmes to restore.
- Wm: **W**orking hours required in **m**anual mode (Wm=20 x P)
- Wa: **W**orking hours required in **a**utomated mode (Wa=10xP)
- N: number of operators
- O: cost of an **O**perator per working hour (FF/h or Euros /h)
- D: cost of a standalone **D**evice.
- U: cost of maintenance and provision per year
- M: total cost of a **M**anual process (mainly operators' costs)
- A: total cost of an **A**utomated process (A = costs of devices +costs of maintenance and provision + cost of operators)
- R: **R**eturn on investments
- Ym: number of working **Y**ears required in a **m**anual process.
- Ya: number of working **Y**ears required in an **a**utomated process

- Ym= Wm/1575xN
- Ya= Wa/1575xNxP
- M =Wm x O
- A= (P x D) + (Ya x U) +(1575 x Y x O x N)
- R= M - A

Case 1:Return on investments for holdings of 500 hours of spliced films to restore.

- One device, one operator
- A manual process would require 6,4 years for one operator
- An automated process. Corresponds to about 3.2 years of one operator.

	calculation	**K FF (1000FF)**	**K Euros (1000Euros)**
Manual process (M)			
Total cost (operator)	500x20x300	**3 000**	**457.3**
Automated process (A)			
Cost of the device		800	122
Maintenance and provision	3x100 (3 years)	300	
Operator's costs	1575x3,2x300	1500	228.7
Total cost for an automated process		**2600**	**396.3**
Return on investment (M-A)	3000-2600	**400**	**61**

Case 2:return on investments for 2000 hours of spliced films to restore.

- One device, one operator
- A manual process would require 25,4 years for one operator
- An automated process would require 12,7 years

	calculation	**K FF (1000FF)**	**K Euros (1000Euros)**
Manual process (M)			
Total cost (operator)	20x2000x300 FF	**12000**	**1830**
Automated process (A)			
Cost of the device	800KF	800	122
Maintenance and provision	13x100 KF	1300	193
Operator's costs	1575x12,7x300	6000	915
Total cost for an automated process		**8100**	**1235**
Return on investment (M-A)		**3900**	**595**

Case 3:return on investments for 2000 hours of spliced films to restore.

- 2devices, 1 operator.
- A manual process would require 26 years with one operator. (13 with 2 …)
- An automated process would require about 6,5 years of one operator (2 devices and 50% time savings)

	calculation	K FF (1000FF)	K Euros (1000Euros)
Manual process (M)			
Total cost (operators)	20x2000x300 FF	**12000**	**18293**
Automated process (A)			
Cost of the devices	2x800KF	1600	244
Maintenance and provision	7*2*100KF	1400	193
Operator's costs	1575x6,5x300	3070	468
Total cost for an automated process		**5800**	**925**
Return on investment (M-A)		**5930**	**904**

Case4: return on investments for 10 000 hours of spliced films to restore.

- 8 devices 2 operators
- A manual process would require 63,5 years with 2 operators.
- An automated process will correspond to about 7,9 years.

	calculation	K FF (1000FF)	K Euros (1000Euros)
Manual process (M)			
Total cost (operators costs)	20x10 000x300 FF	**60 000**	**9147**
Automated process (A)			
Cost of the devices	8x800KF	6400	976
Maintenance and provision (8 years)	8x8x100KF	6400	976
Operator's costs	1575x2x8x300	7500	1143
Total cost for an automated process		**20300**	**3094**
Return on investment (M-A)		**6200**	**6052**

Chapter 9 Appendix 2 - Auto-re-splice (FT1) architecture

In order to remain open to different possible processes, to facilitate exploitation, maintenance and manual control by the operator, the functions of the device have been separated into 2 groups:

- The first one will integrate all the mechanical functions
- the second all the splice removing and cleaning functions

The device will be composed of 2 modules:
- a transport and repair module (TM: transport module)
- a cleaning module (CM)

These 2 modules are independent and separable.

The overall architecture is organised according the following steps:
- film transport and continuous exploration to detect splices
- detection of a splice
- transport of the film in an indexed operational position
- the splice is cut on the junction line
- tape and glue is eliminated on the two tips simultaneously
- the film is cleaned on the two tips simultaneously
- the film is dried on the two tips simultaneously
- a new splice is made
- the film can move until the next detected splice

The mechanical functions of the transport module are:

- film transport
- detection of splices
- indexing the junction
- disassembling
- re splicing

The process functions of the cleaning module are:
- old splices and glue elimination
- cleaning of the two tips
- drying the two tips

In order to facilitate use of liquid baths and guidance, the film will be transported horizontally.

splice detection

The presence of a splice is represented by a local thickness of 8 to 10µm.

Detection will be multi dimensional: thickness but also length in order to avoid detection of glue or dirt on other parts of the film

Detection can be mechanical with oscillating rollers with capacitive detectors and or laser scanning.

In the two cases the signal will be processed: detection of the beginning, amplitude and length of the pulse) in order to bring the splice in a precise geometric position with coders and counters.

Junction cutting

The control of the machine will allow bringing the junction of the film in a precise position. The cutting is made by “guillotine”.

Splice removal

Continuous or pulsed low-pressure jets of solvent (smaller than 100 bars) will remove the splices. These jets will cover all the width of the film. The temperature of the solvent may be different of ambient T°.

Cleaning of the tips

It will be made in a solvent bath with ultrasound assistance (optional) and a light cleaning with brushes.

Drying film tips

Warm and dry air jets on the cleaned tips will dry the film tips.

New splice

If possible, the new splice will be made with a commercial automated splicer adapted to the transport of the machine.

Detection of damaged perforations

A process like artificial vision could be used.

Architecture of the transport module (TR)

Sub-frame and mobile chariot

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The TR is composed of a U sub-frame in the horizontal plan. The sub- frame is on 3 anti vibration feet whose height can be set. The closing part of the U has a guidance system allowing the movement of a chariot. Motorization is electric with transmission of the movement by a notched driving belt. The chariot positions will be controlled by an incremental coding system

Film Transport

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The chariot described above wears a vertical plate. This plate can be tipped over to allow an intervention of the operator. Its nominal position is locked on the chariot.

On the plate there are two motorised hubs driven by an electric motor whose speed is controlled. Speed is a set up parameter of the machine.

The system will ensure a constant speed and constant tension of the film whatever the reels diameters are. The tension is a set-up parameter of the machine.

- The transport plate will include the splice detection system.
- The film indexing system
- The rotating grips pinching the film tips

Rotating grips

Once the film junction is perfectly positioned, the two sides of the unspliced film are gripped by specific grips. Once the splice cut, these grips can rotate down of 90 ° in order to present the film tips in the cleaning module.

Cutting system

The translation of the plate on its chariot allows carrying the film tips pinched by the rotating grips in front of the cutting device.

Automatic Splicer

Once the film cleaned and dried, and the tips repositioned horizontally, the plate will make a last translation towards the automatic splicer where a new splice will be made on the two faces of the film and the perforation of the adhesive tape

architecture of the Cleaner Module (CM)

The Cleaner module has 3 working positions: splice removal, cleaning and drying. These positions are on the upper part of a compartment of the CM. The system is on chassis worn by 4 wheels. The cleaner module can be easily and precisely slot in the TM.

splice removal

The first position enables splice removal. Once the film tips are rotated downward, a movement of the plate allows putting the film tips between the solvent injection pipes. At the end of the cycle a backward move of the plate allow the tips to be rotated upward again. The solvent pipes are in a profiled corridor that allows the recycling of solvent after filtering and elimination of pieces of adhesive tape.

cleaning

In this second position, the cleaning is performed in immersion. Film tips may be gently brushed.

The solvent is kept in a tank. An ultra sound transmitter may help to clean the tips. Contacts of solvent with the atmosphere will be limited by a small aperture during process and closed during other phases. A vacuum system may be added to remove local solvent vapours

film drying

The third position is a drying position. Film tips are positioned the same way as splice removal and cleaning. Drying is made with warm air at a temperature compatible with film base.

Visual control system

An operator makes the visual control. But once the new splice is done the film is moved in front of control system and stops. The operator can control the quality of the splice.

The command control system

The control system will be an industrial PC with a specific microprocessor completed with inputs and outputs cards (analogue to digital converters and drivers) interfaced with sensors on the machine. The control system will be integrated in the TM chassis

The man machine interface will be made on screen and will give information to the operator, about the automatic, semi automatic, manual or maintenance mode of the device.

Drawings

NOTE: drawings are issued from BERTIN's feasibility study.

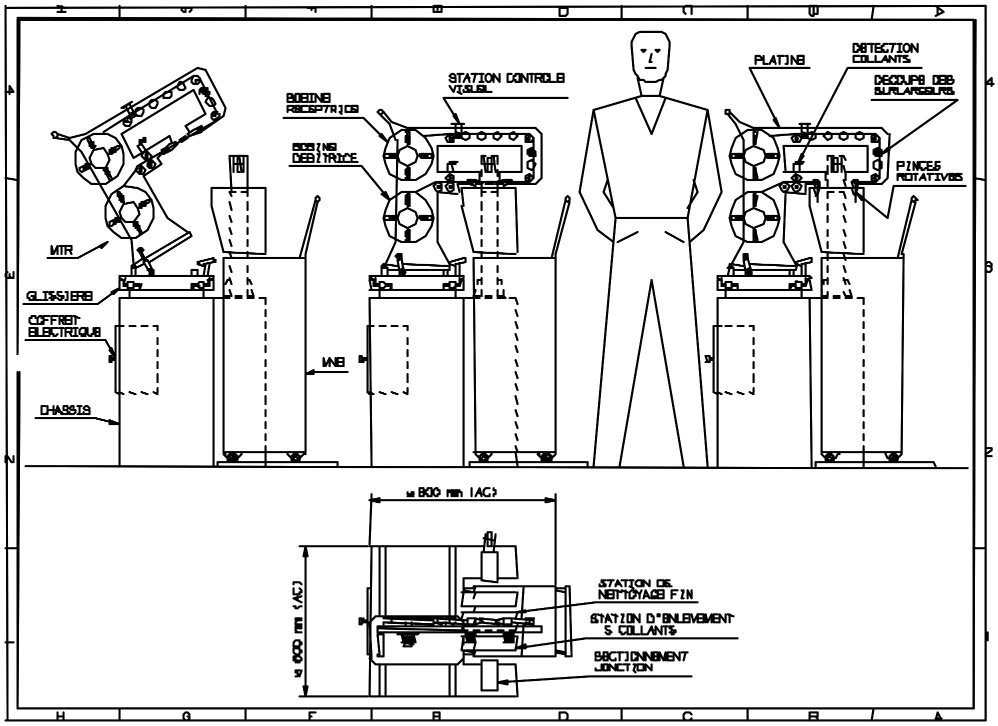

Figure 9.1: FT1-Overview of the future system with the transport module (named here MTR), the cleaning module (named here MNE) and the mechanisms of translation of the different steps of the process.

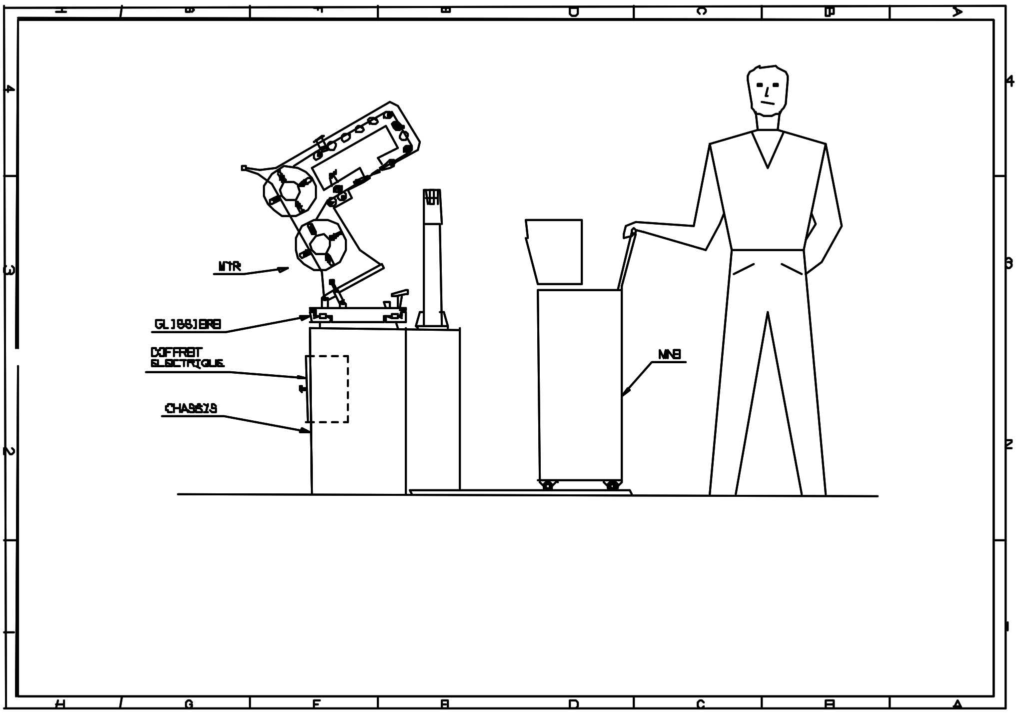

Figure 9.2: FT1-Accessibility of the transport and the removable cleaning module