TTI4A3
Komunikasi Akses Nirkabel

CLO 2 Konsep Kanal Wireless dan Rekayasa Sistem Radio

Minggu 8:
FADING MITIGATION: ADAPTIVE EQUALIZER

Outline

CLO 2

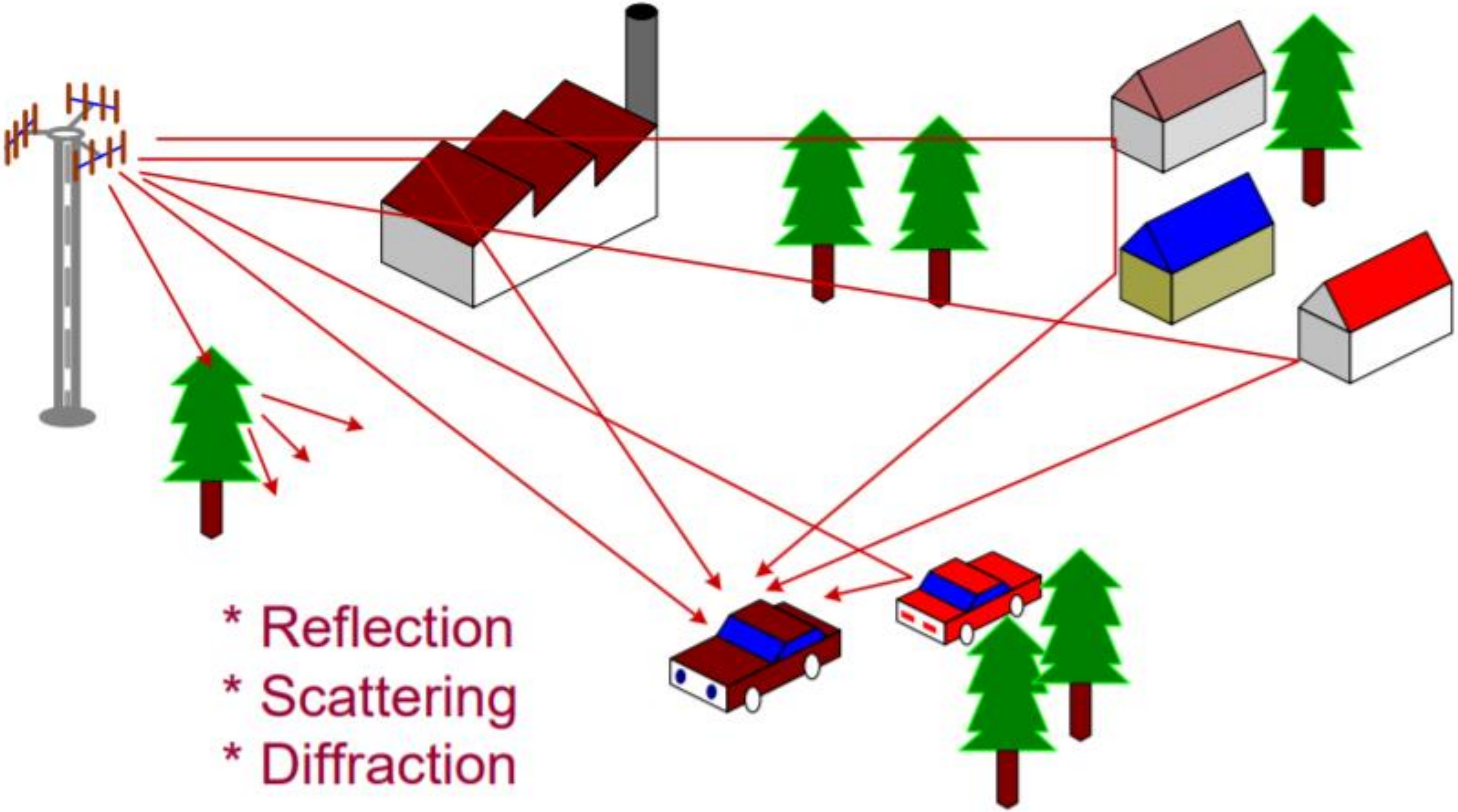
Channel Mobile & Concepts of Wireless Radio Engineering

Minggu 8 Mitigation for Small Scale Fading: Equalizer

- Filosofi Shannon limit
- Elemen-elemen Utama Siskomsel,
- Teknik Multiple Access

Review Small Scale Fading

Multipath channel in wireless communications.



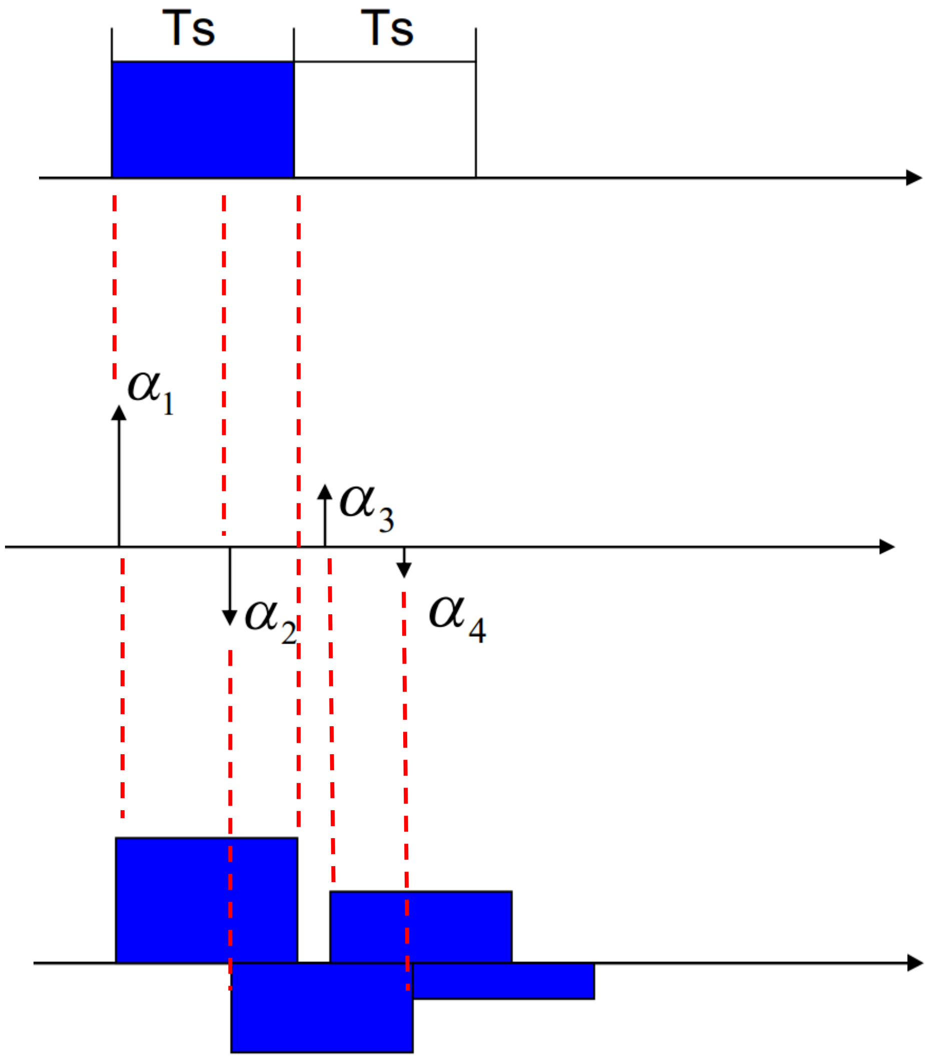
Pengertian Fading: Large Scale Vs Small Scale Fading

- **Definisi:** Fluktuasi daya di penerima

Transmitted signal: $s(t)$

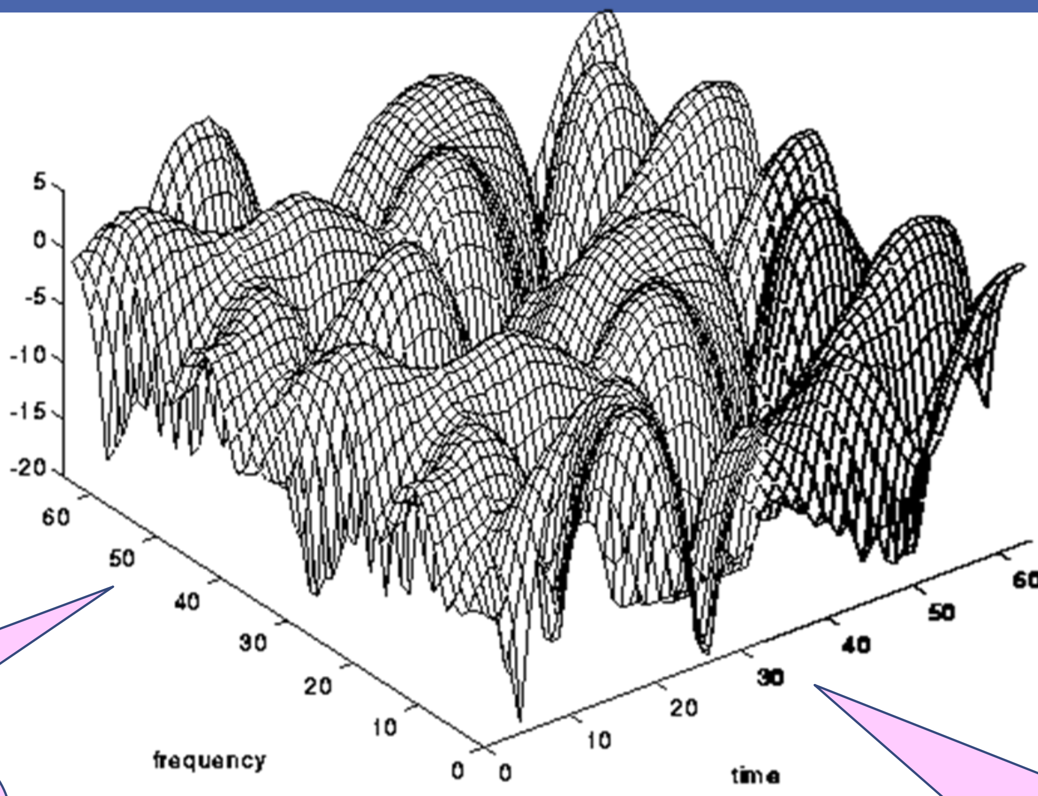
Channel model: $h(t) = \sum_k \alpha_k \delta(t - \tau_k)$

Received signal: $r(t) = s(t) * h(t) = \sum_k \alpha_k s(t - \tau_k)$



Karakterisasi Kanal Multipath

Fluktuasi dalam domain frekuensi



Fluktuasi dalam domain waktu

– ***Multipath* menyebabkan *Time spreading* sinyal →** Akibat sinyal datang dengan delay yang berbeda-beda, dianalisis dengan ***Delay Spread Model***

– ***Mobilitas user* menyebabkan *Time varying of channel* →** akibat pergerakan, dianalisis dengan ***Time Varying Model***

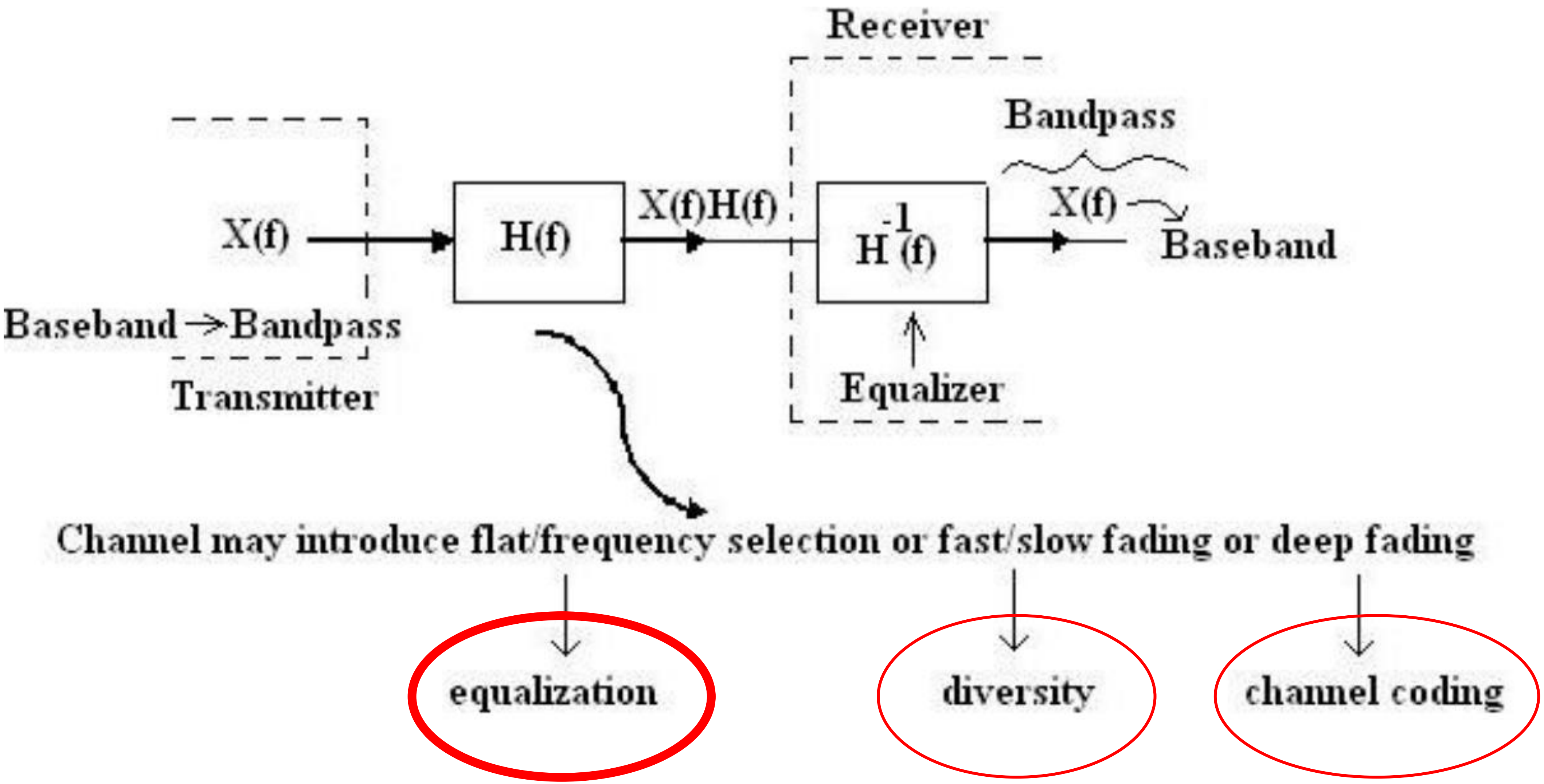
Karakteristik Kanal

- The properties of mobile radio channels:
  - Multipath fading → time dispersion, ISI
  - Doppler spread → dynamical fluctuation
- Mobile communication systems require signal processing techniques that improve the link performance in hostile mobile radio environments
- 3 (tiga) teknik populer:
  - Equalization: mengatasi ISI
  - Diversity: mengatasi fading sepanjang waktu
  - Channel coding: deteksi dan koreksi error

These effects have a strong negative impact on the bit error rate of any modulation

Dapat diimplementasikan secara terpisah atau bersama-sama

3 Teknik Mengatasi Fading



mengatasi fluktuasi dalam domain frekuensi

mengatasi fluktuasi dalam domain waktu

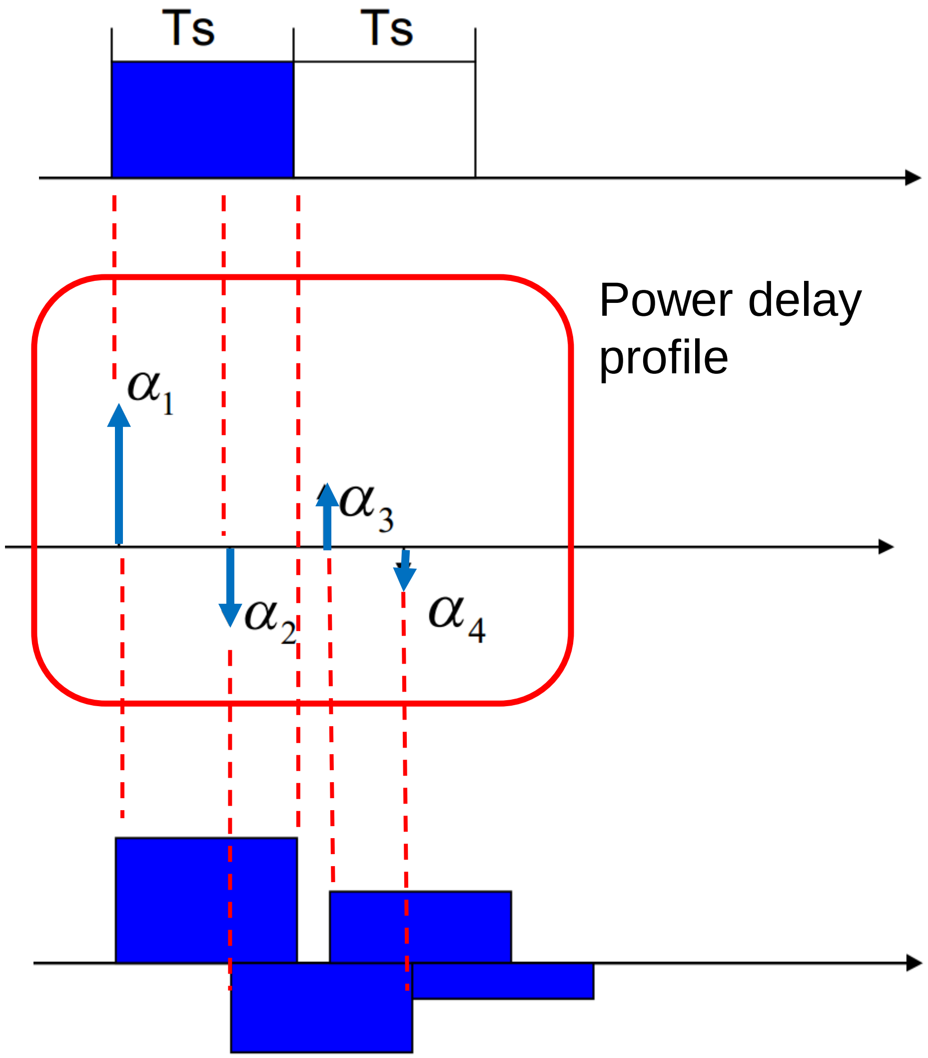
mengatasi fading yang terlalu dalam → koreksi error

Pengaruh kanal multipath terhadap sinyal penerimaan

Transmitted signal: $s(t)$

Channel model: $h(t) = \sum_k \alpha_k \delta(t - \tau_k)$

Received signal: $r(t) = s(t) * h(t) = \sum_k \alpha_k s(t - \tau_k)$



Teknik #1: Ekualisasi Kanal

- Jika bandwidth transmisi melebihi bandwidth kanal (*coherence bandwidth*), maka terjadi ISI
- Ekualisasi kanal bertujuan untuk mengkompensasi intersymbol interference (ISI) yang dihasilkan dari kanal multipath.
- Equalizer harus bersifat adaptif terhadap kanal yang bersifat berubah terhadap waktu

Mengatasi ISI

Operasi signal processing untuk meminimalkan ISI

Mampu melakukan tracking kanal yang berubah secara adaptif

Teknik #2: Teknik Diversitas

- Digunakan untuk mengurangi kedalaman dan durasi fading yang dialami receiver pada flat fading (narrowband) channel. Tanpa meningkatkan daya transmisi atau bandwidth.
- Dapat digunakan di sisi transmisi (Tx) atau sisi penerima (Rx)
- Tipe-tipe diversitas :
  - Diversitas polarisasi antenna
  - Diversitas frekuensi
  - Diversitas waktu. Sebagai contoh: RAKE receiver yang digunakan di CDMA
  - Spasial Diversity → Paling terkenal dan paling umum digunakan. Termasuk dalam spasial diversity adalah konsep MIMO

Teknik #3: Channel Coding

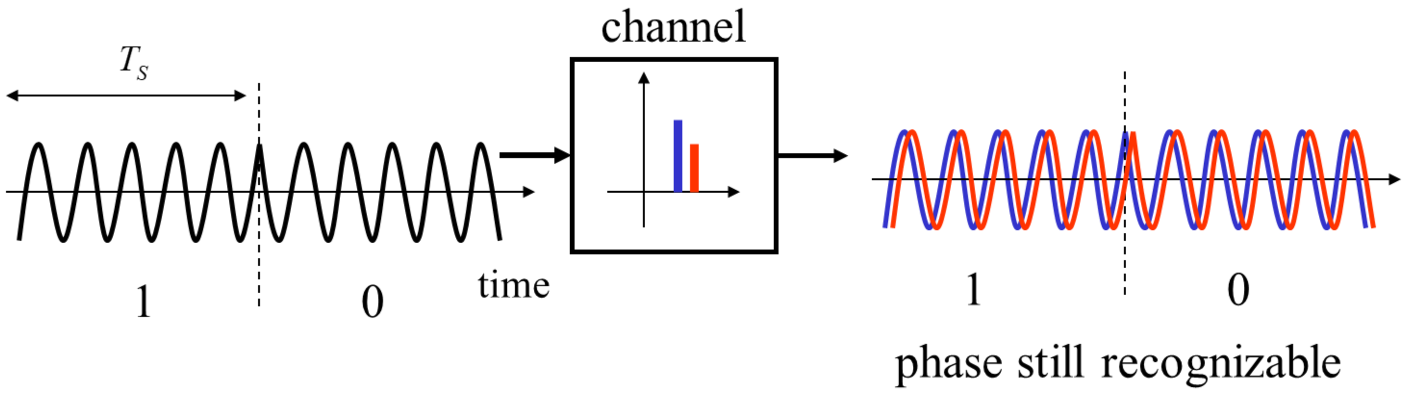
- Channel coding untuk mengatasi fading yang terlalu dalam, untuk deteksi dan koreksi error → dilakukan dengan menambahkan bit-bit redundant pada pesan yang ditransmisikan untuk meningkatkan performance.
  - Pada baseband, channel coder memetakan pesan digital pada pesan spesifik lain dengan jumlah bit yang lebih besar dari jumlah bit pesan asli.
  - Setelah pesan informasi dikodekan, baru dimodulasikan dan ditrasnmisikan pada kanal wireless.
- Proses decoding dilakukan setelah proses demodulasi, sebagai post detection technique
- Ada 2 (dua) kelas besar channel codig
  - block codes
  - convolutional codes.
- Channel coding umumnya diperlakukan terpisah dari tipe modulasi yang dipakai. Tetapi saat ini berkembang skema coding dan modulasi yang didesain secara bersama yang bertujuan untuk mencapai coding gain yang tinggi tanpa memperlebar bandwidth transmisi.

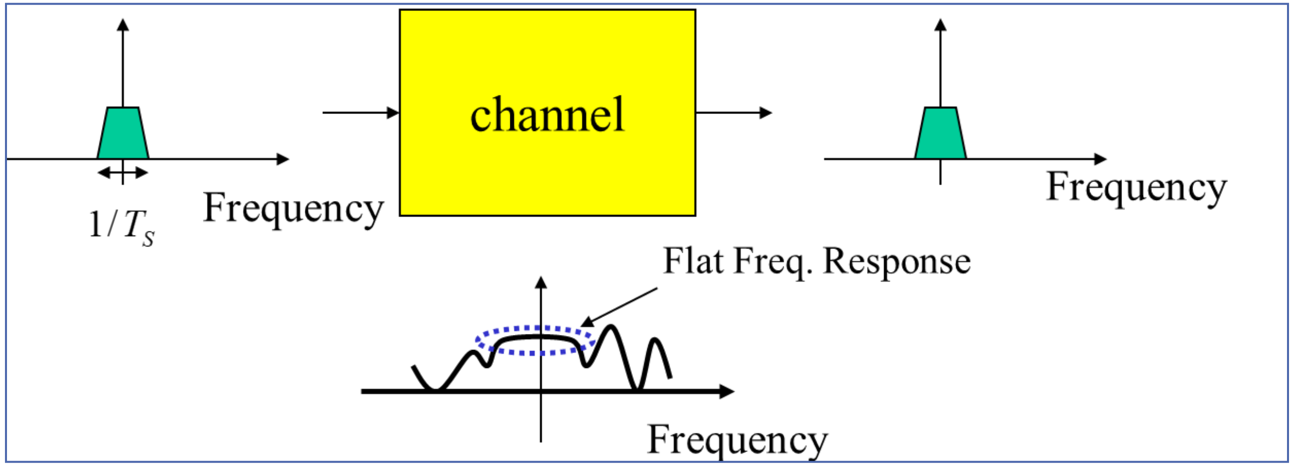
Adaptive Equalizer

Flat Fading Vs Frequency Selective Fading

Jika durasi symbol >> time delay spread → Hampir tidak ada Inter Symbol Interference (ISI).

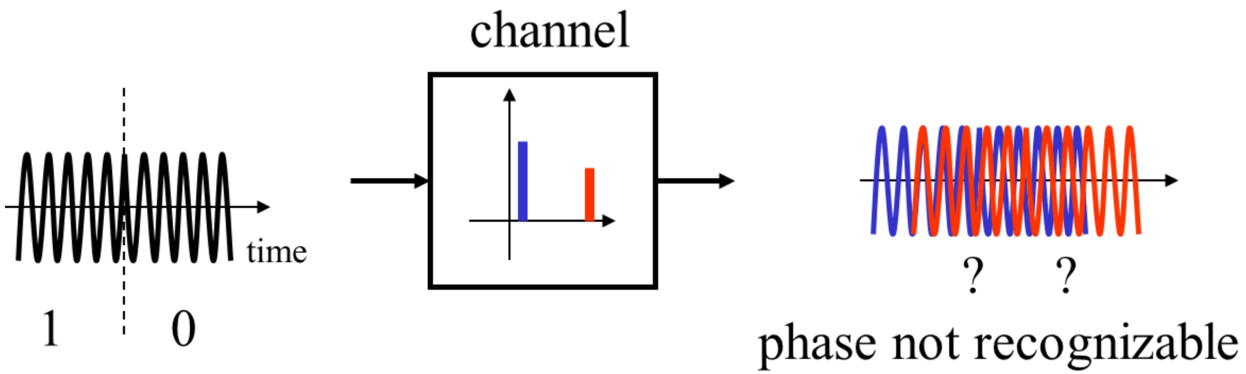
→ FLAT Fading



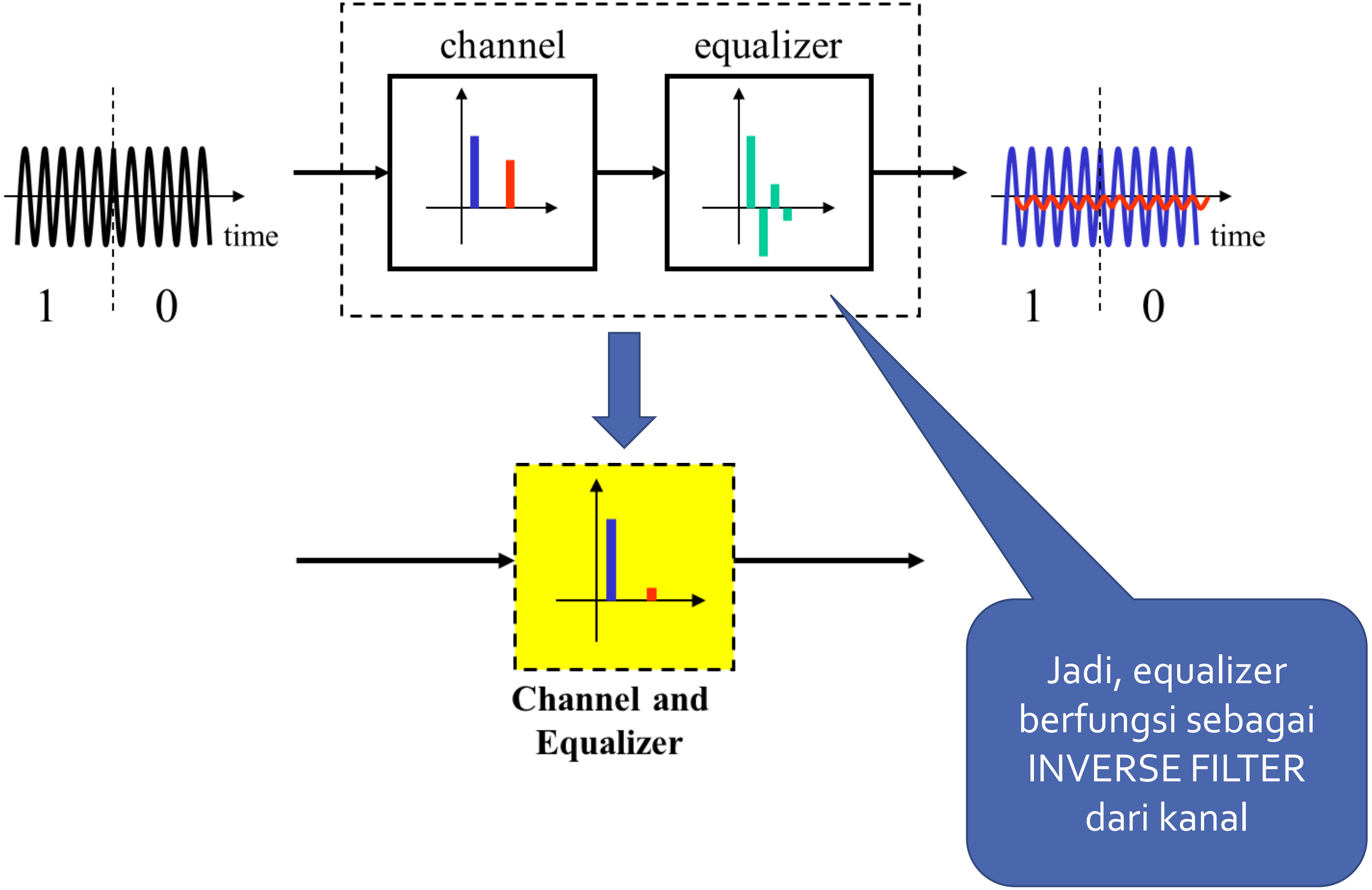


Jika durasi symbol ~ time delay spread
→ Terjadi Inter Symbol Interference (ISI).

→ FREQUENCY SELECTIVE Fading



Salah satu solusi *Frequency Selective Fading*: Equalizer

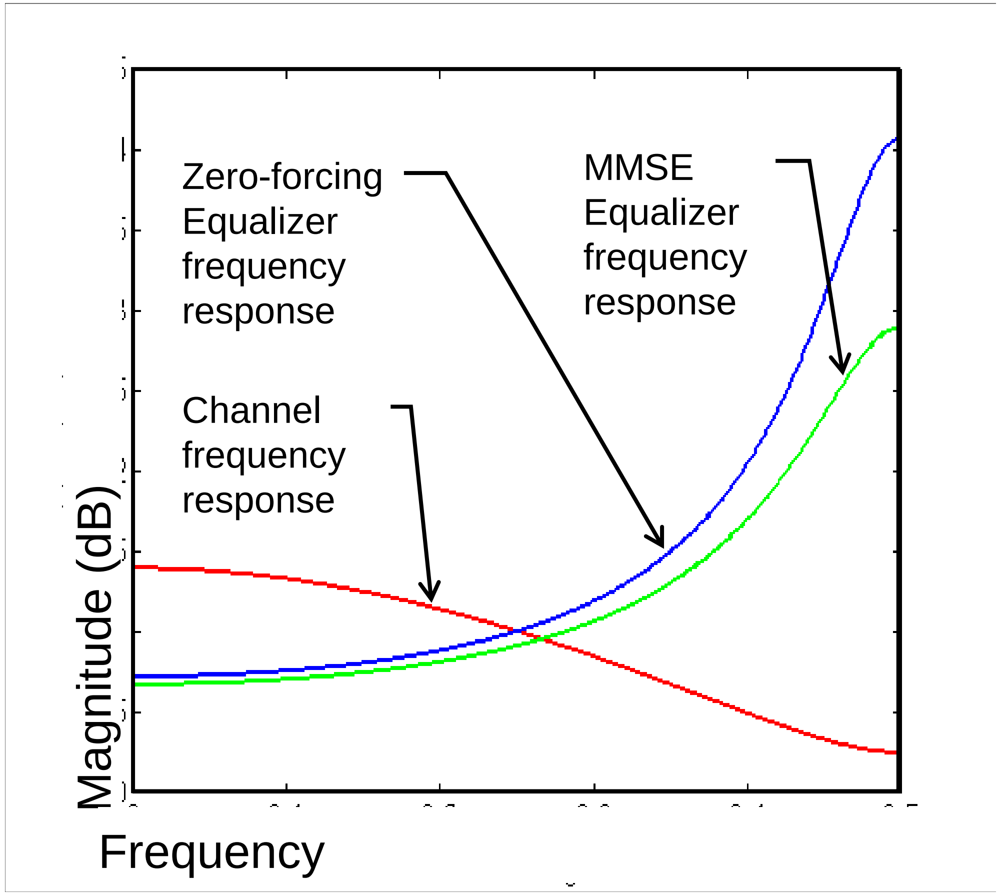


Problem ekualisasi kanal:

- Membutuhkan data latih / training bit (pemborosan bandwidth)
- Jika tidak receiver mengetahui kondisi kanal (*blind*), akan sangat 'mahal' karena komputasi yang berat
- Tetap ada problem jika kanal bersifat berubah terhadap waktu (*time varying*)

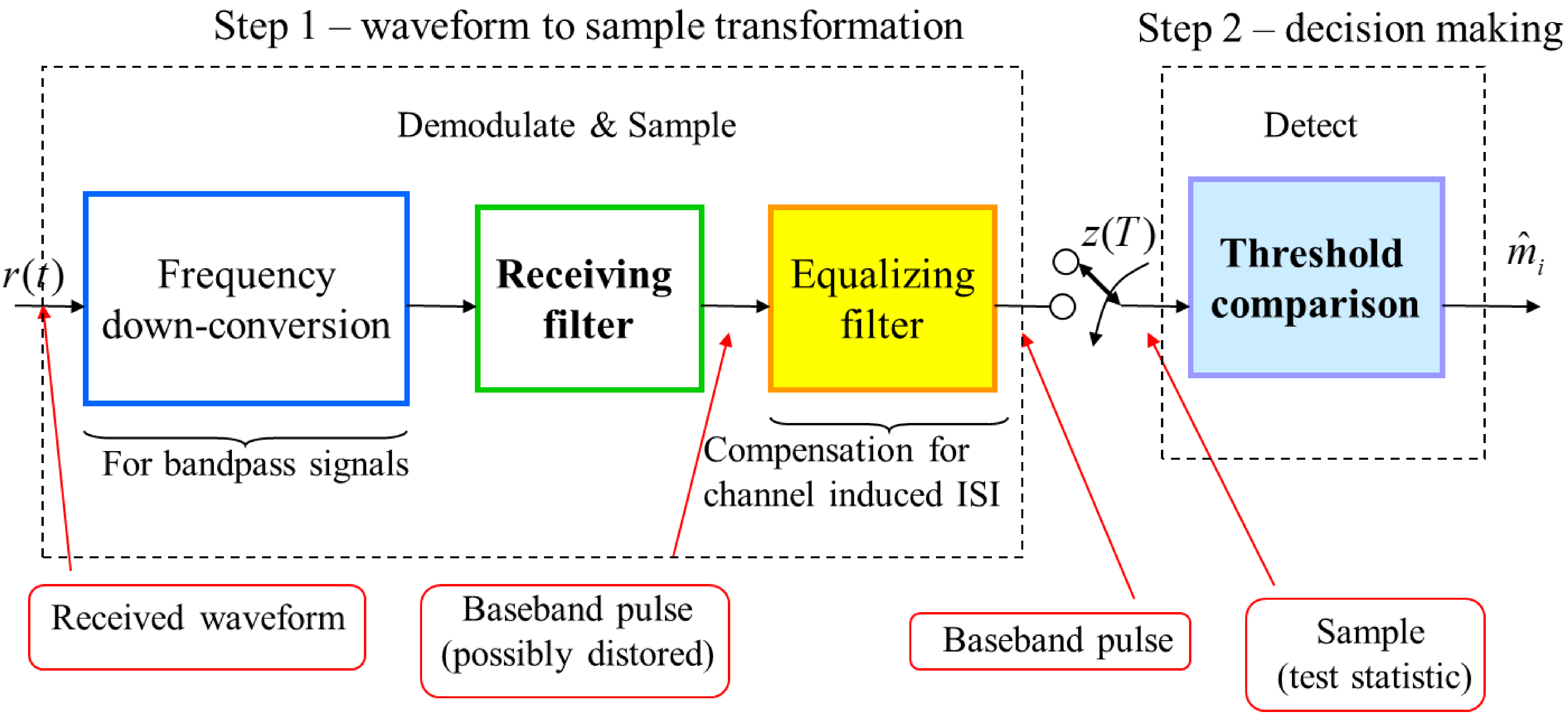
Combat ISI with Equalization

- ***Problem:*** Channel frequency response is not flat
- ***Solution:*** Use equalizer to flatten channel frequency response



- **Zero-forcing equalizer**
  - Inverts channel
  - Flattens frequency response
  - Amplifies noise
- **Minimum mean squared error (MMSE) equalizer**
  - Optimizes trade-off between noise amplification and ISI
- **Decision-feedback equalizer**
  - Increases complexity
  - Propagates error

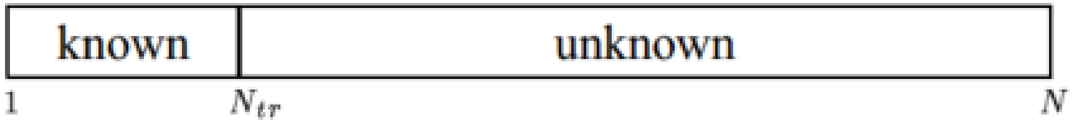
Prinsip Kerja Equalizer



Mode Operasi Adaptive Equalizer

First Stage - TRAINING → **Second Stage - TRACKING**

- Deretan training bit yang diketahui dikirimkan transmitter, agar equalizer di receiver dapat membuat setting yang tepat
- Training sequence didesain agar equalizer dapat menentukan koefisien filter yang tepat pada kemungkinan kondisi kanal terburuk
- Equalizer membutuhkan retraining periodic untuk menjaga ISI cancellation yang efektif → Rentang waktu tergantung dari algoritma equalizer, struktur equalizer, dan durasi fading



- Segera setelah training bit, data user kemudian dikirim
- Setelah data user diterima, algoritma adaptif equalizer akan melakukan tracking perubahan kanal dan mengubah karakteristik filter sepanjang waktu komunikasi
- TDMA wireless system cocok menggunakan equalizer
  - Data diletakkan dalam fixed-length time blocks
  - Training sequence diletakkan di awal blocks

After the demodulation, the I- and Q-component of the continuous-time signal are sampled at the symbol rate and combined into a complex-valued measurement. Consider the received sampled data sequence $\{y(t)\}_{t=1}^{N}$[1], generated by transmission of one data *burst*, $\{d(t)\}_{t=1}^{N}$, over a time-variant and time-dispersive channel. By regarding the entire channel from the symbol sequence $d(t)$ to the sampled measurements $y(t)$ as a discrete-time system, it is possible to model the channel as

$$y(t) = H_t(q^{-1})d(t) + v(t) = h_0^t d(t) + \sum_{k=1}^{m} h_k^t d(t-k) + v(t) \quad t = 1, \ldots, N \qquad (1)$$

$\{h_k^t\}_{k=0}^{m}$ is the complex-valued, time-varying impulse response of the equivalent discrete-time channel. $v(t)$ is measurement noise of zero mean, which is assumed to be independent of the symbol stream. We also make the assumption that the input stream is white, i.e. $E[d(t)d(\tau)] = 0$ for $t \neq \tau$.

Now, if the input $d(t)$ is to be recovered from $y(t)$, it is evident from the right hand side of (1) that the middle term, the intersymbol interference, and the noise $v(t)$ distorts the detection of $d(t)$ from $y(t)$. To extract the symbols, we use an equalizer. The design of the equalizer obviously depends on the impulse response of the channel. Since this is not only unknown, but also time-variant, an adaptive equalizer has to be used.

The problem of designing an equalizer is complicated by the fact that the possible regressors $\{d(t)\}$ are unknown as well. We cannot use adaptive modelling directly, since we do not know the input to the adaptive filter, and we cannot directly use inverse adaptive modelling since we do not know the desired signal. To cope with this problem, the data is transmitted in bursts of length N. A part of each transmitted burst is known at the receiver. This part, the so-called *training sequence*, can be located anywhere in the burst. For the rest of this article however, the training sequence is supposed to constitute the first N_{tr} symbols in the transmitted sequence. See Figure 6.

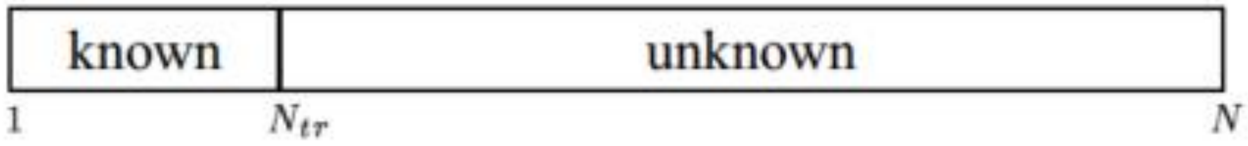


Figure 6: *Organization of the bursts*

It is thus natural to partition the adaptation procedure into two modes:

1. Learning-directed mode, $\{d(t), y(t)\}_{t=1}^{N_{tr}}$: The training sequence $\{d(t)\}_{t=1}^{N_{tr}}$ is utilized to initialize equalizer parameters.

2. Decision-directed mode, $\{\bar{d}(t), y(t)\}_{t=N_{tr}+1}^{N}$: Detected symbols, $\{\bar{d}(t)\}_{N_{tr}+1}^{N}$, are used as substitutes for $\{d(t)\}_{t=N_{tr}+1}^{N}$. The adaptation of the filter parameters is then based on decisioned data.

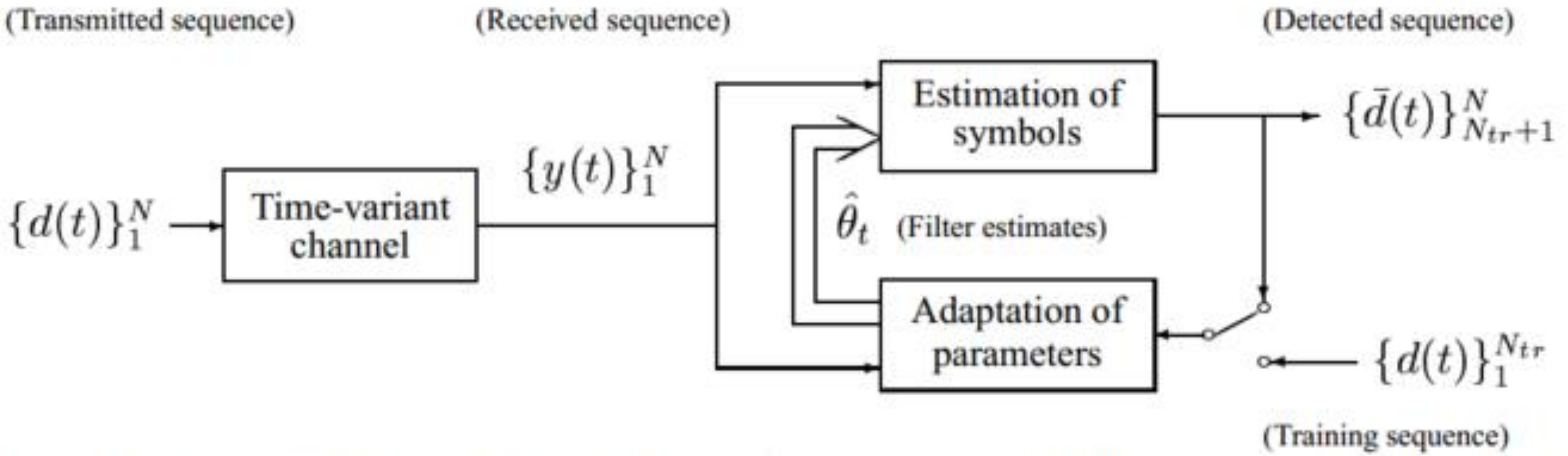
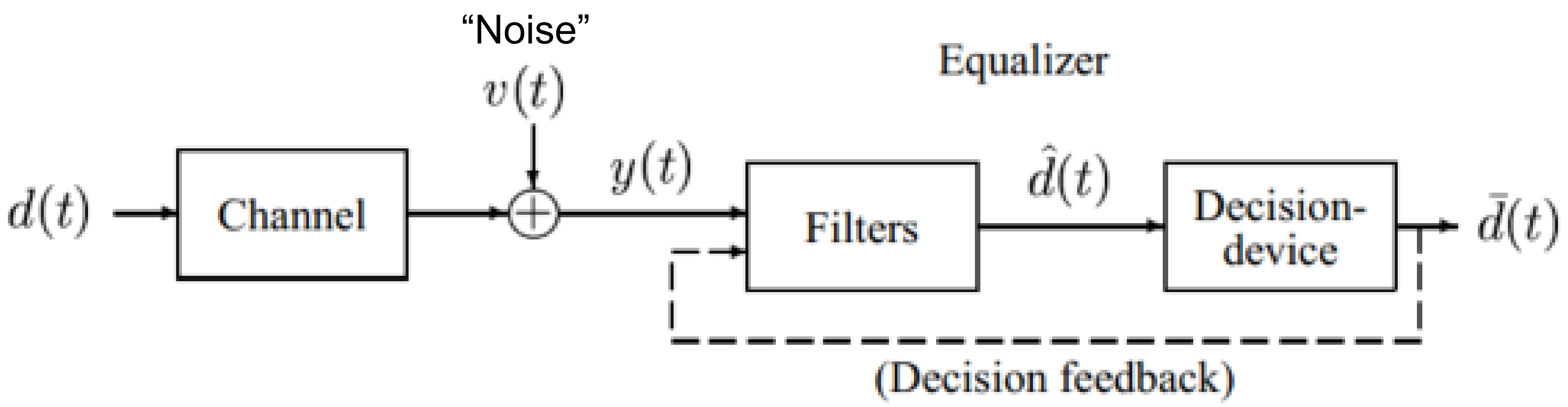


Figure 7: *A transmitted sequence of data $\{d(t)\}$, propagating through a time-variant channel, yields a received sampled sequence $\{y(t)\}$. The training sequence and corresponding data in the received sequence are used to initialize the adaptation. In decision-directed mode, the transmitted (unknown) symbols, $d(t)$, are replaced by decisions $\bar{d}(t)$ and adaptation of filter parameters works in tandem with a symbol estimator.*

Desain Equalizer



<u>Obyektif:</u>

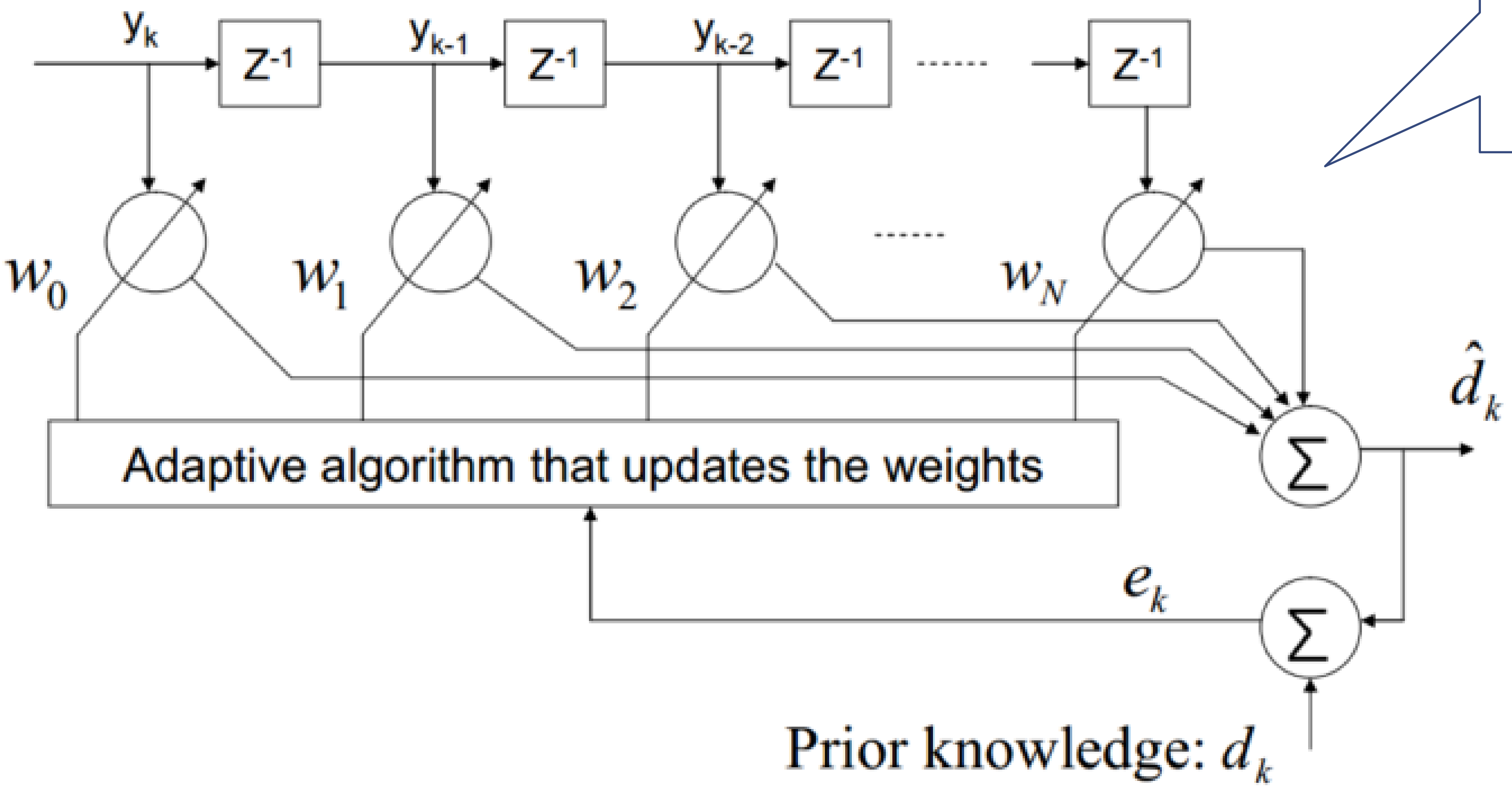
Minimize error $\bar{d}(t) \neq d(t)$.

"dengan fungsi cost tertentu"

<u>Sebagai contoh:</u>

algoritma *Least Mean Square* (LMS) dapat digunakan sebagai fungsi cost

Struktur Umum Equalizer



A transversal filter with
- N delay elements
- N+1 taps
- N+1 tunable complex multipliers
- N+1 weights:

- Bobot diupdate secara kontinyu oleh algoritma adaptive (berdasar sample by sample, atau blok demi blok data)
- Adaptive algoritma dikontrol oleh sinyal error

Contoh Proses Iteratif berbasis LMS

- **Iterative operation based on LMS**

  New weights = Previous weights + (constant) x (Previous error) x (Current input vector)

  Where

  Previous error = Previous desired output — Previous actual output

  This process is repeated rapidly in a programming loop while the equalizer attempts to converge

  Upon reaching convergence, the adaptive algorithm freezes the filter weights until the error signal exceeds an acceptable level or until a new training sequence is sent.

Klasifikasi Equalizer

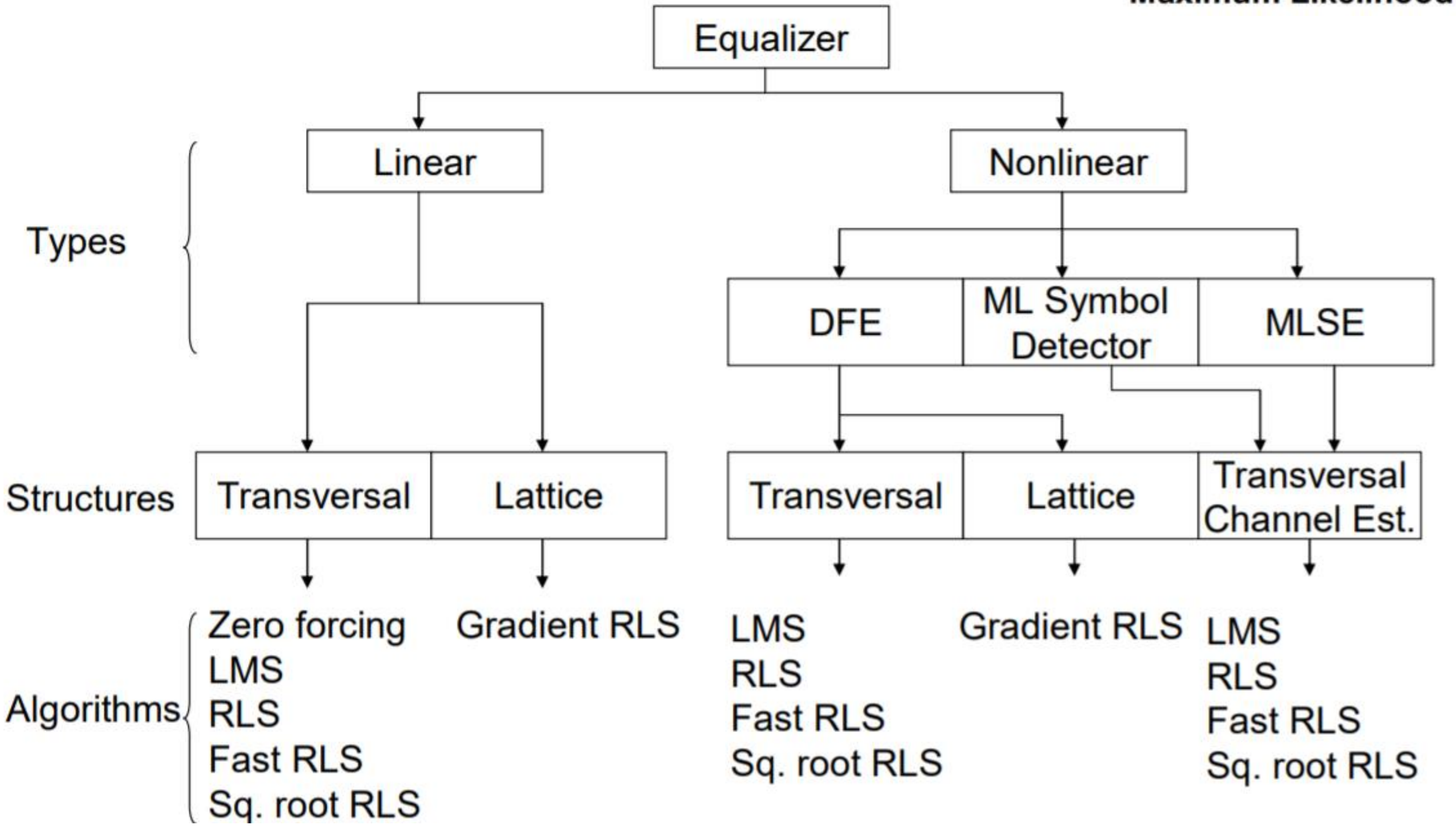
- **Equalization techniques can be subdivided into two general categories:**
  - **linear equalization**
    - *The output of the decision maker is not used in the feedback path to adapt the equalizer.*
  - **nonlinear equalization**
    - *The output of the decision maker is used in the feedback path to adapt the equalizer.*

- **Many filter structures are used to implement linear and nonlinear equalizers**

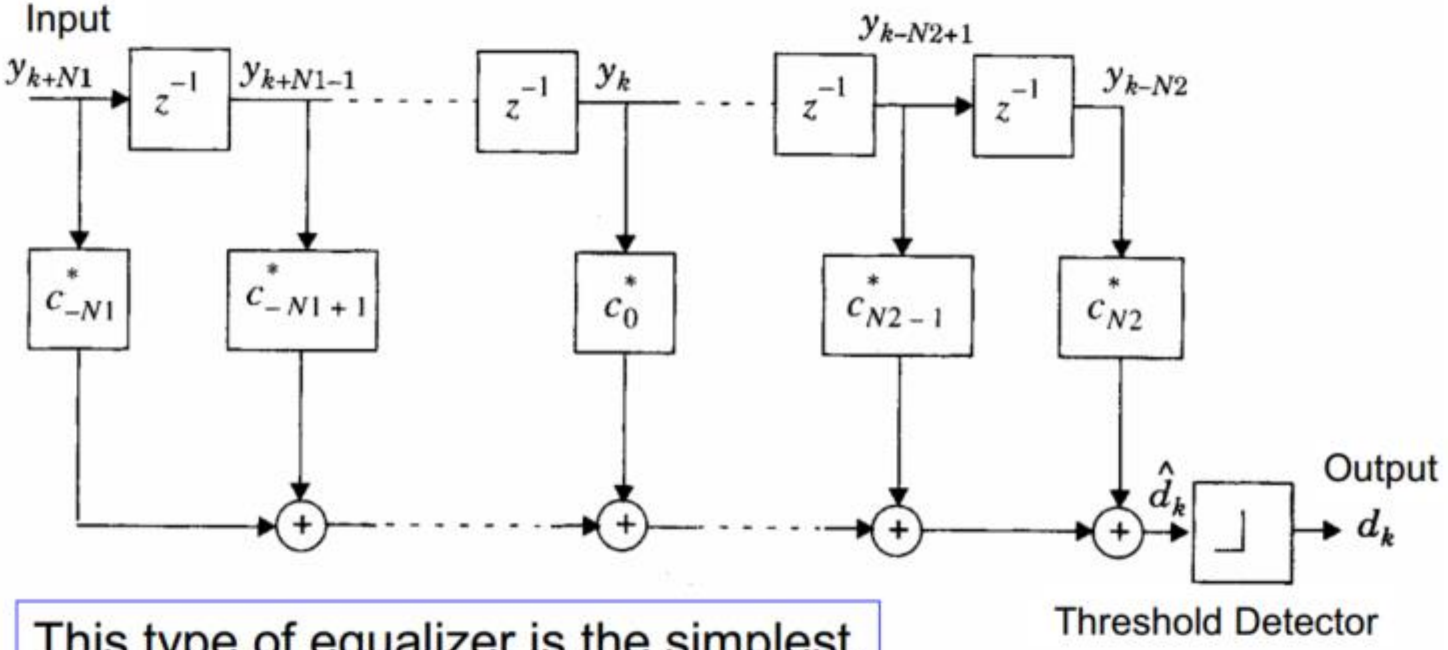
- **For each structure, there are numerous algorithms used to adapt the equalizer.**

Klasifikasi Equalizer

Decision Feedback Equalization (DFE)
Maximum Likelihood Symbol Detection
Maximum Likelihood Sequence Estimation (MLSE)



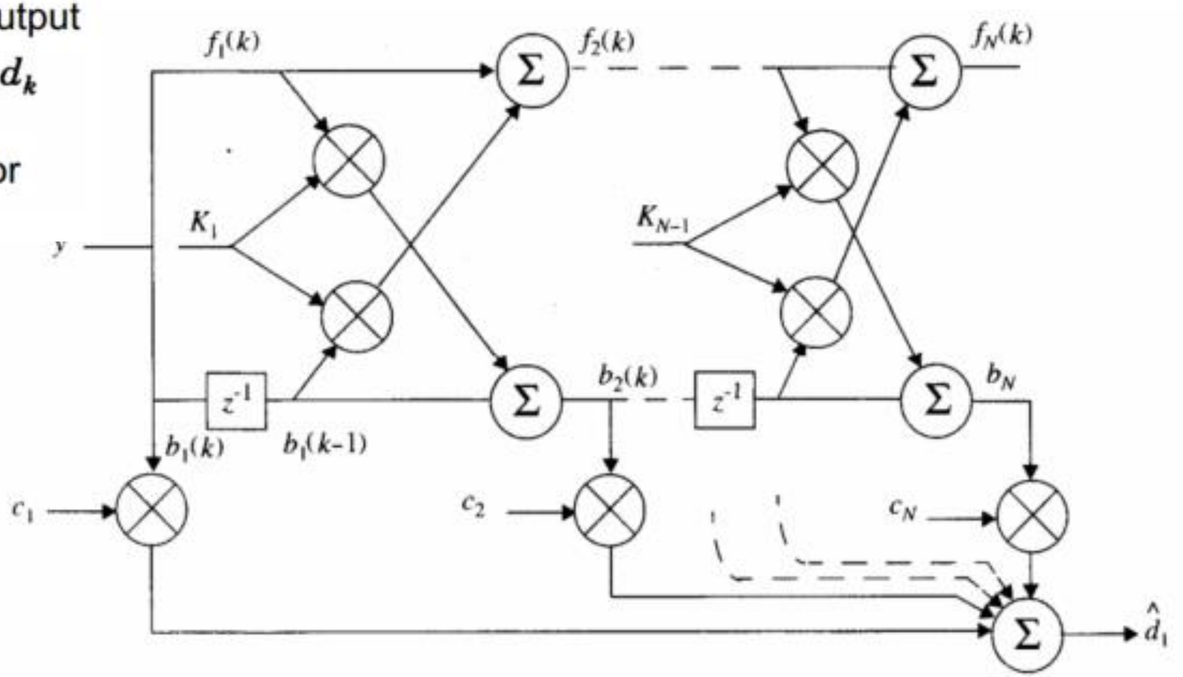
Linear Equalizer: Struktur Linear Transversal & Lattice



This type of equalizer is the simplest.

"Linear Transversal Equalizer (LTE)"

"Lattice Filter Implementation"



Numerical stable, faster convergence, Complicated

Non-linear Equalizer

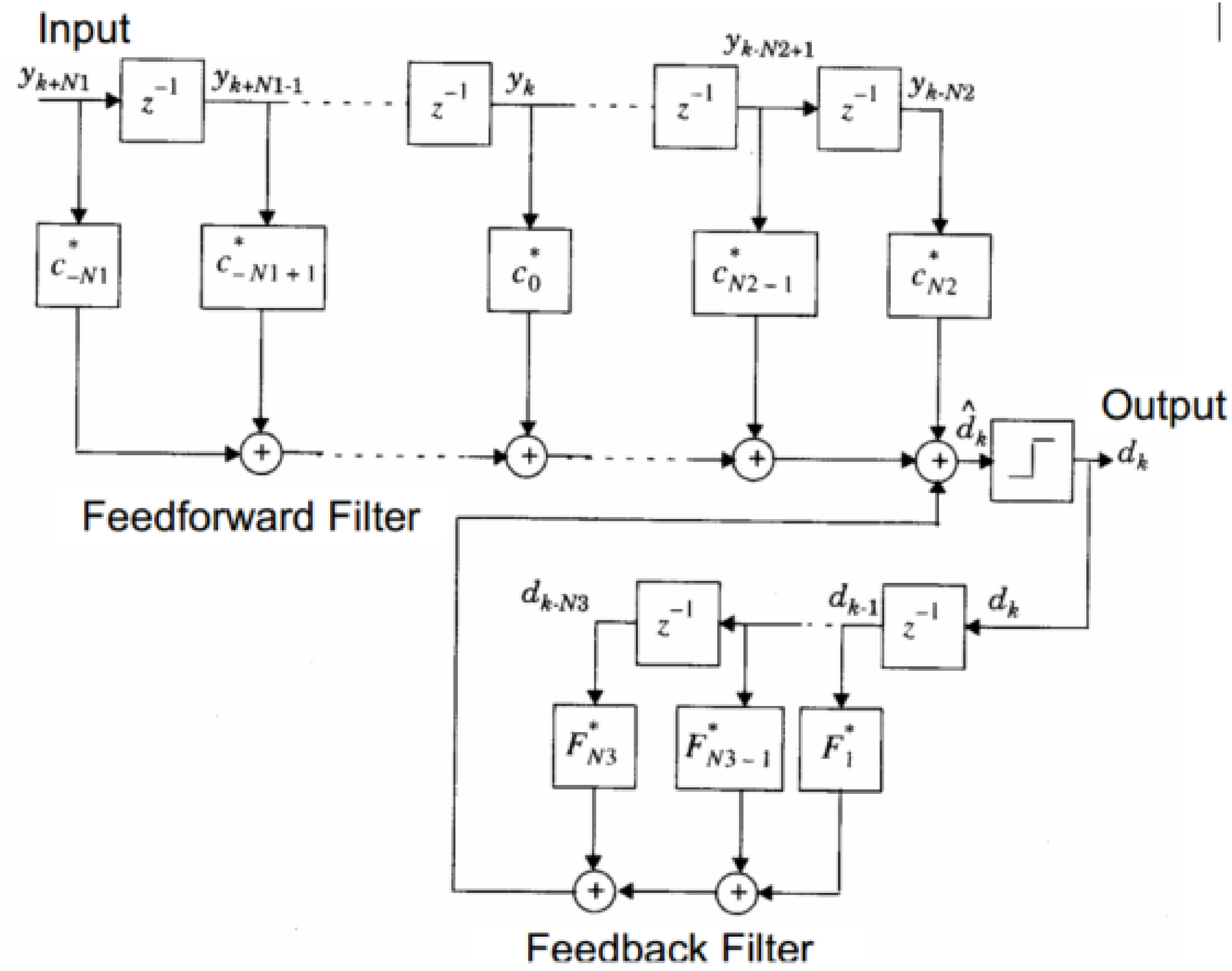
- Linear equalizers do not perform well on channels which have deep spectral nulls in the passband.

  In an attempt to compensate for the distortion, the linear equalizer places too much gain in the vicinity of the spectral null, thereby enhancing the noise present in those frequencies.

- Nonlinear equalizers are used in applications where the channel distortion is too severe for a linear equalizer to handle.

- Three very effective nonlinear equalizer
  - Decision Feedback Equalization (DFE)
  - Maximum Likelihood Symbol Detection
  - Maximum Likelihood Sequence Estimation (MLSE)

Decision Feedback Equalization (DFE)



Basic idea:
once an information symbol has been detected, the ISI that it induces on future symbols can be estimated and subtracted out before detection of subsequent symbols.

Comparison of Various Algorithms for Adaptive Equalization

Algorithm	Number of Multiply Operations	Advantages	Disadvantages
LMS Gradient DFE	$2N + 1$	Low computational complexity, simple program	Slow convergence, poor tracking
Kalman RLS	$2.5N^2 + 4.5N$	Fast convergence, good tracking ability	High computational complexity
FTF	$7N + 14$	Fast convergence, good tracking, low computational complexity	Complex programming, unstable (but can use rescue method)
Gradient Lattice	$13N - 8$	Stable, low computational complexity, flexible structure	Performance not as good as other RLS, complex programming
Gradient Lattice DFE	$13N_1 + 33N_2 - 36$	Low computational complexity	Complex programming
Fast Kalman DFE	$20N + 5$	Can be used for DFE, fast convergence and good tracking	Complex programming, computation not low, unstable
Square Root RLS DFE	$1.5N^2 + 6.5N$	Better numerical properties	High computational complexity

End