2.5 Recording Channel Electronics Technology

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2.5.1 Introduction

This section outlines possible recording channel developments needed to meet the Roadmap specifications. Under the umbrella of "recording channel architecture," we will include any item in the signal path between the heads and the host interface. In this context, the greatest opportunities to advance transfer rate and capacity of future tape systems exist in the signal processing and coding algorithms that shape and decode the analog signal received from the tape. Linear tape recording technology has historically leveraged heavily from HDD development. The technology has benefited from advances in disk head technology, media technology, and signal processing algorithms [1]. However, as will be shown in Section 2.5.3, below, tape signals suffer from substantial impairments not present in disk signals.

The remainder of this section is organized as follows. We begin with a brief discussion of a general recording architecture, Section 2.5.2, including data and servo channels. We then list a set of noise sources that impair the tape read signal in Section 2.5.3. The next section, 2.5.4, briefly presents a set of technologies that enable the data channel electronics to achieve the objectives outlined in the Roadmap. Finally, we list possible university research areas that could help advance the technology in Section 2.5.5.

2.5.2 Data and Servo Channel Electronics Overview

The function of the data channels is to format and filter the user data so that it can be written to and read back from tape with maximum reliability, whereas the task of the servo channels is to extract servo information from the servo patterns written on servo bands. The signal processing blocks of the data and servo channels are shown in Figure 1. The thick connecting lines and double rectangles in the figure indicate that the function exists as many times as there are parallel data channels. Note that the two servo channels receive servo signals that are read from the dedicated servo bands and provide servo information during both data write and read processes.

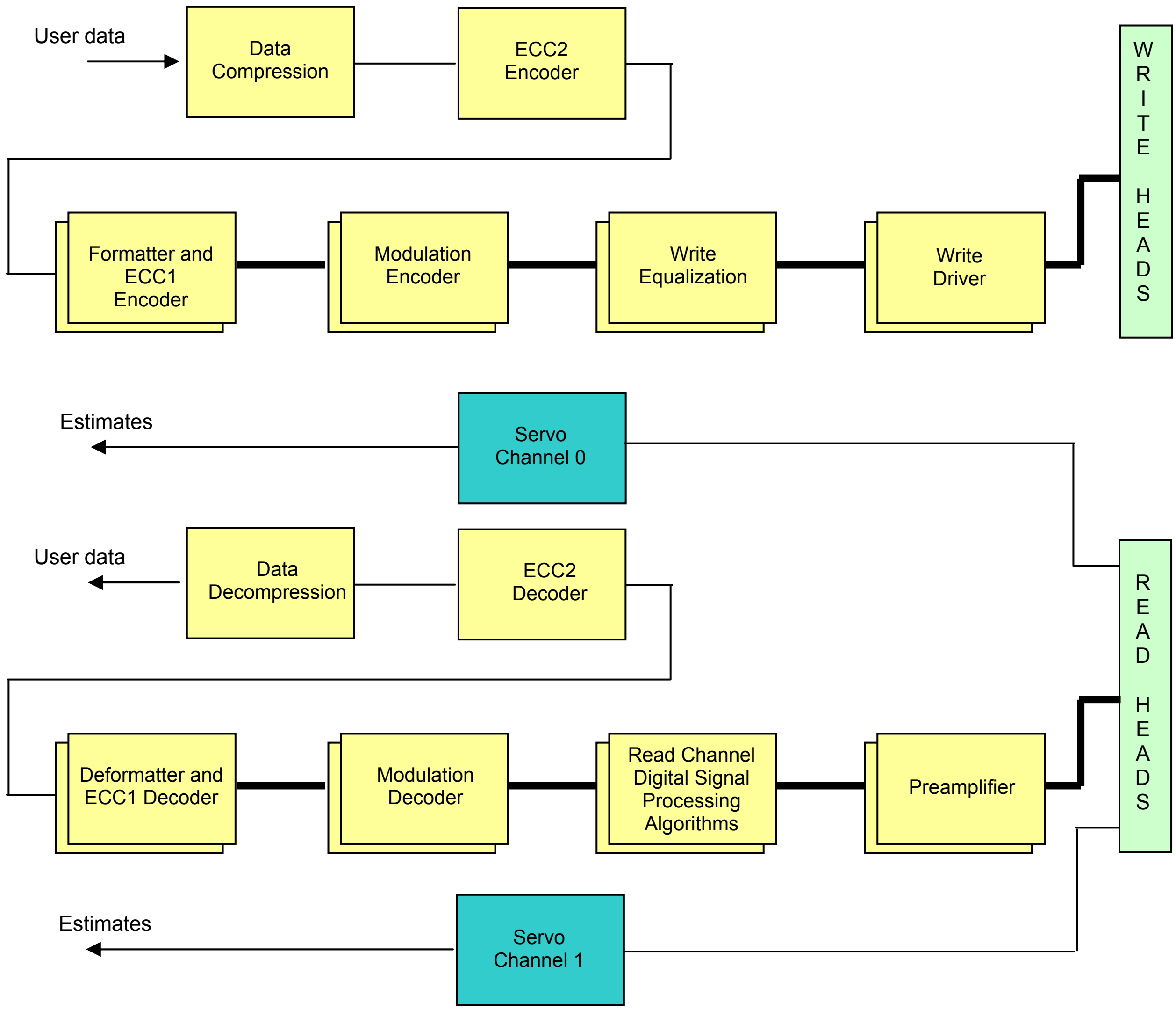


Figure 1: Overview of Data and Servo Channels.

Overview of Data Channel Functions

In the write channel, signal processing starts by compressing the data. With today's compression algorithms, a compression of 2:1 or 3:1 can be achieved. After compression, the data is divided into fixed-size blocks. For each fixed-size block, ECC (error correction code) symbols are computed.

In contrast to HDDs, tape drives usually have two levels of ECC. One level consists of adding redundancy to a string of bytes. This ECC type will be referred to as C1 (or ECC1). The second level consists of computing entire blocks of redundant information over many blocks of data. This ECC type will be referred to as C2. The purpose of C2 is to recover blocks that have been entirely destroyed, for example, because of malfunctioning readers or writers, loss of data synchronization, temporary debris accumulation under a head element, or large defects/scratches on the tape. On tape, a track (i.e., a data stream written by a single channel of the head) consists of concatenated blocks. The blocks that compose a type-C2 ECC (or

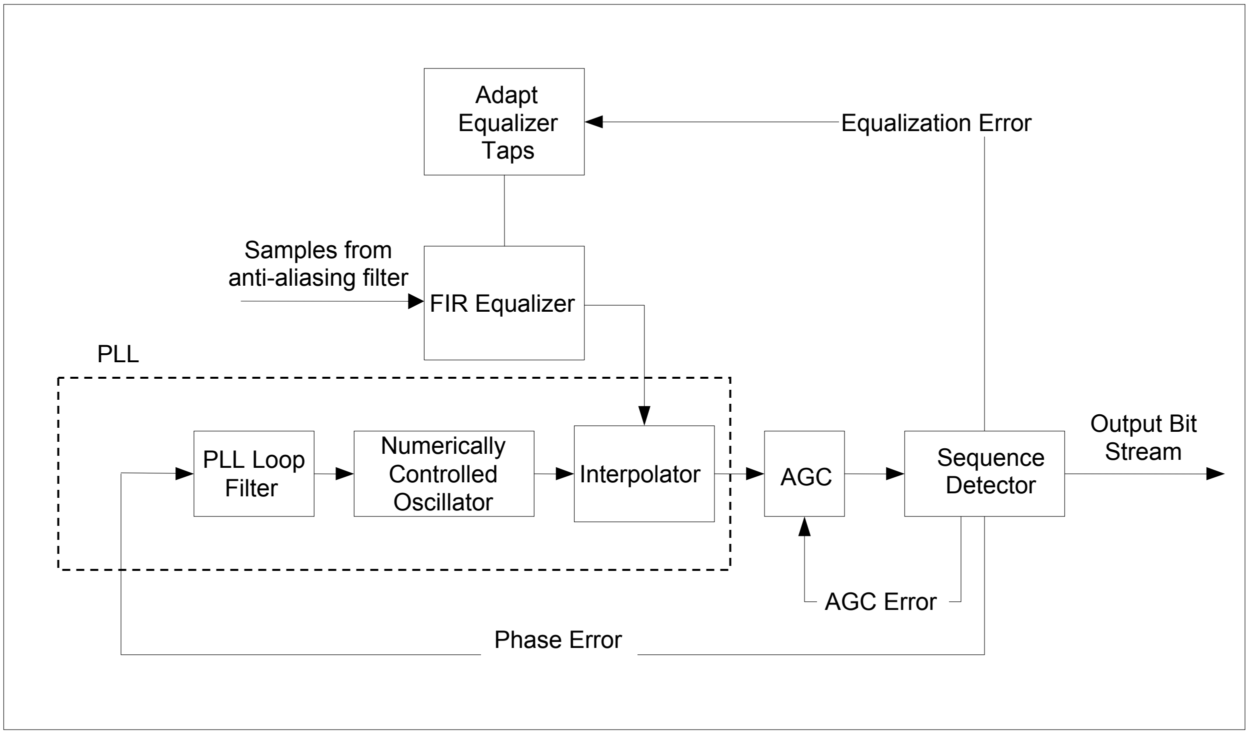
ECC2) codeword are multiplexed across all tracks so that if entire tracks of data are lost because of an inoperative head element, they can be recovered by ECC2.

After ECC symbols have been added, the bit stream is presented to the modulation encoder. The latter processes the data stream to eliminate any undesirable data patterns [2, 3, 4]. For example, a long string of zeros or ones is undesirable because the synchronization and sampling circuitry on readback need transition-switching data in order to continuously compute non-zero error signals that are used by the feedback loops. In recent LTO drives, a rate-32/33 code is used that satisfies various run-length constraints (to facilitate timing recovery and to limit the path-memory length of the sequence detector) [5].

The final signal processing block on the write side is the write equalization circuit. It inserts pulses of extremely high frequency into the write stream to prevent low-frequency components in the readback signal from saturating the sensitive magnetic recording read sensor. Another consequence of write equalization is that the readback signal has a shape that more closely matches the shape expected by the detector, thus reducing the extent of read equalization required. In addition, it has the advantage of providing nearly lossless pulse slimming, a significant improvement over the noise boost associated with readback slimming of pulses. Note that eventually write equalization may be abandoned as thinner tapes, which naturally deliver slimmer pulses, lessen the need for it and higher clock frequencies increase implementation difficulty. However, the tradeoff between implementation complexity and gains associated with write equalization will have to be evaluated on a case-by-case basis for every future tape system design.

The readers detect the magnetization pattern on the tape and create a readback waveform. This read signal is usually in the range of several hundreds of microvolts and needs to be preamplified. The next block, i.e., the Anti-Aliasing Filter and A/D (analog-to-digital) Converter (ADC), samples the signal from the preamplifier. From here on, all processing is done digitally.

The block labeled “Read Channel Digital Signal Processing Algorithms” contains circuits for timing recovery, equalization, and sequence detection. To implement this block, various architectures exist. Figure 2 shows an example architecture that illustrates the typical digital signal processing algorithms residing in today’s tape read channels.



FIR = finite impulse response AGC = automatic gain control PLL = phase-locked loop

Figure 2: Example Read Channel.

The major sub blocks in the read channel are briefly described below:

1. **A phase-locked loop (PLL**) for "timing recovery". The preamplified signal is filtered by an anti-aliasing filter prior to A/D conversion. The anti-aliasing filter removes the out-of-band component of the input signal prior to sampling with the ADC. The samples at the ADC output occur at an oscillator frequency that is usually higher than the detector symbol rate. The detector symbol rate is the frequency of the bits output from the sequence detector. The amount of oversampling is highly system dependent and varies from just enough to account for tape-speed variation (5-15%) to 3 or 4 times the detector frequency. As there are many ADCs in the system and because an ADC typically is an expensive, slow, and power-hungry component, it is desirable to keep the sampling frequency as close to the detector frequency as possible. An alternative solution may be to include fewer, but higher-speed ADCs multiplexed over multiple analog inputs.

The function of the PLL is to downsample the preamplifier samples and to carefully place the resulting samples at expected points for the detector. The incoming sample rate and the detector sample rate generally are not integer multiples of each other. The PLL does not merely choose a subset of the incoming samples to present to the detector, it identifies two or more samples between which a desired sampling point is located and computes the amplitude of the sample at the desired point by interpolating between the incoming samples.

The PLL is implemented using a PLL loop filter, a numerically controlled oscillator, an interpolator, and a phase-error detector. It functions by providing the interpolator with inputs so that it can compute an interpolated value between preamplifier samples based on feedback from the sequence detector.

2. **An equalizer** that filters the signal so that the signal received from the tape better matches the expected waveform in the detector. Usually this involves attenuating low-frequency components and amplifying high-frequency components of the incoming signals as well as phase compensation to redistribute frequency domain energy to make write equalized time domain pulses symmetric and in conformance with the intended partial response target. The equalizer often consists of an adaptive finite impulse response (FIR) filter. The taps of the filter adjust according to an error signal provided by the sequence detector using a least mean square (LMS) adaptation algorithm.

3. **A sequence detector,** such as a partial-response maximum-likelihood (PRML) detector, an extended PRML (EPRML) detector, or a noise predictive maximum likelihood (NPML) detector, that converts the interpolated samples computed by the interpolator into a string of 1s and 0s.

4. **An automatic gain control (AGC).** The sequence detector reads the amplitude of a signal to be able to match it to a list of target waveforms. Large variations in amplitude are devastating to the detector. Therefore an AGC circuit that modifies the incoming signal to match expected amplitudes at the detector is often inserted either in or ahead of the timing-recovery loop.

Because the phase-locked loop, equalizer, interpolator, AGC, and detector are coupled together in a control loop, it is necessary to monitor their effects on each other. The effects of each of these components individually have been intensively investigated in the past, and there are many papers in the literature that focus on equalization, phase locked loops, detection, or AGC. However, it is difficult to determine the exact benefit of each component without taking a more holistic view and simultaneously considering the ramifications on the other components of the loop. Alternative read channel architectures could also be used to minimize interactions between various control loops.

Overview of Servo Channel Functions

In a tape drive, typically two servo readers are part of the read heads, as indicated in Figure 1, above. To read servo information, the servo readers are positioned on top of pre-written servo patterns, which are found in servo bands straddling a data band. Servo information is obtained by a technique known as timing based servo (TBS) [6]. In TBS systems, the recorded servo patterns consist of transitions with two different azimuthal slopes. Tape velocity and head lateral position estimates are derived from the relative timing of dibit pulses generated by a servo reader reading the servo patterns. Furthermore, TBS patterns allow the encoding of additional longitudinal position (LPOS) information without affecting the generation of the transversal position error signal (PES). This is obtained by pulse position modulation, i.e., by shifting transitions from their nominal pattern position. The servo channels process the readback servo waveforms, and provide the above mentioned essential servo parameter estimates, i.e., tape velocity, lateral position of the head with respect to the servo bands, and longitudinal position of the head along the tape. These servo parameters are used to generate control signals for the reel-to-reel and track-following servomechanisms, so that the drive can be operated to write and read data very accurately at specified tape locations and at a predefined linear bit density.

A common approach to obtain the servo parameter estimates is by detecting the locations of the servo waveform peaks resulting from the magnetic transitions of the servo pattern bursts, and measuring the respective distances between the peaks. This approach is quite sensitive to noise and other disturbances. Therefore, more advanced techniques were introduced in a synchronous servo channel architecture [7], which was designed aiming at a near-optimal performance that facilitated assessing the limits of TBS for future ultra-high areal bit-density scaling and multi-Terabyte tape cartridge capacities. The block diagram of such a near-optimal synchronous servo channel architecture is shown in Figure 3, below.

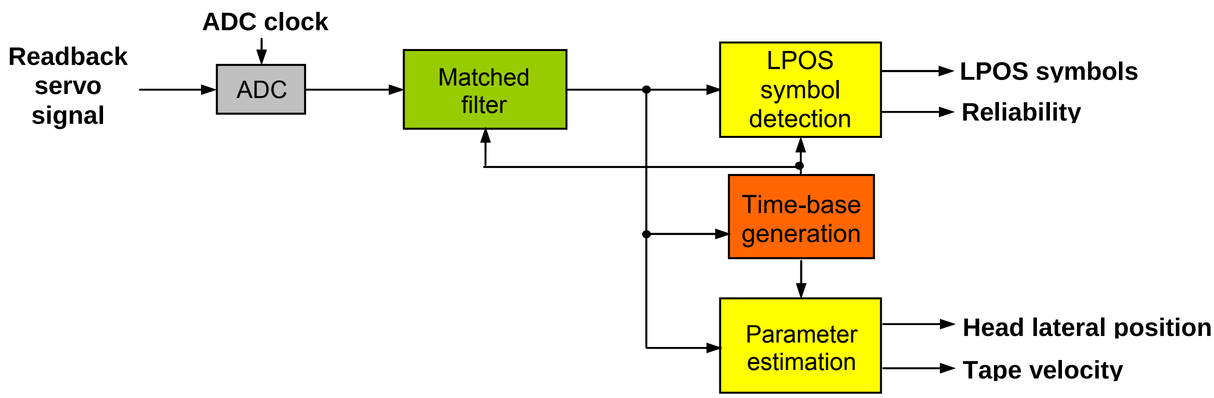


Figure 3: Example Servo Channel.

Two main concepts are incorporated in a synchronous servo channel. The first concept involves the generation of a fixed number of sampling points for each servo frame, thereby enabling a synchronous operation with respect to the distance that the tape has traveled across the tape head. This re-sampling process is done by computing the desired signal samples from the digitized servo waveform received from the servo reader using interpolation. As a result, the estimation of the servo parameters becomes independent of the tape velocity. The second concept applied in a synchronous servo channel is a digital matched-filter approach to minimize the measurement noise in the parameter estimates. The re-sampled servo readback waveform is correlated with a reference waveform, which can be determined, e.g., by *a priori* knowledge about the readback servo signal characteristics. The locations of the peaks of the resulting matched filter output signal are used to estimate the servo parameters. The matched filtering approach is optimal in the presence of additive white Gaussian noise, and considerably increases the system robustness and measurement accuracy in the presence of media noise and other disturbances.

It was mentioned earlier that the servo patterns are characterized by an azimuthal angle α, which in LTO is equal to 6 degrees. It turns out that increasing the angle α leads to improved PES performance. However, the gains diminish for larger values of α, as a consequence of increasing dibit energy loss. A further interesting observation is that PES performance improves, in spite of increasing dibit energy loss, also as the peak-to-peak distance of dibit pulses is reduced below the LTO value of t = 2.1±0.4 μm, and reaches its optimum at around 1 μm. To investigate the impact of servo pattern parameters on PES performance, the PES standard deviation σ_{PES} has been evaluated by simulations using a synchronous servo channel that operates in open loop in the absence of lateral tape motion (LTM) [8]. The value of σ_{PES} as a function of t and α is illustrated in Figure 4, for signal-to-transition-noise ratio of 33 dB, and an electronic noise level that is 30 dB below the servo signal level obtained for α = 6° and t = 2.1 μm. Experimental results of PES standard deviation for a servo pattern characterized by for α = 18° and t = 1.26 μm have also been shown in [8]. In particular, a closed-loop track-following performance of less than 14 nm has been demonstrated.

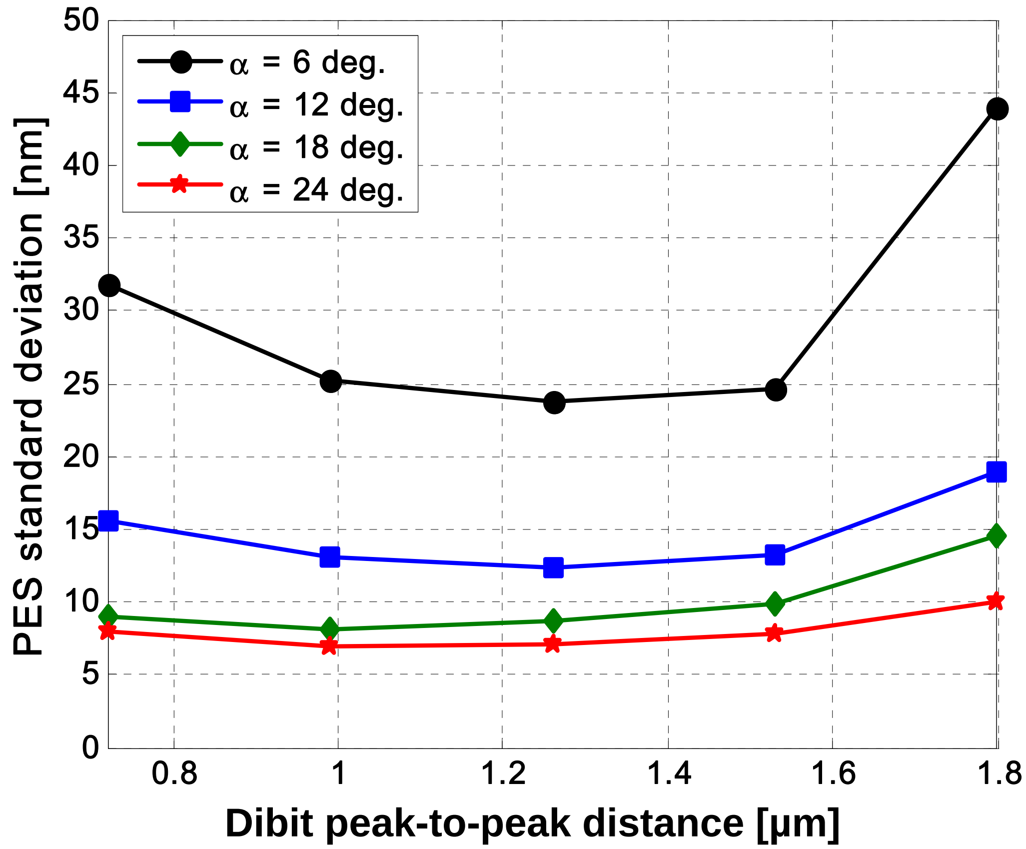


Figure 4: Achievable PES Performance of a Synchronous Servo Channel.

2.5.3 Noise Sources

This section lists the noise sources for tape signals [9, 10, 11, 12, 13, 14]. A "noise source" is any contribution that causes the incoming signal to be reduced in quality. For tape systems, noise sources can be categorized as follows:

- Time-varying (lasting from tens up to thousands of bits) distortion of the signal. A distortion can be linear and characterized by a shift in the frequency spectrum of the signal, or it can be non-linear. Non-linear signal distortions can be due to MR head saturation or the non-linearity of the head response. Interactions between previous transitions' fields and those being written can also create write-induced non-linearities, as can flux rise time limitations in the head, or other rise time limitations in the head/preamp/flex system.

- Loss of signal. This can be due to inoperative head elements or to locations on the medium where magnetic material is missing or some other type of defect occurs. The absence of a signal may be due to reader or writer malfunction, and may be permanent or temporary in nature (an example of the latter may be transient debris temporarily clogging an individual head element).

- Transition jitter. This non-stationary data-dependent noise source is associated with high-frequency issues during write equalization, as well as particle dispersion or exchange coupling that can result in low frequency noise correlation detrimental to sequence detection.

- Crosstalk noise. This is noise due to crosstalk between a write transducer and another transducer (either reader or writer).
- Frequency shift. Because of variations in tape speed, the incoming signal frequency may be shifted up or down.
- Reading of old data that was not adequately overwritten because of a malfunctioning writer, insufficient write current, localized coercivity variations in the media, or increased head/media separation as a result of debris buildup or staining.
- Signal amplitude reductions due to reading off-azimuth data and variations in head-to-medium spacing.

The specific sources of noise are listed in Table 1. The ranking in Table 1 reflects the importance of the corresponding noise source. For example, ranking 1 implies that this noise source or component has the largest impact on performance.

Many, and especially the most severe, of the noise sources do not exist on disk drives for several major reasons:

1. Disk drives benefit from carefully engineered sputtered thin-film media with extremely smooth surfaces. Tape drives rely on mass-produced media that are several miles long, less consistent in their magnetics down the length of the tape, with increased environmental sensitivity, and more prone to damage or physical changes with continued use. Tape is made in huge rolls, called "jumbos," which have multiple miles of tape. The jumbos are sliced into pancakes and the tape on the pancakes is then spooled off the pancakes, servo mastered, and wound into cartridges. One pancake will typically make many cartridges. The tape on that pancake is typically servo mastered continuously.

2. Disk drives can map out defective areas: they do not have to read through defects.

3. Disk drives do not have interchange. Each disk drive has to adjust to the combination of each of its disk surfaces with only a single reader and a single writer. Tapes written on one drive must be readable on all other drives.

4. Disk drives do not have the speed variations that are common for tape drives.

5. Disk drives only have a single reader and a single writer. They do not read and write at the same time and suffer neither from writer to reader crosstalk (also known as feedthrough) nor from writer to writer crosstalk.

We point out the major differences between disk and tape signals because much research has been done on signal processing for disk signals. Tape signals can certainly benefit from this research, but it is important to recognize the major differences between tape and disk signals.

Table 1: List of Noise Sources for Tape.

Noise Source	Description	Importance Ranking	An Issue for HDD
Media noise	Non-uniform size, orientation and distribution of magnetic particles in the media	1	Yes, much less
Head/tape separation increase	Large average head/tape separation	2	Yes, much less
Head/tape separation variation	Dynamic variations in head/tape spacing	2	Yes, much less
Media coating thickness variations	Impact to disk is radically different. Particulate media variation is much worse than thickness variation seen in sputtered disk media	3	Yes, limited
Head stain	Buildup on head that attenuates signal and causes head/tape separation; largely due to particulate media	3	Yes, much less
Pole tip recession	Wearing away of magnetic structure, similar in effect to head stain	3	Yes, much less
Thermal asperities	Head to tape contact causing the MR head to experience a DC shift due to heating or cooling; somewhat less in disk because of head flight	3	Yes, somewhat less
Short dropouts	Small areas on the media where the coating does not function as intended give rise to short dropouts in signal strength. Errors due to short dropouts are correctable by C1 ECC.	4	Yes
Long dropouts	Large areas on the media where the coating does not function as intended give rise to long dropouts in signal strength. Errors due to long dropouts require correction by C2 ECC and may be caused by media defects, debris, or scratches, etc.	4	No
Writer to writer crosstalk	Crosstalk from one writer into adjacent writers, or between write signals on the flex circuit	4	No
Electrical noise	Electrical noise in preamp, printed circuit board assembly, cables, and connectors	5	Yes
Thermal noise	Noise introduced by the read head	5	Yes
Writer to reader crosstalk	Crosstalk during read while write	5	No
Head clogs	Head becomes clogged and produces severely attenuated output	6	No
Barkhausen noise	Head noise arising from fluctuations of magnetic domain walls in the magneto-resistive sensor	6	Yes
Transition jitter	Non-stationary data-dependent noise associated with high-frequency issues during write equalization	7	No
Adjacent track crosstalk	Crosstalk due to reader approaching adjacent tracks; likely to increase with higher track density	7	Yes
Azimuth loss	Skew between head and tape causes azimuth error	7	No
Tape speed variations	Tape speed variation stresses timing recovery	8	No
Overwrite noise	Reading of old data that was not overwritten well due to write process variations and/or separation	9	Yes
Read head nonlinearities	MR heads saturate and/or operate in the nonlinear region	10	Yes
Azimuth loss due to media interchange	Interchange of media between drives with heads at different angles	11	No
Transition noise	Zigzag erasure does not exist in particulate media, but will be seen on conversion to sputtered or metal evaporated (ME) media; ranking will then move to 4 or 5	12	Yes

When discussing noise it is customary to use the signal-to-noise ratio (SNR) as a measure of the severity of the noise. The problem hereby is that this number can only be used to describe very few of the above noise sources for tape. Some noise sources (such as inoperative head elements or non-linearity of MR head elements) can only be described by non-linear methods in simulation. Other sources, such as media defects, are described by specifying typical defect characteristics and their statistical distributions. Each noise source has its unique characteristics and must be considered individually.

Many of the noise sources for tape have been simulated and researched on a standalone basis. In practice, however, the signal processing algorithms in the read channel experience all noise sources in combination. To realistically understand how the signal processing algorithms will respond to the combination of all noise sources, one must model the effect of each noise source, specify a statistical frequency of occurrence for each, and present the resulting signal to the read channel algorithms.

Finally, because many of the noise sources occur infrequently, the use of software simulation to predict the associated error rates has proved to be inadequate, even on the fastest of today's computers. To evaluate the effectiveness of signal processing algorithms, the noise generation needs to be done in hardware or field programmable gate arrays (FPGAs). Fortunately, today's FPGAs are getting easier and easier to program and contain a host of useful features that are specific to digital signal processing.

Many of the noise sources in tape cannot be addressed by C1 ECC. Therefore, C2 ECC absorbs a large fraction of the noise sources and is thus indispensable to a proper operation of a tape drive.

2.5.4 Future Improvements in Signal Processing Algorithms and Coding Schemes

Many signal processing and coding technologies are available today to facilitate achieving higher areal densities despite the migration to ever lower SNRs and the increasing influence of noise sources. Tape technology is fortunate to be able to benefit from the extensive research activity in communications and HDDs. This section is devoted to listing several known signal processing and coding techniques that have the potential of improving the ECC, formatting overhead and the error-rate performance of tape drives.

Advanced Sequence Detection Techniques

Partial response (PR) techniques for storage channels shape the overall channel to some prescribed intersymbol-interference pattern for which simple sequence detection methods are known. In its simplest form, the sequence detector for PR channels is the maximum-likelihood sequence detector for a finite-state channel corrupted by additive white Gaussian noise. This sequence detector is usually implemented by the Viterbi algorithm. The combination of PR equalization techniques with maximum-likelihood sequence detection (MLSD) is known in the storage industry as PRML detection [15].

The discrete-time representation of the PR-shaped linear recording model in the absence of noise is given by the convolution of the binary modulation-encoded non-return-to-zero (NRZ) signal at the input of the recording channel and a short sequence Φ that depends on the type of partial-response shaping used. The D-transform of Φ, where D represents the delay T corresponding to the duration of a modulation-encoded bit, is the PR polynomial f(D) which is selected such that its spectral characteristics match the magnetic recording channel.

The PR polynomial that was first used to introduce the PRML technology into HDD products in 1990 was $f(D)=1-D^2=(1-D)(1+D)$, which is also known as partial-response class-4 (PR4) polynomial. Note that the PR4 polynomial has a spectral null at zero and at the Nyquist frequency 1/(2T), matching the nulls or near nulls of the physical recording channel. As the degree of the PR4 polynomial is two, the sequence detector for binary recording signals can be implemented with four states. In this particular case, however, two 2-state sequence detectors whose outputs are even-odd interleaved can be employed to implement the Viterbi algorithm.

In tape drives, the transition to a PR read channel was facilitated by the adoption of a new modulation code by the LTO Consortium. In particular, LTO-2 abandoned the rate-2/3 RLL(d=1,k=7) legacy modulation code of LTO-1 and adopted a rate-16/17 code that satisfies various run-length constraints (to facilitate timing recovery and limit the path-memory length of the sequence detector) and guarantees at least one isolated peak per 17-bit codeword. The adoption of this code paved the way for the introduction of the PRML technology into LTO tape drive systems. Currently, read channels in tape drives are usually based on the PRML architecture.

The simplest characterization of magnetic recording signals is based on the ratio known as the normalized linear density PW50/T, where PW50 is the pulse width at 50% amplitude of the transition response of the recording channel. PR4 polynomials are a good design choice for recording channels with PW50/T ≈ 2. However, with increasing linear density and PW50/T, higher-order PR polynomials provide a better match to the spectral characteristics of the recording channel. The PR polynomial $f(D)=(1-D)(1+D)^2$ is known as the extended PR4 (EPR4) polynomial. LTO tape drives have used EPR4 shaping at normalized linear densities higher than 2. In general, the family of PR polynomials $f(D)=(1-D)(1+D)^K$, $K\geq1$, are suitable for magnetic recording because their spectral characteristics match the spectral characteristics of the physical recording channel fairly well. It is worth mentioning that the PR polynomial for K=3, $f(D)=(1-D)(1+D)^3$, is known as the E2PR4 polynomial. Figure 5 illustrates the frequency response of a Lorentzian channel with PW50/T= 3.0, the frequency response of a 5x write-equalized Lorentzian channel with PW50/T= 3.0, and the frequency responses of various PR shaping polynomials. Note that a write-equalizer inserts high frequency pulses into the T-spaced data stream, where T corresponds to the recorded bit length. In the case of 5X write equalization, the width of the inserted pulses is T/5.

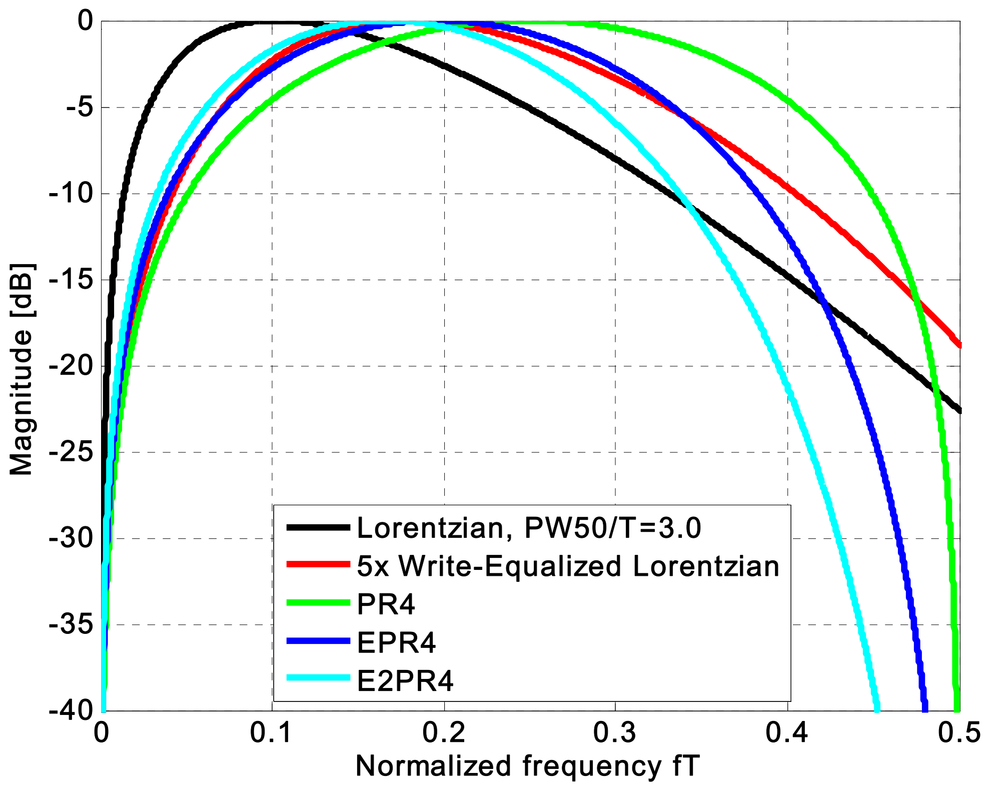


Figure 5: Frequency Responses of Lorentzian Channels and Various PR Polynomials

Clearly, the dibit responses of a tape drive system do not exactly match the Lorentzian dibit response. Moreover, they are time varying because of variations in the medium characteristics and the head-medium interface. Different sections of a medium will exhibit wider or narrower pulses, depending on the local characteristics of the medium. Each tape drive exhibits different head and mechanical characteristics, which further increase the amount of variation. Finally, head recession also contributes to the time-varying nature of the dibit response. It is the function of the equalizer to adaptively transform the signal read from the recording medium into a shape that the PRML sequence detector expects. In doing so, the equalizer must often amplify the high-frequency components of the signal being read from the tape, leading to noise enhancement. In addition, transition jitter from media noise creates a data-dependent correlated noise component that prevents achieving the full benefit of an ideal maximum likelihood sequence detector operating on perfectly equalized samples in the presence of white noise. As a result, the PRML detector experiences colored noise and is forced to function sub-optimally unless the noise can be re-whitened.

In the absence of noise enhancement by the linear PR equalizer and noise correlation at the detector input, the PRML sequence detector performs MLSD. But there is an obvious loss of optimality associated with linear PR equalization as the operating point moves to higher normalized linear densities. Clearly, a very close match between the desired target polynomial and the physical recording channel will guarantee that this loss is minimal. An effective way to achieve near optimal performance independent of the linear recording density and the noise conditions is by means of noise prediction. In theory, an infinitely long predictor is required to minimize the power of the noise sequence at the output of the PR linear equalizer. However, the noise power at the output of the PR linear equalizer can be significantly reduced (almost minimized) by using a sufficiently long whitening filter (1+p(D)), where $p(D)=p_1D+\ldots+p_LD^L$ is the transfer function of a predictor of finite order L. In general, shaping polynomials of the form $g(D)=(1-D)(1+p(D))$ with a spectral null at zero frequency and $g(D)=(1-D^2)(1+p(D))$ with spectral nulls at both zero frequency and the Nyquist frequency render the noise at the input of the sequence detector approximately white. These classes of generalized PR polynomials, which play an important role in practical applications when combined with sequence detection, give rise to NPML systems [1, 16, 17, 18, 19]. NPML and variants thereof have been adopted by the HDD industry since 2000. A read channel architecture based on NPML detection and noise-predictive parity-based post-processing [20, 21] for stationary or data dependent noise has been widely used in HDDs. Clearly, the NPML technology, which was originally developed for HDDs, can also be used in tape drives. It provides a method for accommodating tape noise sources by the PRML detector.

Specifically, the branch metric computation in a conventional Viterbi algorithm is modified to include the effects of known tape noise sources. This general approach can be used to integrate some of the noise sources listed above into the detector. For example, the concept of noise prediction for stationary Gaussian noise sources can be naturally extended to the case in which the noise characteristics strongly depend on the local data patterns [1, 22, 23]. In this case, both predictor coefficients and prediction error depend on the local data pattern and the resulting structure is known as data-dependent NPML detector. Information theoretic limits of binary-input intersymbol interference channels with signal-dependent media noise have been studied in [24].

ECC and Formatting Overhead

To make efficient use of the magnetic recording channel and thus achieve reliable readback operation of the user data, the bit stream written onto the magnetic medium includes overhead associated with error correction redundancy [25, 26, 27, 28], and synchronization patterns. The specification of this overhead data is referred to as formatting or data format. The goal of an efficient format is to introduce as little overhead as possible, while ensuring proper operation of the data acquisition and timing loops and meeting the 1×10^{-20} corrected byte-error rate requirement of tape recording systems. Clearly, improvements in format efficiency (i.e., reductions in overhead) directly lead to higher cartridge capacity.

The format efficiency of recent LTO tape drives is about 78.9%. The 21.1% overhead in LTO-5 can be broken down into about 15.2% for ECC, 2.9% for modulation coding, and about 3.0% for synchronization patterns and data headers. There are several ways to improve format efficiency: One approach is based on using longer ECC codes of higher rates (in particular longer C2 codes), resulting in about a five percentage point gain in format efficiency without sacrificing error-correction performance. Note that currently the C2 Reed-Solomon (RS) code in LTO has a dimension of 84 symbols and a codeword length of 96 symbols, where a symbol is 8 bits long. Figure 6 shows that C2 codes with higher rates can be designed to deliver an even more demanding 1×10^{-22} byte-error-rate requirement at a high byte erasure rate of 1×10^{-3} (corresponding to a low SNR operating point) at the output of the C1 decoder. In the case of a C2 code with a dimension of 240 symbols and codeword length of 256, the gain in code rate over a widely used *[64,54,11]* Reed-Solomon C2 code is about 11% providing a significant increase in format efficiency. An even higher efficiency is achieved by another approach that relies on a coding scheme known as reverse concatenation (RC) architecture. In fact, reverse concatenation in conjunction with long C2 codes can reduce the overhead to about 13%.

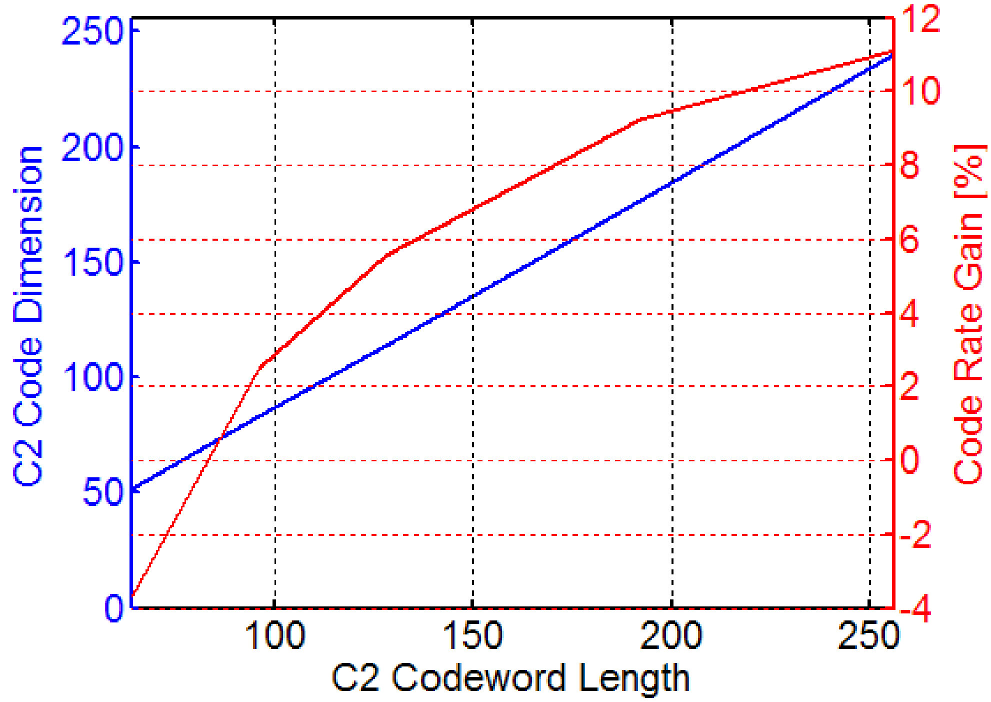


Figure 6: Dimension and Codeword Length of Long C2 Reed-Solomon Codes for Achieving 1×10^{-22} Byte-Error Rate with an Input Byte-Erasure Rate of 1×10^{-3}

Reverse Concatenation

Until a few years ago, forward concatenation of outer RS codes and inner modulation codes was the standard coding scheme used in virtually all magnetic and optical data-storage systems.

In forward concatenation, a sufficiently small block size for the modulation code is selected to reduce or eliminate error propagation at the output of the modulation decoder. However, the requirements of a high-modulation code rate in conjunction with low error propagation usually result in loose code constraints. In this case, the modulation code rate is not very close to the Shannon capacity associated with the imposed code constraints. Moreover, in forward concatenation schemes, the passing of soft information from the detector to the outer ECC decoder is hampered by the modulation decoder located between the detector and the outer ECC decoder.

An alternative approach to forward concatenation is provided by reverse concatenation (RC) [29]. RC architectures reverse the order of concatenation of inner modulation coding and outer error-correction coding in data storage channels to enable the use of very-high-rate efficient modulation codes without suffering from error propagation [30]. Error propagation due to modulation decoding is not a problem in RC architectures because the modulation decoder corresponding to the long efficient very-high-rate modulation code follows the ECC decoder. There are various RC architectures depending on the combination of the particular ECC and RLL (run-length-limited) encoding algorithms used. A simple RC scheme employs an efficient data modulation encoder followed by a systematic ECC encoder and a parity modulation encoder for parity symbols. An alternative RC architecture is based on inserting ECC parity bits into a modulation-encoded data stream [31, 32]. In addition to increased format efficiency, the other important benefit of such an approach is that the detector or the inner parity decoder can readily provide soft information on modulation-encoded data bits to the ECC decoder. Although the general concept of RC was proposed in 1981, it took about a quarter of a century to turn RC into reality. Recently, commercial HDD products were first to implement RC. Currently, all HDD channels support RC, and all new HDD channels use only RC.

ECC codes used in single-channel HDDs are usually RS codes with 10-bit symbol size, whereas more powerful product RS codes with 8-bit symbol size are used in multi-channel tape drives. Therefore, RC schemes for tape will be different from RC schemes for HDDs. RC schemes for tape promise the same benefits of increased format efficiency and ease of exchanging soft information between the detector and the ECC decoder. The introduction of the RC architecture into tape is the prerequisite for implementing iterative decoding and detection in tape to improve the error-rate performance at the lower SNRs expected at increased areal densities. Finally, RC will only become reality in the tape industry if suitable multi-channel RC schemes are discovered for product ECC codes that satisfy desired modulation constraints and are very simple to implement.

Iterative Decoding and Detection Techniques

Iterative decoding and detection techniques are methods that process a set of samples or bits (called a block) more than once. Every processing step provides increasingly reliable information to other parts of the read channel, thus increasing the probability of successful decoding. Iterative decoding techniques are characterized by two fundamental features:

1. The use of “soft information” instead of “hard information”. Hard information is simply a true or false answer. For example a bit is a 0 or a 1. Soft information also describes how reliable the 0 or 1 answer is. In iterative decoding techniques, each subsystem in the read channel (equalizer, detector, or decoder) can be presented with soft information, rather than hard information. It has been shown that doing this, even in a very coarse fashion, greatly increases the probability of correct decoding. Clearly, the drawback to using soft information is implementation complexity: Instead of storing 1’s and 0’s, we must now represent each received bit by a value wider than one bit.

2. The processing of a data block more than once. There are almost infinitely many ways of iterating on data. Perhaps one of the most familiar methods is to iterate within an RS product code. We could let C1 decode and mark its uncorrectable locations. C2 now knows error locations and can therefore double the number of symbols it can correct. The additional information available here consists of the error locations. The process now iterates, giving C1 an additional turn to correct, then C2 again, and so on. Iterations can take place between the ECC and the detector. Numerous papers illustrating variations on this technique exist. However, for magnetic tape systems, this iteration process may pose quite an implementation hurdle. In a tape drive the read-channel signal processing algorithms may occupy one application specific integrated circuit (ASIC), while the error-correction code may be implemented in another ASIC. Furthermore, the read-channel ASIC is often designed by a company, other than the tape drive manufacturer, that is reluctant to customize the chip for a particular tape drive. Therefore, it is logistically difficult to tightly couple the ECC with the detector in an iterative loop.

The ECCs used with iterative decoding techniques often are low density parity check (LDPC) codes or simple random parity checks among the bits [33]. In the past, the design of error control codes focused on increasing the minimum distance between codewords. The focus in iterative decoding is not on achieving maximum distance between codewords, but rather on clever methods of passing soft information from one processing step to the next in the decoding process [34].

2.5.5 Future Research Directions

In this section we outline several areas of university research that could be helpful in achieving the Roadmap objectives.

Iterative Detection/Decoding Schemes with Embedded Timing Recovery

Goal

Develop signal-processing algorithms that enhance the ability of the timing-recovery loop to function at broadband SNRs as low as 10.5-11.5 dB and to read through media defects.

Background

Increasing linear densities, higher bit/PW50 ratios, and lower readback broadband SNRs are clearly the trends for the future in this INSIC International Magnetic Tape Storage Roadmap. Future tape media require operation of the read channel at SNRs that will be 5-6 dB lower than those in today's tape drives. Current timing-recovery architectures, however, will most likely not operate properly in these lower SNR regimes.

Even in EPR4 systems, PR4 timing recovery is often used because of its reduced complexity and higher slicer margins due to its three ideal sample values. EPR4 timing recovery requires less slimming and accompanying noise boost/SNR loss, but its five slicing levels reduce the SNR margin compared to that of PR4, posing a problem when used in low-SNR applications. The use of feedback from the sequence detector can improve timing-recovery decisions but introduces latencies that reduce the tracking performance to levels that can be unacceptable for tape drives, where tape speed can vary relatively quickly because of such mechanical issues as instantaneous speed variations (ISVs). Also, typical tape-speed control (in the few percent range) may provide large frequency steps between data written at different times, necessitating fast acquisition.

To address these conflicting constraints, research is needed into algorithms that achieve robust timing recovery at low SNRs, while still providing an acquisition and tracking performance similar to what we have today. One such class of algorithms could be developed by applying timing-recovery techniques that take advantage of and operate in conjunction with soft data decoding. Assuming that the tape readback signal is buffered before digital timing recovery is applied, reliable data estimates provided by the decoder can be used in a feedback fashion to iteratively resample the readback signal stream prior to performing data detection and decoding. The additional robustness provided by iterative detection/decoding schemes with embedded timing recovery needs to be investigated.

Study of Missing-Pulse Phenomena and Possible Remedies

Goal

Understand missing-pulse statistics, assess impact of missing pulses on read channel performance, and investigate remedies and dropout-tolerant ECC strategies.

Background

The readback signal in tape systems is subject to dropout effects that are also known as missing-pulse effects. The main causes of missing pulses are defects in the basefilm and imperfections in back-coat formulation that manifest themselves as asperity imprints. The missing-pulse phenomenon is well known in storage on tape media, and is said to occur if the signal amplitude drops below a certain threshold. For example, the LTO specification defines the occurrence of such disturbances as an amplitude drop below 30% of the average signal amplitude. Although missing pulses have not been a primary concern for read channel design in tape-recording systems so far, it appears that their effect can be highly detrimental as areal recording densities need to be increased in the future. In fact, the occurrence of missing pulses turns out to be highly dependent on the linear recording density and on the reader width. Projections indicate that in tape systems the raw symbol-error rate caused by missing pulses could degrade by as much as three orders of magnitude within the next 10 years [35].

Therefore, there is strong interest to understand the missing-pulse phenomenon better and to develop signal-processing and coding techniques to mitigate its effect on the drive performance. Research work in this area should focus on developing practical statistical models of missing-pulse phenomena and determining, by means of computer simulation as well as analytically, the impact of missing pulses on the read channel performance. A topic of particular interest is the determination of the impact of missing pulses on the error-rate performance as the linear density increases according to the tape Roadmap. Read-channel signal-processing techniques, possibly including the investigation of original ECC strategies, should also be developed to achieve an overall dropout-tolerant tape-drive system operation. Finally, another topic of significant interest is the loss of signal because of much-reduced track widths at future operating points.

Read Channel Design for Perpendicularly Oriented Media with Soft Underlayer

Goal

Understand the properties of tape recording based on perpendicularly oriented media with soft underlayer and develop the main building blocks needed in read channels for such media.

Background

Traditionally, tape storage has been based on the use of metal particulate (MP) media with longitudinal orientation of the magnetic particles. While this technology has served the tape-recording industry well over a rather long period of time, more recently, alternative solutions have been developed, such as those based on the use of BaFe particles, that allow significantly higher broadband SNRs, and hence higher linear recording densities to be achieved than the traditional MP designs. To be able to push the state of the art even further in future tape-recording systems, it is envisioned to adopt a recording technology similar to the perpendicular recording technology that has been successfully commercialized for a few years in HDD systems now. This technology has enabled a continuing increase in areal recording densities for HDDs. In perpendicular recording, the writing element creates a perpendicularly oriented magnetic flux that is propagated through both the magnetic medium and a magnetically soft underlayer placed underneath it. Such an approach could also be of interest for magnetic storage on tape.

It is therefore highly desirable to understand the potential of perpendicular recording with soft underlayer for tape systems. To this end, models must first be developed for the transfer and noise characteristics of this recording channel, e.g., using micromagnetic simulations. Based on the models developed, the main building blocks of a read channel for perpendicularly oriented media with soft underlayer can be designed and the error-rate performance of the overall recording channel investigated.

Investigation of Adjacent-Channel Cancellation Techniques to Facilitate the Use of Wider Readers

Goal

Develop read-channel signal-processing techniques to realize efficient cancellation of disturbances due to signal interference from adjacent tracks.

Background

According to the INSIC Roadmap, the areal recording density of tape systems will grow at an average rate of about 33% every year until 2022. The largest portion of this growth will be due to the increase in track density (expected to be 23.3% per year on average). This trend will therefore pose considerable challenges for future tape-drive technology, in particular for the design of the reader elements. Track-pitch values as low as about ½ micron will need to be achieved by 2022. It can therefore be expected that the combined effects of nonaccurate track writing and read-head positioning will result in adjacent-track interference within the data channels. Moreover, very narrow readers will need to be produced at low cost and with stringent manufacturing tolerances.

It is envisioned that by developing and applying appropriate read-channel signal-processing techniques, this particular set of problems could be alleviated to a significant extent. One technique that appears promising in this context is adjacent-channel cancellation. By canceling essentially those signal components that are due to data recorded on the tracks that are adjacent to the track to be read, it should be possible to boost the overall channel performance and/or facilitate the use of readers of larger width or with relaxed manufacturing tolerances. To be able to develop meaningful cancellation techniques and assess the performance gains they can achieve, realistic models of adjacent-channel interference need to be developed as well.

Investigation of Read-Channel Design to Mitigate Writer Crosstalk Resulting from Reduced Head Pitch

Goal

Understand crosstalk effects that are due to the close packing of write elements and their impact on the read-channel performance. Develop suitable strategies and techniques to mitigate such deleterious effects.

Background

Achieving extremely high areal recording densities in future tape systems (in excess of 50 Gbit/in^2 by the 2022 time frame) poses stringent requirements in terms of track density and track pitch, as mentioned above. In particular, the need to position writing elements very close to each other will unavoidably lead to crosstalk coupling phenomena between writers. Therefore, if such disturbances cannot be mitigated, then writers can induce undesired signal components into neighboring tracks, thereby reducing SNR margins and compromising detection reliability on those tracks.

It is therefore necessary to develop reliable models that capture the effects of crosstalk coupling between writers at very low head pitch. These models will, on one hand, be used to evaluate the expected degradation in channel performance due to multitrack writing. On the other hand, they will be employed in the design of efficient strategies and signal-processing techniques to be incorporated into the read channel to mitigate writer crosstalk effects.

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