Measurement of Non-Stationarity of 10 Gb/s Multimode Fiber Links

Authors: Oscar Agazzi, Broadcom Corporation, and Tom Lenosky, Finisar Corporation.
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1. Introduction

This document suggests a standard procedure for the collection of data regarding the non-stationarity of a multimode fiber link. This data will then comprise a database which can be used to validate equalization methods.

In on-off keying (OOK) type optical links, a binary data stream is translated into a stream of on and off pulses of a laser or LED. Multimode optical fiber supports a number of optical modes, each propagating with slightly different modal delays, a phenomenon known as differential modal delay (DMD). Over a length of optical fiber, the differential velocities distort the signal and cause Inter-Symbol Interference.

Equalization schemes have been proposed to recover the original signal from the received signal. In an adaptive equalizer, the receiver is continuously adjusted to optimally resolve the correct signal, by a feedback loop that uses residual errors as input.
One key parameter for designing an adaptive equalization scheme for multimode fiber is the rate at which adaptation will have to occur. If the channel characteristics change too rapidly, the receiver will be unable to adapt. Time variance can occur because of motion of the fiber, because of temperature variation, or because of modal noise, i.e. time-varying selective excitation of fiber modes coupled with mode selective losses at connectors or within the fiber.

An overview of the test procedure we are proposing is as follows: (1.) A periodic signal is transmitted over a multimode fiber link. (2.) A high-speed receiver is used as input for an oscilloscope, which in effect acts as an analog to digital converter. (3.) The oscilloscope data is then collected and processed to compute a measure of time variance (or lack of stationarity) of the channel.

Additionally, we recommend procedures for introducing fiber motion and quantifying its effect. This is done by means of a specified "fiber shaker." Thermal variations are also prescribed. We also discuss how imperfect fiber connectors can be used to introduce modal noise.

2. Measurement Apparatus

A. Transmit sequence

The transmit sequence is a 2^7-1 bit PRBS, with at least a 1 GHz clock rate. Faster clock rates are desirable and encouraged, subject to equipment availability. If a faster clock rate is employed, then the sampling rate mentioned in (D.) below should be suitably rapid.

B. Laser diode

The laser diode is a gain-switched DFB at 1310 nm or a VCSEL at 850 nm. VCSELs with multiple transversal modes may degrade the SNR (see "Interpretation of results") substantially, so when conducting this experiment, the specifications of the VCSEL used should be carefully documented.

C. Fiber link

The fiber is supplied from a standard collection of "worst-case" fibers. It is already cut to a known length and fitted with connectors. Whenever possible, the fiber chosen should be sufficiently long so as to generate substantial intersymbol interference (ISI). The launch condition is arbitrary but should be carefully controlled: either a center launch or an offset launch is acceptable.

D. Receiver and Oscilloscope

The oscilloscope samples the received data at 1 GHz or higher. In effect, it is used as a high speed A/D converter. The receiver should have enough bandwidth to not limit the collection process. Faster sample rates are desirable, though equipment limitations may preclude their use. As many data points as possible should be collected in each data set, and archived to a networked computer.

The oscilloscope sampling clock should be synchronous with the transmit clock. If the accuracy of both transmit and sampling clocks is high enough, it may not be necessary to use explicit synchronization circuitry. The accuracy required from the clocks is such that the phase slip accumulated over the length of the block of samples captured is a small fraction of the sampling period. For example, if the block size is 65536 samples and the maximum deviation between transmit and sampling clocks is 1ppm, the accumulated phase slip will be $0.06T_s$ (where T_s is the sampling period). This is sufficiently small that it can be neglected.

3. Sources of Non-Stationarity

A. Fiber Motion

A fiber shaker should be built. It may be of any design as long as it induces large amplitude movements (> 1 in) and flexing in the optical fiber. The construction of a suitable fiber shaker is discussed in FOTP-142. Data should be collected with and without motion for comparison.

B. Temperature Change

The temperature of the fiber should be changed over a 10 degree range near room temperature, with a rate of change of a few degrees per minute. Data should be collected with and without thermal cycling for comparison.

C. Modal noise

To study the effect of modal noise, it is necessary to have mode-selective loss in the link. This can be done by introducing lossy connectors as discussed in FOTP-142. For the purposes of this document, it is sufficient to have one lossy connector which should be placed near the transmitter. This mode-selective loss connection shall have an optical loss of 1.0 dB +/- 0.25 dB when measured in accordance with FOTP-34, method 1.

4. Interpretation of Results

In the following it will be assumed that the samples of the signal are taken synchronously with the transmit clock as recommended in Section 2D. If this were not the case, the signal would have to be synchronously resampled using interpolation techniques. However the use of interpolation techniques is possible only if the signal has been sampled at a rate of at least twice its bandwidth. Since the bandwidth is not well controlled in these experiments, it may be necessary to use a high sampling rate. Therefore, the entire measurement procedure is simplified by using synchronous sampling.

The length of the block of samples captured should be adjusted to an integer multiple of the length of the PRBS. For example, if the transmit clock rate is 1Gb/s, the sampling clock frequency is 8GHz, and the block size is 65536 samples, it will be necessary to delete 512 samples and reduce the block size to 65024. This results in 64 repetitions of the 127-bit PRBS oversampled by a factor 8. By doing this, it is possible to periodically repeat the block of signal samples without edge effects.

An adaptive canceller as shown in Figure 1 is used to test the time invariance of the signal. If the signal is oversampeld, the canceller should have a polyphase structure as shown in Figure 2. These cancellers are implemented in software. The documentation of this experiment should include the description of the canceller software in source code, pseudo-code or a flow-chart form.

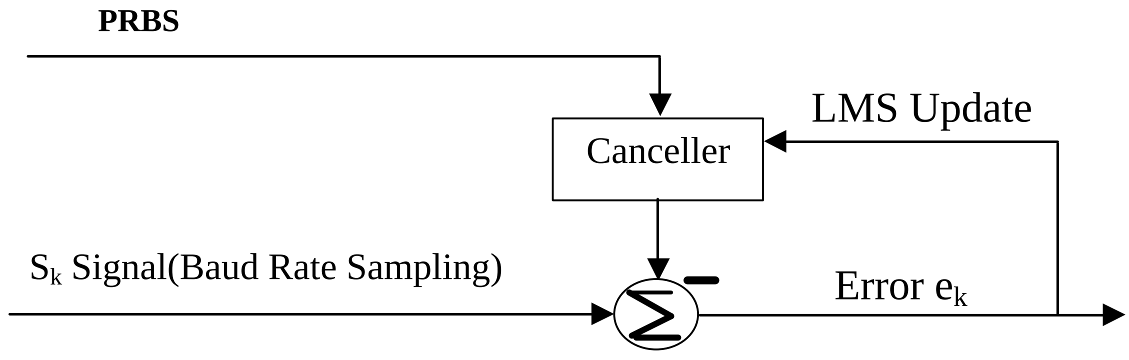

Figure 1

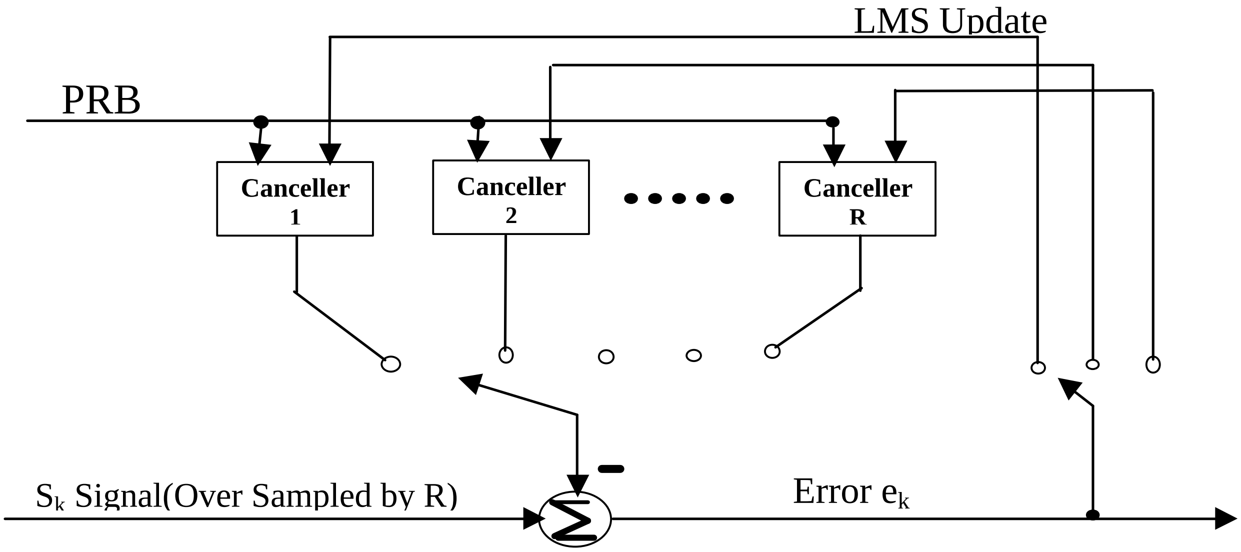

Figure 2

Before running the cancellation algorithm, it will be necessary to estimate the delay that needs to be introduced in the PRBS to achieve the best possible time alignment between the PRBS and the signal. This delay estimation can be easily done by inspection, using a plotting program. Alternatively, it can be done automatically by running an exhaustive search algorithm that minimizes the Euclidean distance between the vector of signal samples and the PRBS.

The canceller is adapted using the LMS algorithm for a sufficiently long time to ensure convergence, periodically repeating the signal and the PRBS if necessary. After convergence of the canceller, the samples of the error are stored in a file for the entire signal block (in the previous example, 65024 samples of the error are stored). Then the variance of the signal and the error are estimated using the expressions:

$$\sigma_s^2 = \frac{1}{N}\sum_{k=0}^{N-1} S_k^2 - \left(\frac{1}{N}\sum_{k=0}^{N-1} S_k\right)^2 \qquad (1)$$

$$\sigma_N^2 = \frac{1}{N}\sum_{k=0}^{N-1} e_k^2 - \left(\frac{1}{N}\sum_{k=0}^{N-1} e_k\right)^2 \qquad (2)$$

where S_k and e_k are the signal and error samples respectively.

The signal to noise ratio (SNR) is defined as:

$$\mathrm{SNR} = 10\log_{10}\left(\sigma_S^2/\sigma_N^2\right) \qquad (3)$$

The SNR is a measure of both the noise present in the observed signal and any non-stationarity of the signal that cannot be tracked by the LMS adaptation algorithm of the canceller. Slow time variations that can be tracked by the LMS algorithm will be automatically discounted and will not reduce the SNR as computed using (3). Although the SNR as defined in (3) is not equal to the slicer SNR in a receiver, the latter can be estimated from the former for a particular receiver structure, and related to the expected bit error rate (BER).