Note to Editor – this has been modified from the revision presented by Norm Swenson on the TP2 call on Jan 6, 2005. The changes are purely editorial, in the first paragraph only, and anticipate that this document will go into an Annex.

I do not know what the requirements are for formatting, headers, credits, etc., so hopefully the editor is willing to make any further necessary modifications.

Technically, this writing must follow any changes that the committee makes to the content of the actual algorithm.

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Proposed TP-2 Test Methodology

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Description of the processing algorithm
for
Transmitter optical waveform and dispersion penalty test
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This annex outlines a TP-2 test methodology for measuring a penalty for purposes of determining compliance with 10GBASE-LRM specifications. An upper limit on penalty thus measured will be compared against a limit specified by the 802.3aq standard. The penalty is defined as the difference (in dB) of the equivalent signal to noise ratios (SNR) at the slicer input for the reference ideal channel model and for the measured waveform after propagation through a simulated fiber channel model.

1. Reference Channel Model

For the reference channel, rectangular on-off keyed pulses are transmitted and the laser and fiber are assumed perfect. The received pulse is a rectangular pulse with an amplitude measured in OMA of OMA_{RCV} and time duration of one bit period T. The receiver has a perfect matched filter front end matched to the rectangular receive pulse. The output of the matched filter is sampled once per bit period (without timing error) and presented to the decision element (binary slicer).

The signal to noise ratio at the slicer input determines the reference bit error rate (BER). The bit error rate is given by

$$BER_{REF} = Q(OMA_{RCV}(T/2N_0)^{1/2})$$

where N_0 is the one-sided power spectral density of the additive white Gaussian noise assumed for the receiver, and Q( ) is the Gaussian error probability function

$$Q(y) = \int_y^\infty \frac{1}{\sqrt{2\pi}} e^{-\frac{x^2}{2}} dx .$$

N_0 is set 13 dB electrical (6.5 dB optical) below the level required to give a BER of 1e-12. Hence

$$10 \log_{10}(OMA_{RCV}(T/2N_0)^{1/2}) = 8.47+6.5 = 14.97 \text{ dB} = SNR_{REF}$$

Without loss of generality, OMA_{RCV} is normalized to 1, and N_0 is set accordingly.

2. Waveform Measurement and Processing

The TP-2 penalty is calculated by Matlab code using a model as shown in Figure 1.

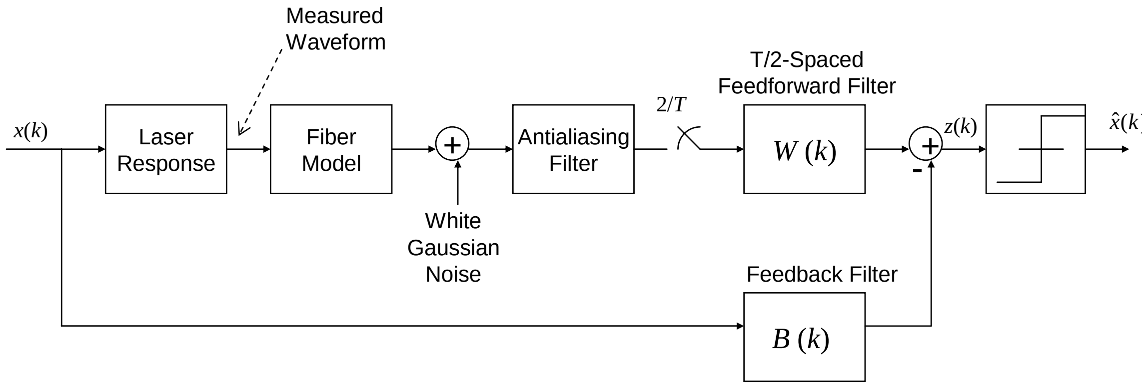

Figure 1. Model for TP-2 Penalty Calculation

The measured waveform for the transmitter device under test (DUT) is captured with a sampling oscilloscope. The data sequence driving the DUT is a PRBS9 or similar data pattern. The scope is set to capture at least one complete cycle of the data pattern, with at least seven or eight samples per bit period. (Fewer samples per bit period may be used depending on the high frequency content of the signal. The effective sampling rate must be high enough to avoid aliasing.) The scope includes a fourth-order Bessel Thompson filter with 3-dB electrical bandwidth of 7.5 GHz to filter the captured waveform. The scope is set to average over at least 16 patterns to average out noise in the captured waveform.

The inputs to the Matlab code are the following:

- The captured waveform (resampled, if necessary) corresponding to one complete cycle of the data sequence. The resampled waveform has 16 samples per bit period.
- The measured OMA of the sampled waveform, and the measured “off” power level (i.e., the steady state power level corresponding to a string of transmitted zeroes). Refer to the “off” power level as the bias.
- The data sequence used to generate the transmitted sequence. The data sequence must be aligned with the captured waveform (i.e., a rectangular pulse train based on the data sequence is aligned with the captured waveform within one bit period).

The captured waveform is processed as follows:

1) The waveform is passed through the simulated fiber channel(s) (Cambridge 2.1 or simulated channels equivalent to the channels in the TP3 test).
2) The bias is removed from the waveform and the OMA of the waveform is scaled to 1 to match the reference channel model. (Note: Scaling the OMA to 1 effectively sets the ratio of received OMA to

N_0 to the minimum allowed by the link budget.)

3) The simulated channel output signal is passed through an antialiasing filter. A fourth-order Butterworth filter of bandwidth 7.5 GHz is used for this purpose. (Note: removal of this filter is under consideration, since the signal has already been filtered by the Bessel-Thompson filter when it was captured.)

4) The antialiasing filter output signal is sampled at rate 2/T.

5) The sampled signal is processed by a standard fractionally-spaced MMSE-DFE receiver with 100 feedforward taps (at T/2 spacing) and 50 feedback taps. The feed-forward and feedback tap coefficients are calculated using a least-squares approach that minimizes the mean-squared error at the slicer input for the given measured waveform, assuming the noise properties defined in Section 1. Figure 1 shows the channel and equalizer model used for the least-squares calculation. The channel input is a periodic data sequence

$$\{x(0), x(1), \cdots, x(N-1)\},$$

where N is the length of one period (e.g. 511 for PRBS9). The measured waveform is assumed to be the output of an arbitrary laser response. According to the steps described above, the measured waveform is propagated through the fiber model and antialiasing filter, and then sampled at rate 2/T. (The periodicity of the measured waveform is utilized in the simulated propagation order to avoid edge effects caused by filter memory.) The reference DFE consists of a feedforward filter

$$\{W(-25), W(-24.5), \cdots, W(24.5)\}$$

and a feedback filter

$$\{B(1), B(2), \ldots B(50)\}.$$

Note that the feedforward filter is fractionally spaced and consists of 50 anticausal taps and 50 causal taps (including the tap at k=0). The feedback filter is symbol spaced and strictly causal, and does not have a tap at k=0. The input to the slicer is the periodic sequence

$$\{z(0), z(1), \cdots, z(N-1)\}.$$

The least-squares solution for the feedforward and feedback filters minimizes the quantity $\mathrm{E}\left\{\sum_{k=0}^{N-1}\left(z(k)-x(k)\right)^2\right\},$

where the expectation operator refers to the random sequence generated by the additive white Gaussian noise, filtered through the antialiasing filter and feedforward filter. (The periodicity of the inputs to both the feedforward and feedback filters is utilized in order to avoid edge effects caused by filters' memories.)

6) The bit-error rate is calculated by the semi-analytic method
   a) The Gaussian noise variance at the input to the slicer is calculated.
   b) For each bit in the data sequence, the equalized input to the slicer is calculated. For each input (bit) to the slicer (over one complete period of the data sequence), the probability of error is

calculated based on the variance of the filtered Gaussian noise and the distance from the slicer input to the decision threshold.

c) The probabilities of error are averaged over all slicer inputs (bits) to compute a total probability of error BER_{DUT}

7) The equivalent SNR in optical dB is deduced from the BER_{DUT} as follows:

   a) $SNR_{EQUIV} = 10 \log_{10}(Q^{-1} (BER_{DUT}))$

8) The penalty *p* is equal to the difference (in optical dB) between the equivalent SNR and SNR_{REF} from the reference model. Hence

$$p = SNR_{REF} - SNR_{EQUIV}$$