Algorithms for Coherent Detection

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What can we learn from other fields?

- Fiber optics is converging with other digital communications fields – cellular, satellite, digital subscriber line, etc.
- But techniques from existing fields are of limited help
  - our carrier linewidth is much higher, 10^{-4} baud rate
  - our impairments are of long duration, >100 symbols
  - our data rate is much higher, 100Gb/s
- We must develop our own algorithms
- Existing field of estimation theory says how to develop algorithms
  - best possible estimate is the optimal estimate
  - other ad hoc estimate may be good enough, or not
  - must compare ad hoc estimate with optimal estimate to find out how good it is

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When is an optimal estimate needed?

Continuous signal

Non-optimal estimate will work fine.	Near-optimal estimate needed, otherwise excess bit errors or other failure occurs.	Failure occurs even when optimal estimate is used. Need a new strategy.

slowly varying parameter ———————— rapidly varying parameter

- Laser phase is example of parameter in orange region today

Burst signal

Near-optimal estimate needed. If not, bit errors occur, or data loss because of time to acquire parameter at start of burst.	Failure occurs even when optimal estimate is used. Need a new strategy.

slowly varying parameter ———————— rapidly varying parameter

- In the future all parameters will benefit from optimal estimation

General references

Digital communications

- John G. Proakis, "Digital Communications," McGraw-Hill, 4th ed., 2000.

Parameter synchronisation (e.g. phase estimation)

- U. Mengali, A.N. D'Andrea, "Synchronization techniques for digital receivers," Plenum Press, 1997.
- H. Meyr, M. Moeneclaey, S.A. Fechtel, "Digital communication receivers: synchronization, channel estimation & signal processing," Wiley, 1998.

IC design considerations

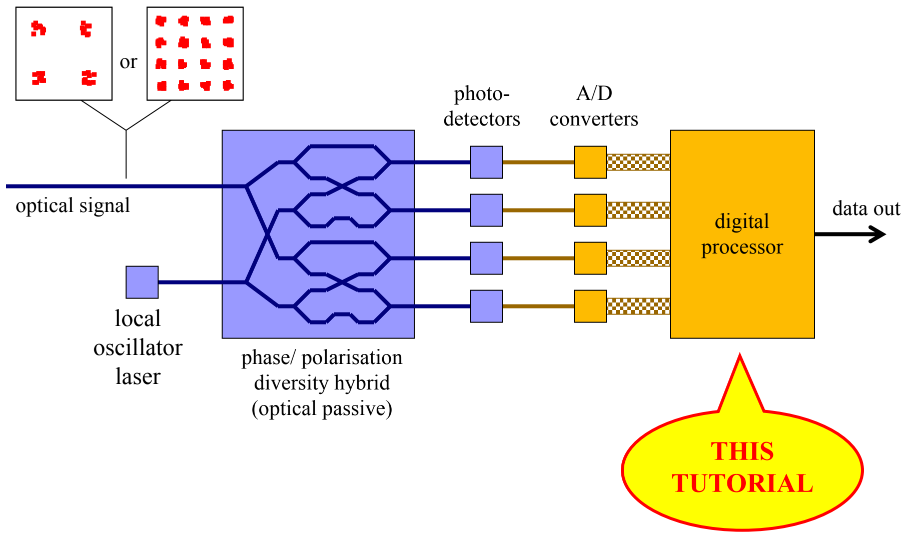
- Keshab K. Parhi, "VLSI digital signal processing systems: Design and implementation," Wiley, 1999.

Outline

- What is inside a coherent receiver?
- Correction of hybrid imperfections
- Polarisation estimation
- Phase estimation
- High order modulation formats, e.g. QAM
- Impairment compensation
  - chromatic dispersion
  - polarisation mode dispersion
  - nonlinear refractive index

Coherent receiver hardware

Coherent receiver hardware



Published ASIC hardware designs

	Nortel (now Ciena)	U. Paderborn
A/D converters	CMOS, same chip as processor	SiGe BiCMOS 5 bits, 12.5GSa/s
processor fabric	90nm CMOS 20 million gates	130nm CMOS 16 degree parallel 319000 gates + devices
signal processed	11.5Gbaud DP-QPSK	2.5Gbaud DP-QPSK demonstrated (limited by inadequate noise filtering), design target 10Gbaud
impairment compensation	50000ps/nm CD 50ps 1st order PMD	
	Sun, Optics Express, vol. 16, p. 873-879, 2008.	Herath, OFC 2009, paper OThE2. Pfau, OFC 2009, paper OThJ4. Adamczyk, LEOS summer topical 2008, paper TuC3.2.

- Main constraints are number of gates on chip and A/D bandwidth
  - algorithms must be computationally efficient

Correct non-ideal hybrid

Correct for imperfections in optical hybrid

- Phase/polarisation diverse may deviate from ideal
  - polarisation axes not 90° apart
  - phase difference between outputs not exactly 90°
- Correct actual A/D voltages via 4x4 matrix multiplication

$$\begin{pmatrix} V_1' \\ V_2' \\ V_3' \\ V_4' \end{pmatrix} = \begin{pmatrix} \mathrm{Re}[\hat{p}_{1x}] & \mathrm{Im}[\hat{p}_{1x}] & \mathrm{Re}[\hat{p}_{1y}] & \mathrm{Im}[\hat{p}_{1y}] \\ \mathrm{Re}[\hat{p}_{2x}] & \mathrm{Im}[\hat{p}_{2x}] & \mathrm{Re}[\hat{p}_{2y}] & \mathrm{Im}[\hat{p}_{2y}] \\ \mathrm{Re}[\hat{p}_{3x}] & \mathrm{Im}[\hat{p}_{3x}] & \mathrm{Re}[\hat{p}_{3y}] & \mathrm{Im}[\hat{p}_{3y}] \\ \mathrm{Re}[\hat{p}_{4x}] & \mathrm{Im}[\hat{p}_{4x}] & \mathrm{Re}[\hat{p}_{4y}] & \mathrm{Im}[\hat{p}_{4y}] \end{pmatrix}^{-1} \begin{pmatrix} V_1 \\ V_2 \\ V_3 \\ V_4 \end{pmatrix}$$

The $\begin{pmatrix} \hat{p}_x \\ \hat{p}_y \end{pmatrix}$ refer to Jones vectors of output ports of hybrid, calibrated at factory

- Can correct for large deviations in hybrid angle, e.g. 120° if hybrid is a 3x3 coupler
  - fails only when angle close to 0°

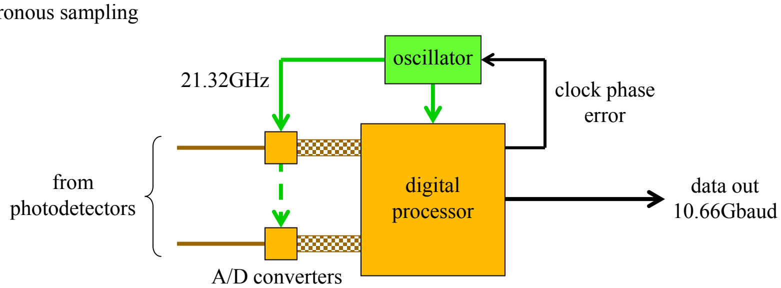
Clock recovery

Clock recovery/retiming

In principle A/D sampling can be asynchronous with signal symbol rate, use interpolation in DSP to calculate symbol centre voltages

- E.g. 20GSa/s A/D conversion, 10.66Gbaud signal
- mixed clock processor fabric
- FIR filter performs interpolation, continually updated coefficients

In practice that uses too much processor space, better to have hardware-based clock control with synchronous sampling

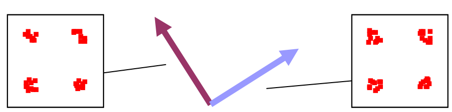


- For most modulation formats optical power vs. time contains symbol clock content
- Many algorithms available for clock phase error, e.g. early-late detector, zero crossing detector, Mueller & Mueller detector

Polarisation estimation

Polarisation estimation

- Polarisation axes of signal may not be aligned with optical hybrid

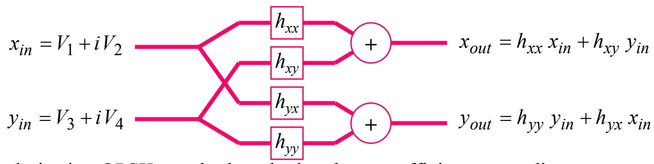


- Stokes matrix multiplication aligns performs realignment
- Need estimate of signal's SOP to obtain Stokes matrix

- SOP estimate is straightforward for single polarisation signal
  - transform each observed E-field (Jones vector) at symbol centre into point in Stokes space (on Poincaré sphere)
  - average over time, according to rate of change of signal SOP
  - transform back to Jones vector

Polarisation estimation

- Butterfly structure performs Stokes matrix multiplication



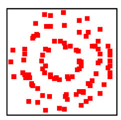
- For dual polarisation QPSK standard method updates coefficients according to constant modulus algorithm (CMA)

$$h_{xx} = h_{xx} + \mu\,\varepsilon_x\,x_{out}\,x_{in}^* \qquad h_{xy} = h_{xy} + \mu\,\varepsilon_x\,x_{out}\,y_{in}^*$$

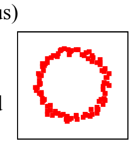
$$\underbrace{h_{yx}}_{\textit{next iteration}} = \underbrace{h_{yx} + \mu\,\varepsilon_y\,y_{out}\,x_{in}^*}_{\textit{current iteration}} \qquad \underbrace{h_{yy}}_{\textit{next iteration}} = \underbrace{h_{yy} + \mu\,\varepsilon_y\,y_{out}\,y_{in}^*}_{\textit{current iteration}}$$

  - where $\varepsilon_x = 1 - |x_{out}|^2$ and $\varepsilon_y = 1 - |y_{out}|^2$
- CMA tries to keep constellation a narrow ring (i.e. constant modulus)

polarisation axes misaligned



polarisation axes aligned



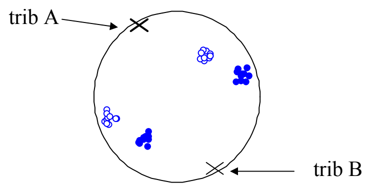
Savory, Optics Express, vol. 16, p. 804-817, 2008. Godard, IEEE Trans. Commun., vol. 28, p. 1867-1875, 1980.

Polarisation estimation

- Two outputs of CMA may converge independently – corrects for polarisation dependent loss
  - but must use special initial condition to avoid converging to same tributary [Liu, OFC 2009, paper OMT2]
- Constant modulus algorithm uses feedback
  - feedback from previous symbol not feasible in parallel processor
  - must use feedback from distant past symbol
  - limits ability to track fast SOP changes
- Other similar SOP estimates proposed, data aided estimates using stochastic gradient algorithm, with different error functions ε_x , ε_y
  - pilot symbol approach [Han, Optics Express, vol. 13, p. 7527-7534, 2005]
    - probably worse than CMA because of infrequent pilot symbols
  - decision feedback [Savory, Optics Express, vol. 16, p. 804-817, 2008]
    - better error function than CMA once approximate *h* coeffs found (by CMA)
    - but in real processor available decided symbols are even older than CMA feedback variables
    - decision feedback better when delay up to 4x longer than CMA delay
    - need further study to decide which is best for a realistic processor

Polarisation estimation

- Feedforward SOP estimate will not suffer reduced bandwidth due to long feedback delay
- With dual polarisation QPSK signal has four possible polarisation states lying on great circle on Poincaré sphere
  - tributary SOPs are poles of great circle



- Find tributary SOPs by
  - cross product of neighbouring symbol signal Stokes vectors to give two poles
  - polarise result in direction of coarse estimate of trib SOP (a hemisphere) to obtain one pole
  - average over time
  - convert back to Jones vector
- Feedforward method should have more bandwidth that CMA
  - need to compare realistic CMA (with finite feedback delay) with feedforward method

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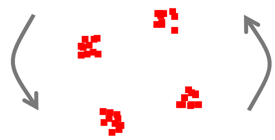
Phase estimation

Phase estimation

- Constellation spins over time



- Two contributions to rotation
  - fixed rotation speed, equal to difference frequency between signal and LO
    - easy to estimate
  - random walk in phase due to finite linewidth of signal & LO lasers
    - hard to estimate
    - may add significant penalty to receiver sensitivity unless special narrow linewidth lasers are used

Phase estimation

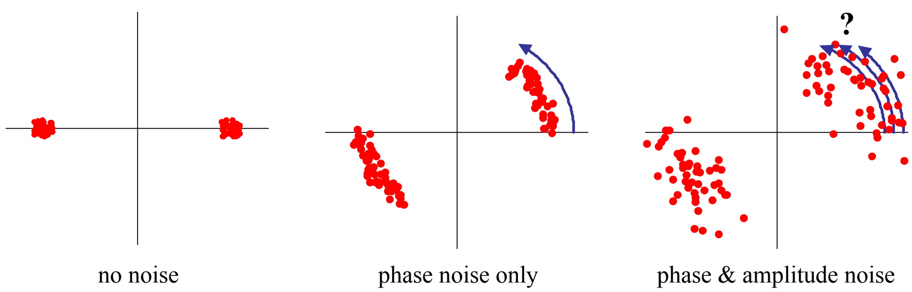
- To estimate frequency offset of QPSK signal
- Raise to 4th power to remove data modulation

$$y = x^4$$

- Either time average of phase jump from one symbol to next $y^*(n-1)\ y(n)$

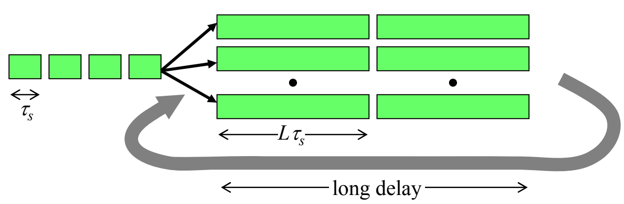
  [Leven, IEEE Phot. Tech. Lett., vol. 19, p. 366-368, 2007]
- Or find peak of short term Fourier transform of y

Phase estimation

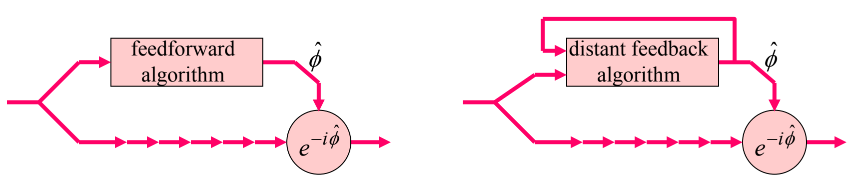


- Choose time-averaging function that tracks phase noise but filters out most of amplitude noise
- Even optimal phase estimate may deviate significantly from actual phase when both phase noise and amplitude noise present (realistic scenario)
  - causes increase in bit errors and cycle slip events
  - if optimal phase estimate is not accurate must upgrade to narrower linewidth lasers
- Need a phase estimate that is close to the optimal phase estimate

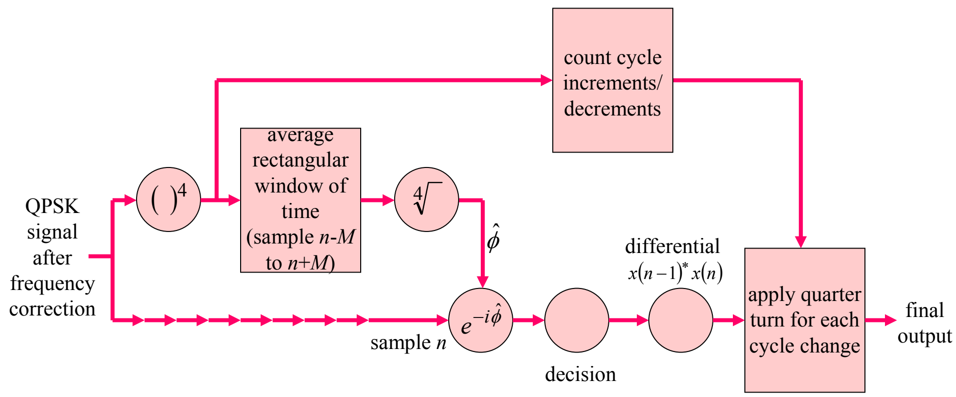
Parallel DSP architectures



- The DSP must operate in parallel because maximum clock speed < symbol rate
  - parallel operation is equivalent to a delay in computing a result
  - result n-1 is not available to compute result n
  - algorithms employing feedback of previous result cannot be executed
- Two solutions considered



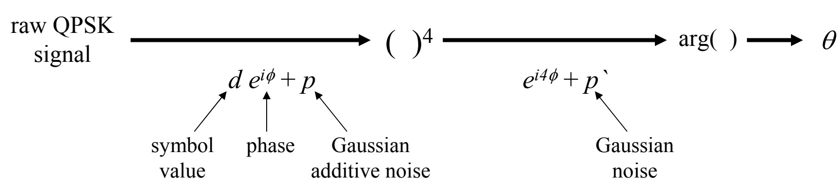
All-feedforward phase estimate



[Noe, IEEE Phot. Tech. Lett., vol. 17, p. 887-889, 2005]

- No feedback, so can be implemented in parallel processor
- Rectangular window deviates from optimum window shape, but result is acceptable unless laser linewidth very close to limit

Open loop Wiener phase estimate



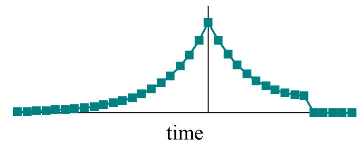
- Applying small angle approximation

$$\theta \;=\; 4\phi \;+\; \text{(additive noise component)}$$

Wiener process (Gaussian random walk) Gaussian noise

- Estimation theory says that best linear estimate of ϕ is Wiener filter applied to θ

$$\hat{\theta}(z) = \frac{(1-\alpha)\left(\alpha^D + (1-\alpha)\sum_{k=1}^{D}\alpha^{D-k}z^{-k}\right)}{1-\alpha z^{-1}}\theta(z)$$



- Non-causal filter sees forward D symbols (chosen by designer) as well as to infinite past

[Taylor, IEEE JLT, vol. 27, p. 901-914, 2009. Ip, IEEE JLT, vol. 25, p. 2675-2692, 2007.]

Look-ahead computation

- Wiener filter may be close to optimal, but it uses feedback from immediately preceding result – not allowed!

$$\frac{(\cdots)}{1-\alpha z^{-1}}$$ feedback from previous result

- To resolve, apply look-ahead computation so these relationships refer to a distant past result, L symbols ago [Parhi, "VLSI digital signal processing systems," Wiley (1999)]
  - multiply numerator and denominator by same polynomial

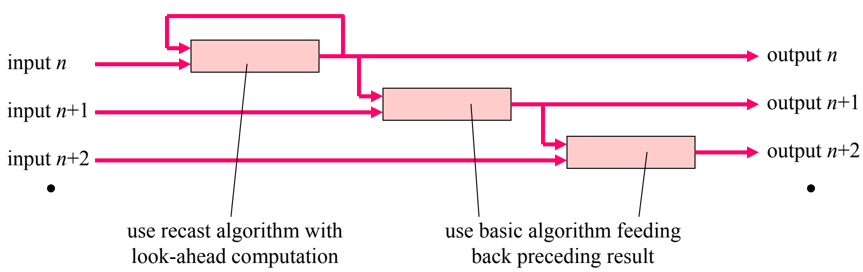
$$\frac{(\cdots)}{1-\alpha z^{-1}} \frac{\sum_{k=0}^{L-1}\alpha^k z^{-k}}{\sum_{k=0}^{L-1}\alpha^k z^{-k}} \Rightarrow \frac{(\cdots)\sum_{k=0}^{L-1}\alpha^k z^{-k}}{1-\alpha^L z^{-L}}$$

L symbols past

  - now uses feedback to L symbols ago
  - but at expense of additional L feedforward taps

How to reduce amount of computation

- Incremental block processing allows L-1 of L paths to use original algorithm



- Power-of-2 decomposition replaces sum of D terms in feedforward polynomial with product of log2(D) polynomials
  - corresponds to sequence of FIR filters, each coarser in time than the previous one
  - use in L-1 paths that have instant feedback
- Example of number of multiplications per symbol for D = 16 Wiener filter

base Wiener filter	18
recast Wiener filter with L = 16	33
recast Wiener filter with incremental block processing & power-of-2 decomposition	8.93

Management of cycle slips

- Cycle slips occur sometimes for realistic linewidth, additive noise level
- Phase estimate becomes wrong by quarter turn, and stays there indefinitely
  - causes continuous bit errors after FEC decoding
- To avoid continuous errors, precode at transmitter, and then differential logical operation after symbol decision
  - cycle slip event becomes 1 bit error
  - bit error in channel becomes 2 bit errors
  - doubling of BER equivalent to 0.8dB penalty
- Penalty can be avoided by carefully chosen FEC code
  - bit error occur in short bursts up to 4 bits long
  - many codes available that correct short bursts, e.g. cyclic codes
  - interleaving spreads out burst, so reorganise FEC code structure
    low gain FEC decode, then interleave, then high gain FEC decode

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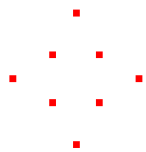
High order modulation formats - QAM

Quadrature amplitude modulation

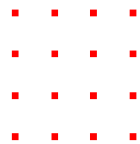
- Several system demonstrations using quadrature amplitude modulation

8-QAM



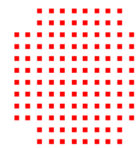
Zhou, OFC 2009, paper OWG3.

16-QAM



Sakamoto, ECOC 2008, paper Tu.1.E.3.
Winzer, to be published in IEEE JLT.
Mori, OFC 2009, paper OWG7.

128-QAM



Nakazawa, OFC 2009, paper OThG1.
(uses optical PLL)

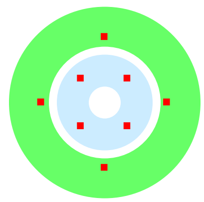
- Unlike QPSK, constellations do not have constant modulus, so CMA tracks SOP changes slowly
- Raising to 4th power (or any other power) does not remove data modulation
- All experiments to date use offline processing
  - use decision directed algorithms for polarisation estimation & phase estimation with 1 symbol delay
  - not representative of realistic digital processor

QAM phase estimation

- Decision directed operation is alternative to raising to 4th power to remove modulation
  - but in parallel processor decision result is many symbols old
    - causes phase estimate to be in error
    - may lead to bit error extension events, where bit errors cause phase error which causes more bit errors – lead to burst of errors even after FEC decoding
    - feedback delay reduces linewidth tolerance by factor of 8.5 for typical processor [Pfau, IEEE J. Lightwave Technol., vol. 27, p. 989-999, 2009]
- Radius directed phase estimate proposed [Seimetz, OFC 2008, paper OTuM2]
  - acts on only a subclass of constellation points that have 4-fold symmetry
  - predicts 30kHz laser linewidth tolerable for 10Gbaud 16-QAM
- Feedforward phase estimate proposed [Pfau, IEEE J. Lightwave Technol., vol. 27, p. 989-999, 2009]
  - searches over many phase values in parallel, e.g. 16 to 64 values
  - selects phase having lowest mean Euclidean distance metric
  - predicts 700kHz laser linewidth tolerable for 10Gbaud 16-QAM

QAM polarisation estimation

- Radius directed version of CMA demonstrated with 8-QAM [Zhou, OFC 2009, paper OWG3]



- Estimate which ring current point belongs to from instantaneous power
- Treats different rings of constellation separately
  - CMA behaves similarly to QPSK
- Should work with other QAM constellations

Orthogonal frequency division multiplexing

How to process OFDM signals

- With OFDM information imposed in frequency domain on many digitally generated subcarriers
  - one symbol is long, e.g. 100s bits
  - receiver performs digital Fourier transform of each symbol to see subcarriers
  - signal appears like noise in time domain
- Before DFT operation receiver must
  - locate symbol boundaries
  - correct for frequency offset due to LO off center
  - correct drifting phase due to finite laser linewidth

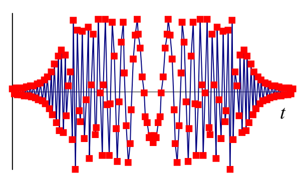
How to process OFDM signals

- Timing & frequency offset obtained via two synchronisation symbols [Schmidl, IEEE Trans. Commun., vol. 45, p. 1613-1621, 1997]
  - first symbol has second half repeat of first half – all odd subcarriers are zero
  - autocorrelation at (symbol time)/2 spacing has peak indicating symbol boundary
  - value of autocorrelation peak provides initial frequency estimate, within ±1/2T
  - second symbol is training symbol that gives exact frequency estimate
- Phase is estimated by averaging many separate subcarrier phase estimates
  - either raise each subcarrier to 4th power, or leave some subcarriers unmodulated as pilot carriers [Yi, IEEE Phot. Tech. Lett., vol. 19, p. 919-921, 2007]
  - OFDM phase estimate involves averaging over subcarriers, while TDM phase estimate averages over time

Impairment compensation

CD compensation by FIR

- Propagation impairment compensation executed before data recovery
- Basic method of CD compensation is FIR filter, typically 2 taps per symbol

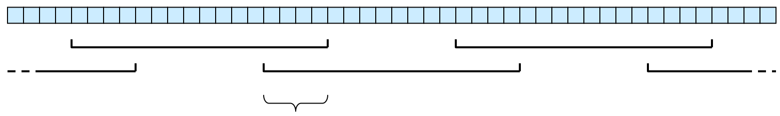


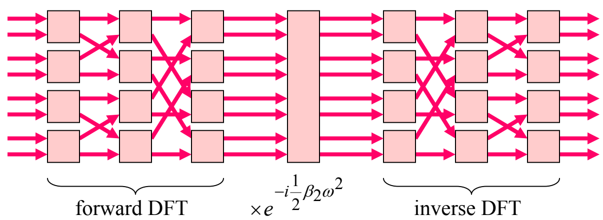
  - filter coefficients are inverse Fourier transform of CD transfer function $\exp(i\beta_2\omega^2/2)$, truncated in time [Taylor, IEEE Phot. Tech. Lett., vol. 16, p. 674-676, 2004]
  - number of taps $\approx 4\lambda^2 D/c\,\tau_{sym}^2$
  - for compensation of long systems number of taps becomes very large, e.g. 140 taps for 2500km NDSF at 10Gbaud

OThL4.pdf

CD compensation by DFT

- Discrete Fourier transform method user fewer computations
  - process samples in large overlapping blocks

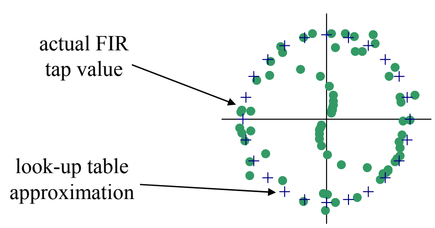




- Number of multiplications per symbol ~ $\log(D/\tau_{sym}^2)$
- DFT typically requires fewer multiplication than FIR for > 50 FIR taps

Alternative CD compensation algorithms

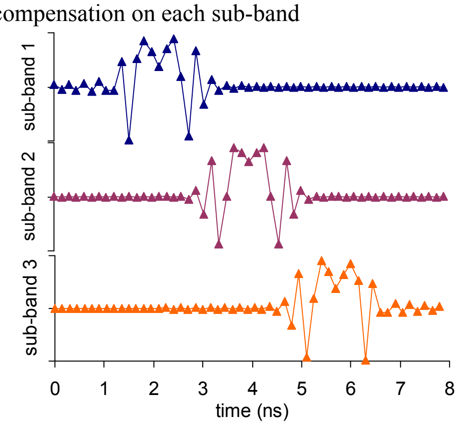
- IIR filter proposed [Goldfarb, IEEE Phot. Tech. Lett., vol. 19, p. 969-971, 2007]
  - fewer multiplications than FIR filter because it uses feedback to generate long impulse response
  - 40 real taps compensates for 2500km NDSF at 10Gbaud
  - but feedback delay not considered
- Circular coefficient approximation [Taylor, OFC 2008, paper OTuO1]



  - most FIR coefficients lie near circle on complex plane
  - use look-up table of coeff values on circle instead of calculating each time
  - for 10Gbaud over 2000km NDSF reduces number of effective additions from 1485 to 384, compared to standard FIR

Alternative CD compensation algorithms

- Sub-band processing [Taylor, OFC 2008, paper OTuO1]
- Divide signal into several spectral sub-bands using bank of FIR filters
  - each sub-band has narrow spectrum and is described by lower sample rate
  - execute CD compensation on each sub-band



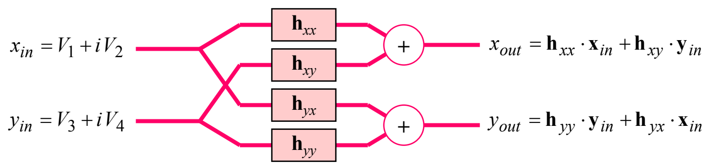
- Saves on FIR taps by (number of sub-bands), but requires extra processing for filter bank
  - marginal improvement vs. standard FIR, but may benefit from IIR in sub-bands

Alternative CD compensation algorithms

- DFT chromatic dispersion method uses fewest computations
  - but it does not take advantage of any features of CD transfer function
- Worth seeking alternative CD compensation algorithm
  - over half of real estate of realistic processor working on CD compensation

PMD compensation

- Polarisation mode dispersion has much shorter impulse response than CD, a few unit intervals, but it varies over time
- PMD compensated by adaptive equaliser [Savory, Optics Express, vol. 16, p. 804-817, 2008]



  - four FIR filters in butterfly configuration, filter coefficients set by CMA

$$\mathbf{h}_{xx} = \mathbf{h}_{xx} + \mu\varepsilon_x x_{out} \mathbf{x}_{in}^* \qquad \mathbf{h}_{xy} = \mathbf{h}_{xy} + \mu\varepsilon_x x_{out} \mathbf{y}_{in}^*$$

$$\underbrace{\mathbf{h}_{yx}}_{\textit{next iteration}} = \underbrace{\mathbf{h}_{yx} + \mu\varepsilon_y y_{out} \mathbf{x}_{in}^*}_{\textit{current iteration}} \qquad \underbrace{\mathbf{h}_{yy}}_{\textit{next iteration}} = \underbrace{\mathbf{h}_{yy} + \mu\varepsilon_y y_{out} \mathbf{y}_{in}^*}_{\textit{current iteration}}$$

  - where $\varepsilon_x = 1 - |x_{out}|^2$ and $\varepsilon_y = 1 - |y_{out}|^2$
  - also tracks varying signal SOP
- CMA tested in simulations & experiment against 1st order PMD, rapidly changing SOP
  - not yet tested against all orders of PMD, time-varying PMD
- Solution where 1st & 2nd order PMD are estimated and then compensated should track varying PMD better than CMA

How low can sample rate go?

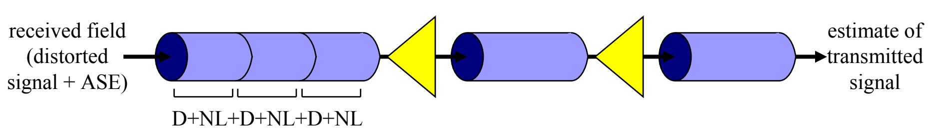
- Executing impairment compensation at lower samples/symbol (ideally 1 sample/symbol) is better
  - FIR-type CD compensation ~ (samples/symbol)^2
  - DFT-type CD compensation ~ (samples/symbol)
  - A/D converter operates at lower sample rate
- Symbol spaced equalisation possible in principle
  - sinc shaped transmit pulses
  - sinc shaped receiver impulse response
  - 1 sample/symbol satisfies Nyquist criterion
- But sinc shaped impulse response impossible via analog hardware
- Solution: 2 samples/symbol front end followed by sinc impulse receive filter, then 1 sample/symbol impairment compensation

How low can sample rate go?

- For conventional pulse shapes symbol spaced equaliser is of some benefit [Ip, IEEE J. Lightwave Technol., vol. 25, p. 2033-2043, 2007]
  - digital signal waveform is compensated correctly by optimal equalising filter (zero forcing filter), but it enhances noise (line noise and digitisation noise)
  - can equalise differential group delay (CD or PMD) of about 0.6 symbol
  - useful for CD compensation trimming application, where DCF is already in place
- Ip & Kahn's study shows lower limit on sampling rate using realisable filters is 3/2 samples/symbol
  - unlimited CD can be compensated
  - moderate penalty compared to CD compensation with 2 samples/symbol
  - Butterworth anti-aliasing filter
  - alternate symbol centre values must be calculated by interpolation

Compensation of fiber nonlinear effects

- Ability to numerically compensate nonlinear effects means power per channel can be raised above today's limit to increase SNR
- Nonlinearity compensation done by backpropagation using split step Fourier method



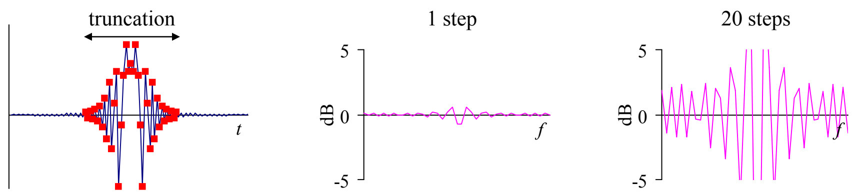
- Issues to be addressed are:
  - Will NL compensation take too many computations?
  - Will parameters not known by algorithm limit its benefit?
    - backpropagated ASE noise
    - NL phase shifts from other WDM channels
    - exact polarisation evolution through fiber

Compensation of fiber nonlinear effects

- Single channel NL compensation demonstrated with 1 step/span
  - experiment extended distance from 520km to 1180km [Goldfarb, IEEE Phot. Tech. Lett., vol. 20, p. 1887-1889, 2008]
  - simulation extended distance from 2000km to 6400km [Ip, IEEE J. Lightwave Technol., vol. 26, p. 3416-3425, 2008]
    - 3 samples/symbol needed
    - 1 step/span has 1dB worse phase error than 10 steps/span for 10.7Gbaud
    - parametric amplification of backpropagated ASE increases apparent noise level above actual noise level
    - no indication that backpropagated ASE amplification rolls off as launch power increases, so this effect may limit usefulness of NL compensation

Compensation of fiber nonlinear effects

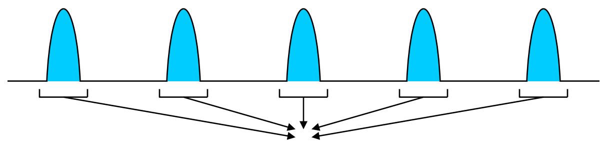
- But number of computations needed much larger than CD compensation only
  - dominated by computations for CD compensation
- FIR filter for CD for one backpropagation step calculated by truncation of CD impulse response



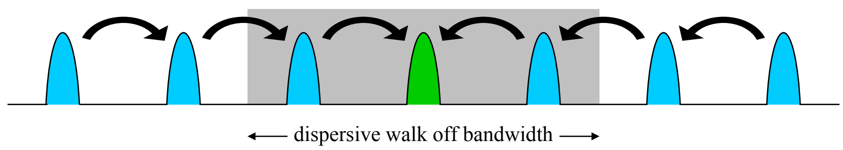
  - truncation at usual places means frequency response no longer flat, e.g. 0.5dB ripple
  - forces each element to be truncated much wider than usual – 15X more taps needed overall than for lumped CD compensation by FIR filter [Zhu, IEEE Phot. Tech. Lett., vol. 21, p. 292-294, 2009]
- Complementary filter pairs designed with different windows to be flat together reduces number of FIR taps by 2.1 [Zhu, IEEE Phot. Tech. Lett., vol. 21, p. 292-294, 2009]
- FIR filter designed by wavelet transform instead of truncation of Fourier tansform reduces number of taps by 4.7 [Goldfarb, OFC 2009, paper JThA47]

Compensation of fiber nonlinear effects

- Multichannel NL compensation possible by combining sampled signals from multiple receivers



- Set of coupled equations executed [Mateo, Optics Express, vol. 16, p. 16124-16137, 2008]
  - no need to construct field of whole optical bandwidth within processor
  - no need to know phases of receiver LOs w.r.t. one another
  - takes XPM into account but ignores FWM
  - short step size, 3km
- Cascading XPM mechanism seen, where modulation passed (as noise) from neighbour to neighbour
  - causes distant channels to impair one another, even beyond walk off bandwidth
  - therefore all channels must be included in backpropagation



47