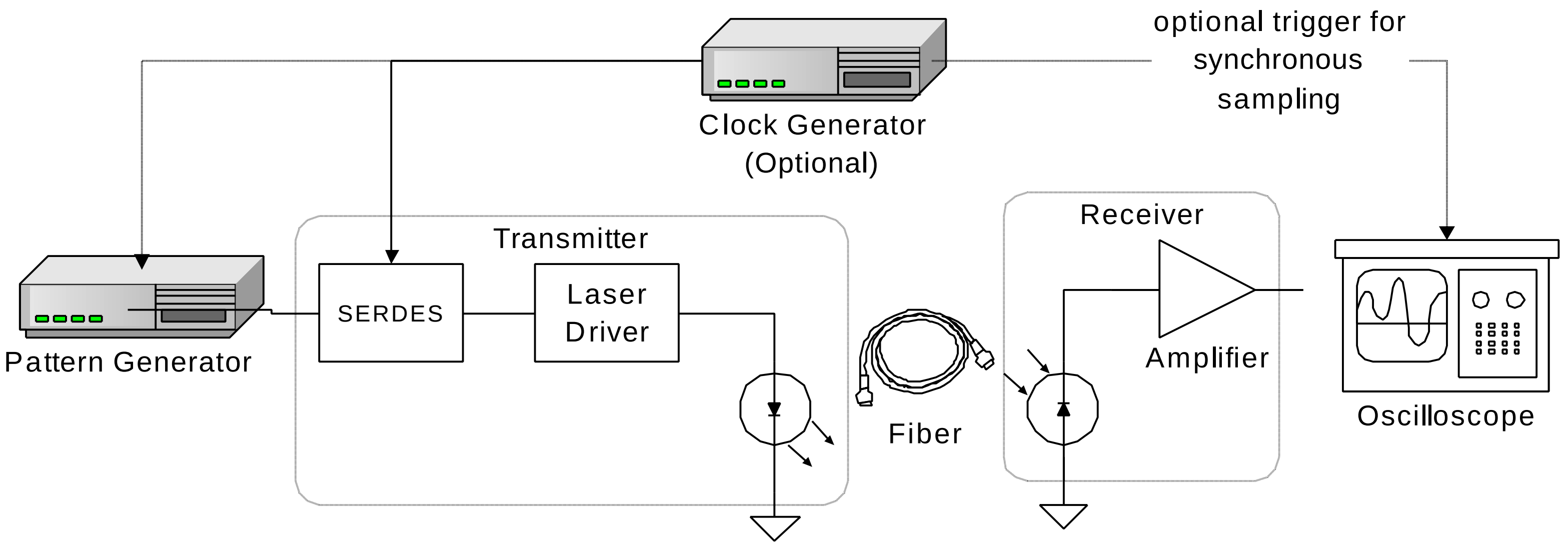
Proposed Method for Channel Estimation

Kishore Kota
Cicada Semiconductor Inc.
Austin, TX

Goals

- Understand the effectiveness of equalization on real data
- Determine non-linearities that might affect equalization
- Understand time-variance of the channel

Proposed Setup



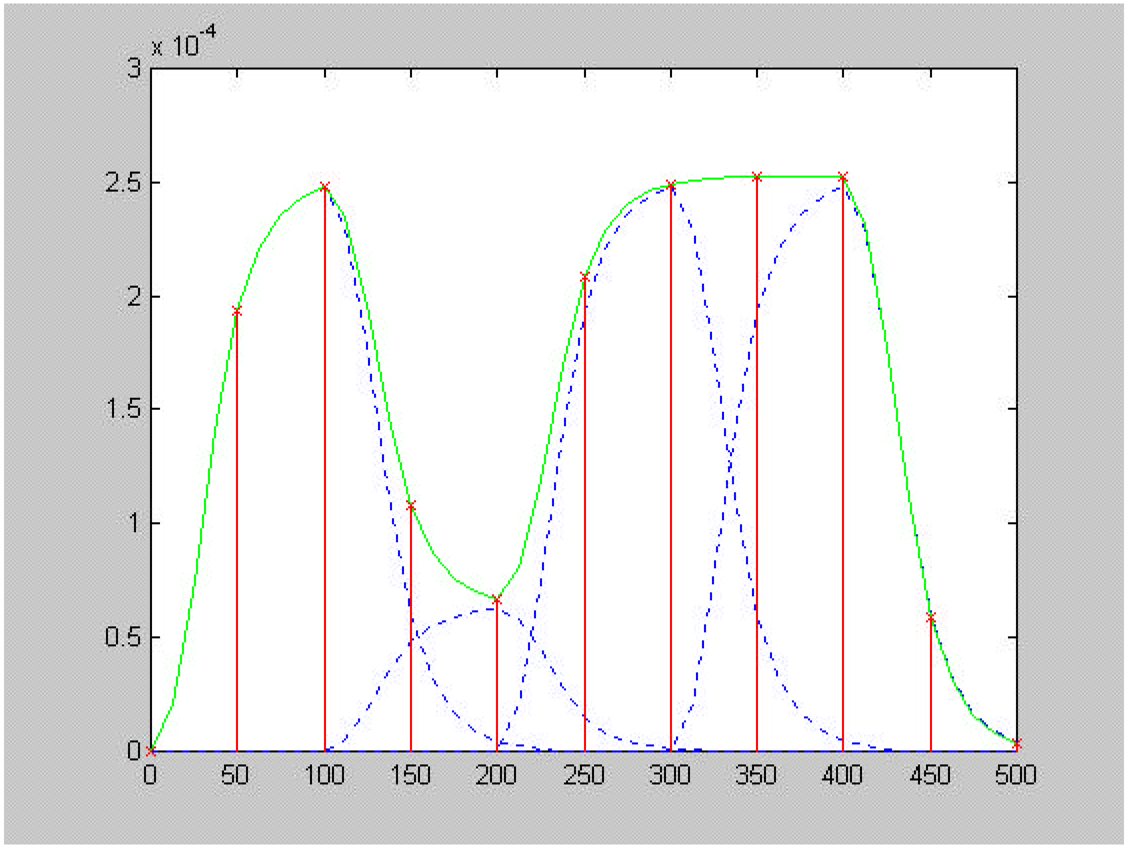
Proposed Setup

- Transmit a real bit stream through the laser+fiber+TIA and capture a real-time snapshot of the signal and save it in datafiles for post-processing
  - To simplify post-processing, the transmitted bit sequence is a repetitive pattern with a period no less than the expected length of channel impulse response
  - Use an off-the-shelf transmitter
  - On the receive side, the signal should be captured before any “clipping” introduced by the high-gain TIA. It will be necessary to operate the transmitter at a power level which ensures this condition
  - Sampling period should be atleast twice the baud-rate. The goal should be to capture the longest snapshot allowed by the equipment without violating this condition. A longer snapshot is preferable to higher time resolution
  - Capture multiple snapshots using the same setup to assess repeatability of the experiment and to measure channel variations over a period of seconds to minutes

What will this data tell us?

- the expected performance of various equalizer architectures
- whether the channel can be considered linear
- the degree of non-linearity modelling should it be necessary
- short term channel variations which occur on the order of a few 100s to 1000s of symbols can be deduced by processing data within a snapshot
- longer term variations on the order of a few seconds to minutes can be deduced from the variations from one snapshot to another
- If there are frequent persistent abrupt changes in the channel response, it is likely that the data will show this

Quasi-linearity of the channel



Quasi-linear Model of the Data

- Suppose a_k is the k^{th} transmitted symbol, the instantaneous transmitted optical power can be written as:

$$\sum_{k=-\infty}^{+\infty} a_k p(t-kT)$$

- The n^{th} receive sample y_n can be written as:

$$y_n = \sum_{k=-\infty}^{+\infty} a_k h(nT_s - kT)$$

- p(t) is the time domain pulse transmitted for each symbol
- "T" is the baud interval
- "T_s" is the sample interval
- h(t) is the received time domain pulse for each symbol

Example

- baud rate is 10Gb/sec implies T=100ps
- extinction ratio of 6dB implies a_k is drawn from the set {1,1/4} corresponding to the power levels for 1 and 0 respectively
- p(t) would be the transmitted time domain pulse for an isolated "1"
- The received pulse h(t) is the transmit pulse modified by the channel and receiver characteristics
- The received signal is the superposition of the pulses h(t) spaced "T" apart and modulated by the transmitted symbol

Modelling non-linearities

- Previous example assumed that the pulse for any symbol is scaled version of a base pulse p(t)
- First order model: We could assume that the pulse shape for a_k will be different depending on whether a 0 or 1 is transmitted i.e. $p_0(t)$ for a transmitted “zero” and $p_1(t)$ for a transmitter “one”, which might not be scaled versions of a base pulse
- Second order model: We can assume that the pulse shape for a_k will depend on the previous symbol a_{k-1} in addition to the current symbol i.e. $p_{00}(t)$, $p_{01}(t)$, $p_{10}(t)$, $p_{11}(t)$ for the four combinations
- In general, we can assume that the current pulse depends on the last K-1 symbols and the current symbol
- Complexity of the model grows exponentially as 2^K possible pulses

Post-processing steps

- Resample the received samples synchronously
  - assume that the signal is bandlimited
  - resample using any commonly known interpolation techniques
  - estimate the correct phase and frequency from the received data using spectral line techniques
  - standard deviation of the estimates will be improve as sqrt(N) where N is the number of samples in the snapshot
  - Can bypass this step if the equipment allows synchronous sampling of the received signal
- Choose a sliding window of “M” samples for the channel estimate
  - Channel estimate obtained through least-squares solution of an overconstrained system of linear equations
  - Complexity of the system of equations will increase with the degree of non-linearity
  - Higher degree of non-linearity will require larger M

Limits on the estimates

- If “N” samples are obtained in any snapshot, and each channel estimate is obtained using “M” samples, then it is possible to obtain N-M+1 estimates of the channel impulse response from each snapshot
- The accuracy of estimates improves as sqrt(M)
- However, a larger M will average any channel variations over the same time interval
- The value of “M” needs to be small enough to observe any channel variations while keeping the error bounds on the estimates within reasonable limits
- This is the classic “uncertainity” principle. There is a limit on simultaneously resolving the variation in channel response and obtaining accurate estimates
- The limit will depend on the noise in the measurements

A caveat about time-variance

- What kind of channel variations can we expect?
  1. Large abrupt variations
     - Defined as variations which cannot be tracked by any signal processing and will cause a loss of link
     - Needs to be quantified as “link outage” per year and kept within reasonable limits
     - this experiment is unlikely catch these variations unless they occur very often
  2. Fast variations
     - Defined as variations which cannot be tracked by signal processing but do not result in a loss of link
     - These channel variations will appear as additional noise
     - In this experiment these variations cannot be distinguished from other noise sources
  3. Slow variations
     - Defined as channel variations which can be tracked by signal processing
     - the experiment can measure these variations limited by the uncertainity principle within each snapshot
     - Can measure variations across snapshots within limits of the error bounds

What's left?

- Identify the available equipment
  - What wavelength should we use?
  - What baud rate should we use?
  - What type of fiber should we use?
  - Is it possible to control the laser transmit power to prevent clipping in the TIA?
  - Is it possible to capture a long enough snapshot of the output of the TIA?
  - What type of oscilloscopes are available for this purpose?

Pros and cons of this setup

- Pros
  1. the setup makes use of off-the-shelf optical transceivers
  2. Allows measurement noise to be averaged enough to reduce error bounds on channel estimates
  3. this is probably the closest you can get to demonstrating equalization without actually building a chip
  4. preserves most non-linearities, noise, channel variations, jitter and other impairments which can affect equalization
  5. allows a tradeoff between accuracy of estimates versus channel variations in the post processing step
- Cons
  1. It is only one sample point. Does not tell you anything about all possible variations in the field
  2. fundamental limits on how accurately channel variations can be measured