UNIVERSITY OF CALIFORNIA

Los Angeles

DIVERSITY COMBINING, ADAPTIVE ANTENNAS, AND EQUALIZATION FOR DIGITAL RADIO

A dissertation submitted in partial satisfaction of the requirements for the degree Doctor of Philosophy in Electrical Engineering

by

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1995

The dissertation of Eldad Perahia is approved.

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1995

Dedication

Contents

List of Figures

List of Abbreviations

AM	amplitude modulation
ARPA	advanced research projects agency
AWGN	additive white Gaussian noise
CCI	co-channel interference
DC	diversity combiner (single tap)
DFE	decision feedback equalizer
DPSK	differential phase shift keying
erf	error function
erfc	complementary error function
FM	frequency modulation
FSK	frequency shift keying
iid	independently identically distributed
ISI	intersymbol interference
LE	linear equalizer
LMS	least-mean-square
MC	multitap diversity combiner
NCFSK	non-coherent frequency shift keying
PAM	pulse amplitude modulation

PC	personal computer
PM	phase modulation
PSK	phase shift keying
QAM	quadrature amplitude modulation
RW	receiver weights
SINR	signal-to-interference-and-noise ratio
SNR	signal-to-noise ratio
UCLA	University of California, Los Angeles
WAMIS	wireless, adaptive, and mobile information systems
ZF	zero-forcing

List of Symbols

e^x, $\exp[x]$	exponential of x
$\sum$	summation
π	pi
$\det(\cdot)$	determinant
$\prod$	product
$E[\cdot]$	expected value
$\mathcal{L}^{-1}\{\cdot\}$	inverse Laplace transform
!	factorial
$\lvert \cdot \rvert$	absolute value
$\begin{pmatrix} x \\ y \end{pmatrix}$	$= \frac{x!}{(x-y)!y!}$ combination
$\sqrt{\cdot}$	square root
j	imaginary unit, $\sqrt{-1}$
χ^2	chi-squared distribution
$\int$	integral
$\Re[\cdot]$	real part of a complex value
$\Im[\cdot]$	imaginary part of a complex value
∞	infinity

sin	sine
cos	cosine
tan	tangent
c	speed of light
$(\cdot)^*$	complex conjugate
$(\cdot)^{-1}$	inverse
$(\cdot)^x$	raised to the power of x
$(\cdot)^T$	transpose
$(\cdot)^H$	complex conjugate transpose
dB	decibel
MHz	megahertz
$\frac{\partial}{\partial x}(y)$	partial derivative of y with respect to x
$\mathbf{0}$	all zero matrix
$\mathbf{I}$	identity matrix
$\mathbf{1}$	all ones vector
$\min(x, y)$	minimum value between x and y
$\max(x, y)$	maximum value between x and y
P_e	probability of error
P_c	probability of a correct decision
$Prob\{\cdot\}$	probability
σ^2	variance

$p(\cdot)$	probability distribution
L	number of antennas or diversity order
λ	wavelength

Glossary

antenna array A set of antenna elements used in conjunction for transmission or reception.

beampattern The transmitting or receiving gain pattern of an antenna or antenna array.

beamforming Shaping the beampattern of an antenna array by adjusting complex weights on antenna elements.

capacity Maximum number of users a system can support.

cellular radio A system in which a service area is subdivided into smaller regions or *cells*. Radio frequencies are reused in each cell.

channel A specified frequency or time band for transmission and reception of radio signals.

co-channel interference (CCI) Interference caused by users competing for the same channel.

coherent demodulation The channel phase shift is used to demodulate the received signal.

decision device A device in which the estimate of the transmitted signal is quantized to the nearest information symbol. If the output is not identical to the transmitted information symbol, an error has been made.

decision feedback equalizer (DFE) A combination of a feedforward and a feedback filter. The feedforward filter is a linear equalizer. The output is passed through a decision device and is fed back through another linear equalizer, the feedback filter.

delay spread The time extent between the arrival of the first delayed response and the last delayed response of a multipath signal.

demodulation The processing of the channel-corrupted transmitted waveform to a single number representing an estimate of the transmitted data symbol.

diversity combiner (DC) A device which receives multiple copies of the transmitted information, and then combines them to mitigate the effects of channel fades.

equal-gain combining Applying a weighting factor to the received signal on each channel and summing all the signals. The weighting factors are equal to unity.

equalizer A filter used to deal with ISI.

fade, channel A large attenuation of the transmitted signal causing errors in reception.

flat-fading A channel distortion causing equal attenuation over the entire signaling frequency band.

frequency hopping Users transmit in a single sub-frequency band and periodically switch to a different frequency band.

frequency selective channel A channel which severely distorts the transmitted signal since the bandwidth the channel is small in comparison to the bandwidth of the transmitted signal.

impulse response A function which describes the output response of a system to an input of an impulse.

interference Undesired signals in the communication environment.

intersymbol interference (ISI) Delayed multipath signals interfering with subsequent transmitted data symbols.

least-mean-square (LMS) An algorithm to iteratively adjust weights based on the steepest descent method.

linear equalizer (LE) A filter which is comprised of the weighted sum of the received signal and past received signals.

line-of-sight A direct unobstructed path between a transmitter and receiver.

maximal-ratio combining Applying a weighting factor to the received signal on each channel and summing all the signals. The weighting factors are equal to the magnitude of the channel.

mobile radio A general class of radio communication in an outdoor environment.

modulation The conversion of digital data into waveforms that are compatible with the characteristics of the channel.

multipath The large set of transmission paths taken by the radio waves to the receiver due to scattering by the terrain.

multitap diversity combiner (MC) A receiver which is comprised of a linear equalizer on each antenna branch. The output of the equalizer for each channel is summed.

non-coherent demodulation The channel phase shift is ignored in demodulating the received signal.

omnidirectional transmission Transmitting with an antenna or antenna array with equal gain in all directions.

peer-to-peer A direct link between a transmitting a receiver pair of users, with no intermediaries.

ray tracing Channel modeling by computing the paths of the transmitted radio waves (rays) that arrive at the receiver.

tracking, channel Estimating the channel phase shift.

transceiver A transmitter and receiver unit.

Acknowledgments

I would like to express my genuine gratitude to my advisor Gregory J. Pottie for his guidance and assistance. Through his tireless efforts, Professor Pottie has created an environment at UCLA in which his students have interesting and challenging problems to investigate in the area of wireless digital communication system theory.

I would like to thank my fellow classmate, Charles C. Wang, for assistance with Chapter 4 on power control. His research will entail a detailed analysis on power control and bit allocation for the WAMIS project. In addition, I would also like to thank my fellow classmate, Christopher J. Hansen, for his insightful advice and discussions with research related to the WAMIS project. His research will include an investigation on channel probing for the WAMIS project.

I would also like to thank my friends at MARK Resources Inc. for their continual assistance and support.

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E. Perahia and G. J. Pottie, "On Diversity Combining for Correlated Slowly Flat-Fading Rayleigh Channels," *Proceedings of the IEEE International Conference on Communications,* vol. 1, pp. 342–346, 1994.

Abstract of the Dissertation

Diversity Combining, Adaptive Antennas, and Equalization for Digital Radio

by

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Doctor of Philosophy in Electrical Engineering

University of California, Los Angeles, 1995

Professor Gregory J. Pottie, Chair

We consider the problem of high performance wireless communications system incorporating multiple antennas. In the first part of this research, we present exact and approximate analytical expressions for the error probabilities for communications systems with coherent and non-coherent modulation schemes. We model our system with correlated, flat-fading Rayleigh channels with additive white Gaussian noise. Our solutions hold for M-ary signaling, L-order diversity, and a correlation coefficient of ρ.

In the second part of this research, adaptive antenna arrays and equalizers for use in transceivers for peer-to-peer wireless communications in local area networks are examined. A new adaptive transmission algorithm which uses receiver diversity combiner weights for transmission is developed. This method provides large gain with a simple design suited for a distributed network. The effects of network topology

and channelization on convergence are discussed. Joint adaptation of the antenna arrays with equalizers and power control is examined.

Chapter 1

Introduction

The wireless communications industry has experienced explosive growth in recent years. In our fast paced world, cordless phones, pagers, mobile phones, and personal phones have provided people freedom and convenience to communicate “on the go”. The decreased cost of wireless products has enabled widespread public use of wireless communications. Increased use has caused present cellular systems to reach their capacity. New digital technology in low-power personal communications transceivers (transmitter and receiver) is necessary to cope with the continually growing cellular radio industry. Moreover, changes in the regulatory environment portend a future where many organizations compete for local telephone service, placing wired and wireless communications in direct competition. To compete, the wireless industry must reduce costs and improve quality. Both objectives can be met with the sensible application of digital communications techniques.

The next generation in wireless communications will also include wireless computer networks transmitting voice and video as well as data. One motivation is the large cost and difficulty of rewiring offices for computer networks. In a wireless network, portable computers could access any peripherals in communications range, such as printers and disk drives. In addition, the peripherals would not need to be connected by wire to the network server.

These systems will be required to transmit data at high speeds with reliable performance, comparable to a wireline network. The transmission characteristics will be governed by the indoor radio environment. There will be short ranges between transmitter and receiver as compared to mobile communications. The systems must operate without line-of-sight, and they must adapt to slow changes. In addition, they must be able to contend with interference from other users in the network. This will allow communication between portable wireless computers in different rooms in a building.

The system design issues of adaptive, high bit-rate, wireless transceivers for the indoor local area network are significantly different than those of mobile communications. The short distances will greatly affect the transmission environment. We are not as limited by power consumption and size as in a mobile system, but require higher performance. Therefore we have access to a richer and more complicated set of communications techniques to deliver this performance.

Major multidisciplinary research programs in both low-power handheld transceivers and highly adaptive transceivers for indoor computer networks have been established at UCLA [60, 61]. Our research has focussed on antenna diversity combining techniques for implementation in the systems of these two programs. We have analyzed many algorithms and developed new combining methods. But our analysis must go beyond an analytical study. We must also consider the practicality of an implementation. In addition, we examine the interaction of other system components with antenna diversity combining.

The goal of the first research project was to develop a low-power personal communications transceiver for wireless digital data transmission. In addition to innovative hardware technologies to ensure a low-power solution, advanced systems techniques were designed to achieve robust performance in a multipath fading environment. A synchronous frequency-hopped system was selected instead of a significantly more complex wideband, direct sequence, spread-spectrum system which would result in much higher power consumption. However, fast hopping generally precludes the use of a coherent receiver [60]. Therefore a non-coherent binary frequency shift keying (NCFSK) modulation system was implemented which results in a performance loss of roughly 6 dB as compared to coherent modulation. However, in a synchronous system, hopping patterns are arranged to significantly reduce interference from other users. In addition, channel coding will also provide frequency and interferer diversity protection. Hopping over narrow frequency bands creates a flat-fading environment, eliminating the necessity of a power consuming adaptive equalizer. Two antennas provide diversity combining to reduce the effects of multipath fading. Adaptive power control was employed to minimize the required transmitted power.

The goal of the second research project is to develop a high-speed wireless transceiver for the application to nomadic computing. To achieve as high a data rate as possible requires the implementation of many advanced adaptive algorithms to reduce interference in a slowly changing environment. To conserve bandwidth, quadrature amplitude modulation (QAM) signaling will be employed. An adaptive modulator will allow variable signal set sizes from binary phase shift keying (PSK) and 4-QAM to 64-QAM depending on channel conditions. Slow frequency-hopping with a combination of interleaving and coding with orthogonal signaling in cells will be used to achieve frequency diversity and reduce co-channel interference (CCI). Antenna arrays will provide directional transmission to also reduce CCI. In addition, large performance gain is achieved by adaptive beamforming with antenna arrays on reception to null out interferers, and increased gain in the direction of the desired signal. Adaptive equalizers will mitigate the effects of intersymbol interference (ISI). Adaptive power control will reduce the require transmitted power level to also limit co-channel interference. All the adaptive algorithms will be implemented in a distributed fashion to allow peer-to-peer communication without a centralized basestation.

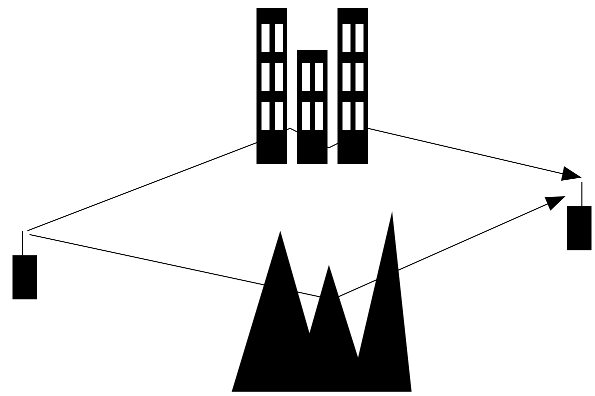
The remainder of this chapter will be devoted to a general introduction to digital radio communications and the inherent difficulties. We begin with a discussion the radio environment and sources of interference in Section 1.1. Section 1.2 illustrates a general communication system comparing digital and analog systems. Sections 1.3 provides a tutorial on diversity combining. An introduction to interference cancellation is given in Section 1.4. Lastly, in Section 1.5 we present an outline of the research, and how the work advances the present algorithms and technology.

1.1 The Radio Environment and Interference

Suppose information is to be transmitted via radio waves. Many sources create noise and interference, rendering reliable reception difficult. In this section we outline the various impairments, while in later sections we introduce techniques to reduce the error probability or increase the number of users who may share the common band of frequencies

The received signal is typically corrupted by random noise across the entire frequency band. This is modeled as an additive, white Gaussian random process or additive, white Gaussian noise (AWGN).

In the radio environment there are large variations in the signal level, known as fading. Transmission during a severe channel fade will cause a significant increase in the probability of error in reception. In the mobile radio environment, many factors will cause amplitude variations in the received signal. The transmitted signal will follow many paths to arrive at the receiver, termed multipath.



These multiple paths have random amplitudes, phases, and delays. Therefore we can describe the received signal, $r(t)$, as the sum of the signals on the n multiple paths. On each path, the transmitted signal, $u_m(t)$, will be attenuated by the channel attenuation $\alpha_n(t)$ and delayed by a propagation delay of $\tau_n(t)$. The transmitted signal will experience a phase shift proportional to the propagation delay. The received signal will also be corrupted by AWGN.

$$r(t) = \sum_n \alpha_n(t) e^{-j2\pi f_c \tau_n(t)} u_m[t - \tau_n(t)] + z(t) \qquad m = 1, \ldots, M \tag{1.1}$$

where

$\alpha_n(t)$	:	attenuation of the n^{th} path
$\tau_n(t)$	:	delay of the n^{th} path
f_c	:	center frequency
$u_m(t)$	:	transmitted signal, m^{th} of M size symbol set
$z(t)$	:	AWGN, i.i.d. , zero mean, $\sigma_z^2 = N_0$
$r(t)$	:	received signal

With random amplitudes and phases, signals can arrive on different paths with similar attenuation but completely out of phase. In these situations the multipath can add destructively resulting in the received signal being very small or even zero. Without any techniques to mitigate against the multipath fading, system performance is severely degraded. Over an AWGN channel, the probability of error drops exponentially with an increase in the signal-to-noise ration (SNR). However in a fading environment, the probability of error only decreases linearly with SNR. Thus, a huge power penalty must be paid. To illustrate, Figure 1.1 displays the variations in received power over a region caused by multipath. While the deep fade regions (in black) represent a small fraction of the total area, they determine the error rate of the system.

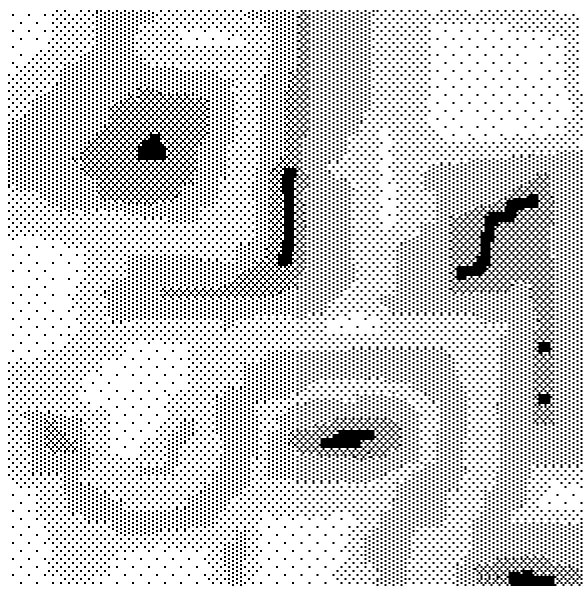


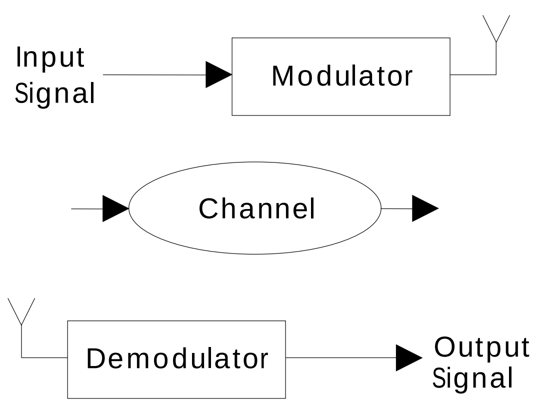
Figure 1.1: Multipath fading; signal amplitude is represented by shading; white indicates strong signal, black indicates a deep fade.

When the delays of the multiple paths are large (a dispersive channel), the delayed signal responses interfere with the reception of subsequent transmitted data symbols.

This type of interference is termed ISI. To compensate for ISI we use an equalizer.

Another form of interference is caused by communication in an environment in which many users are transmitting simultaneously on the same channel. Upon reception we must detect the desired signal from among all the transmitting users. The interference from other users is termed CCI.

1.2 Communications Systems



A radio communication system transfers information signals from a source to a user destination. The information signals may be generated from voice, video, or data. To be transmitted, the information must be converted into waveforms that correspond to the characteristics of the communication environment, or channel. This operation is termed modulation.

The distance between transmission and reception points may be small as in an indoor radio system or large as in a mobile radio system. Sources of degradation in the channel will vary in the different environments. Interference and noise in the communication environment will corrupt the received signal.

The task of the receiver is to recreate the original information from the received signal that has been corrupted after propagation through the channel. The demodulator is responsible for recovering the transmitted information from the received signal with minimal distortion. The type of modulation and demodulation schemes used will depend on the specific degradation caused by the channel.

The choice in modulation method also involves a decision between analog or digital communication. Voice and video signals are inherently analog. Thus, when transmitted with analog modulation they only require temporary frequency translations. With digital transmission, voice or video signals must be converted to a digital format with a source encoder. Data signals are inherently digital.

Examples of analog modulation methods are amplitude modulation (AM), frequency modulation (FM), and phase modulation (PM). To transmit digital information we must convert the data into analog waveforms that correspond to the characteristics of the communication environment, or channel. We take a block of $k = \log_2 M$ binary digits at a time and map it to one of $M = 2^k$ waveforms [51]. Upon reception, we must extract the binary data from the waveform.

There are many methods for mapping binary data into waveforms. We can map the binary information into a set of discrete amplitudes, phases, or frequencies [51]. Mapping into a set of discrete amplitudes yields pulse-amplitude modulation (PAM).

Mapping into a set of discrete phases results in phase shift keying (PSK). Mapping into a special set of combined amplitude and phases gives quadrature amplitude modulation (QAM). And mapping into a discrete set of frequencies produces frequency shift keying (FSK). We will also encounter a variation of PSK which is differential PSK (DPSK).

First generation cellular systems were implemented in analog. Analog systems constitute a much more mature technology, and consequently this was a natural first choice. However, present cellular systems are reaching full capacity. Advancement in low bit-rate speech coding and the ability to contend with the increased bandwidth requirements make a digital solution a viable option. Second generation digital systems will provide significant improvement in both capacity and quality of service.

Digital transmission has many advantages over analog channels. Once the analog signal is encoded to a digital sequence, residual transmission loss variation of analog source signals is reduced. In addition, with digital data complete signal regeneration is possible. Many advanced algorithms are available with digital communication since digital data is being manipulated instead of the actual analog signal. Some of the techniques that may be used to mitigate noise, interference, and fading include antenna arrays, channel coding, equalization, and power control. Therefore we can achieve better error performance and higher system capacity.

1.3 Diversity Combining

If we provide the receiver with multiple copies of the transmitted signal each with different channel attenuation, the probability that all copies will be severely faded is reduced. This is possible since the channel fades are random. Replicas of the transmitted signal are created by several means. We can transmit the signal multiple times: time diversity. We can also transmit the signal over multiple frequencies: frequency diversity. In addition, we can achieve diversity by receiving with multiple antennas: spatial diversity. The communication environment for the different times, frequencies, or antennas are each viewed as separate channels. We then scale the signal from each channel with a weighting factor and sum, or combine, the results. Many methods exist to determine proper weighting factors.

Diversity combining has been shown to be an effective means of improving system performance. Early experiments in the principles of diversity combining were first reported in 1927 [31]. There are many textbook examples illustrating the reduction in the receiver error rate proportional to the increase in the number of channels used for diversity combining [51]. A prime benefit is vastly decreased power consumption.

For an example, we will illustrate signaling with coherent binary differential phase shift keying (DPSK). The probability of error for signaling in an AWGN channel is of the form

$$P_e = \frac{1}{2}e^{-\gamma} \tag{1.2}$$

where γ is the SNR. Therefore the probability of error decreases exponentially with an increase in the SNR. However, signaling in a Rayleigh fading environment results in error rates of

$$P_e = \frac{1}{2(1+\gamma)} \tag{1.3}$$

We will require a large increase in power to achieve error rates similar to the AWGN channel.

However, if multiple copies of the transmitted signal are received over independently fading channels, the effects of fading are reduced. In a system with L diversity channels, the probability of error is approximated by the expression

$$P_e = \left(\frac{1}{2\gamma}\right)^L \begin{pmatrix} 2L-1 \\ L \end{pmatrix} \tag{1.4}$$

Now the error rate is proportional to the inverse L^{th} power of the SNR. Error performance is significantly improved and power consumption is dramatically reduced with increased diversity, as illustrated in Figure 1.2.

An algorithm is necessary to combine the received signals on the multiple channels. We will examine three classic combining methods. The first method is selection diversity. With selection diversity we send the signal from the channel with the largest SNR to the receiver. Such a system greatly reduces hardware complexity since only one receiver branch is necessary for all channels. However, a large penalty is paid in performance.

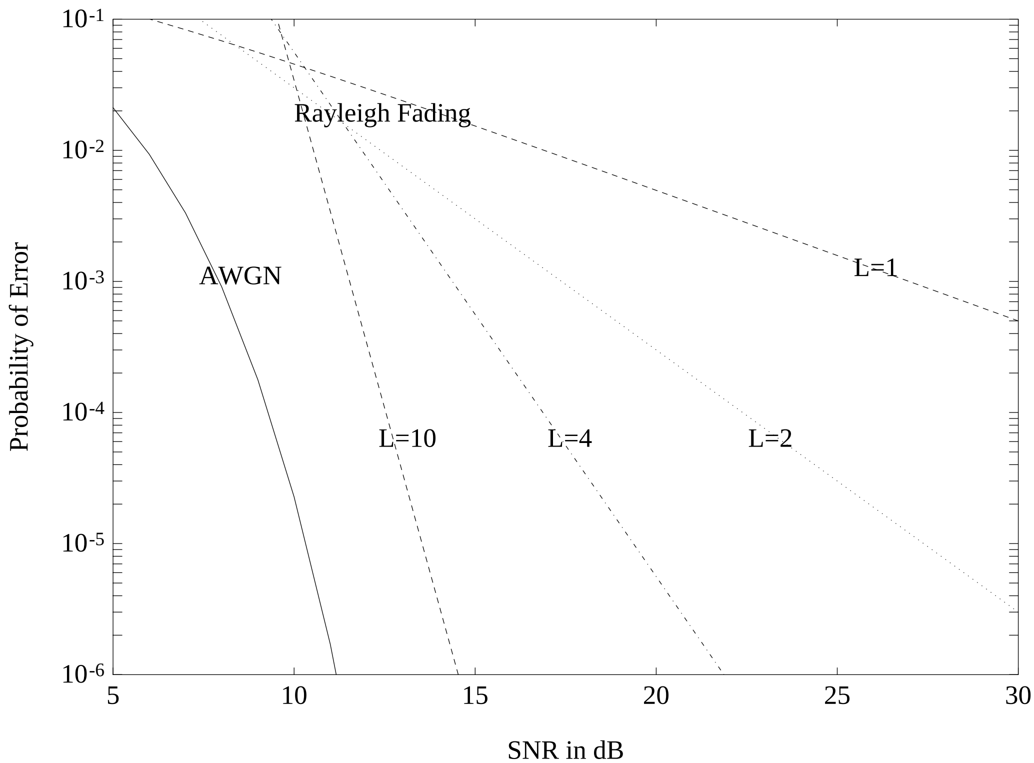


Figure 1.2: Comparison of error performance between signaling in an AWGN channel versus a Rayleigh fading channel with diversity.

In the second method, each of the L signals are weighted by the attenuation factor of their respective channel and summed. This is termed maximal-ratio combining and was first proposed by [32] in 1954. To perform maximal-ratio combining, we measure the attenuation and track the phase for each channel. With this added complexity much larger gain is achieved over selection diversity.

A simpler combining scheme, with only a small reduction in performance is equal gain combining. In this method, the received signals for each channel are equally weighted with unity gain and summed. Therefore measuring the attenuation of each channel is not required.

Figure 1.3 illustrates a comparison of the three methods for increasing number of channels. Maximal-ratio combining provides the largest gain, with the highest complexity. Equal gain combining only results in a $1dB$ loss in performance with as many as 10 channels and does not require the channel attenuation to be measured. Selection diversity results in greatly reduced performance for large numbers of channels. However, for a smaller number, such as two or three channels, the performance loss may be a tolerable trade-off between the reduced complexity of not needing a receiver branch for each antenna.

In the previous discussion and analysis, fading on the channels was assumed to be independently distributed. However, if for example antenna elements in an array were spaced very close together, the distribution of the fading on the channels would be correlated. It is necessary to determine the reduction in performance of a

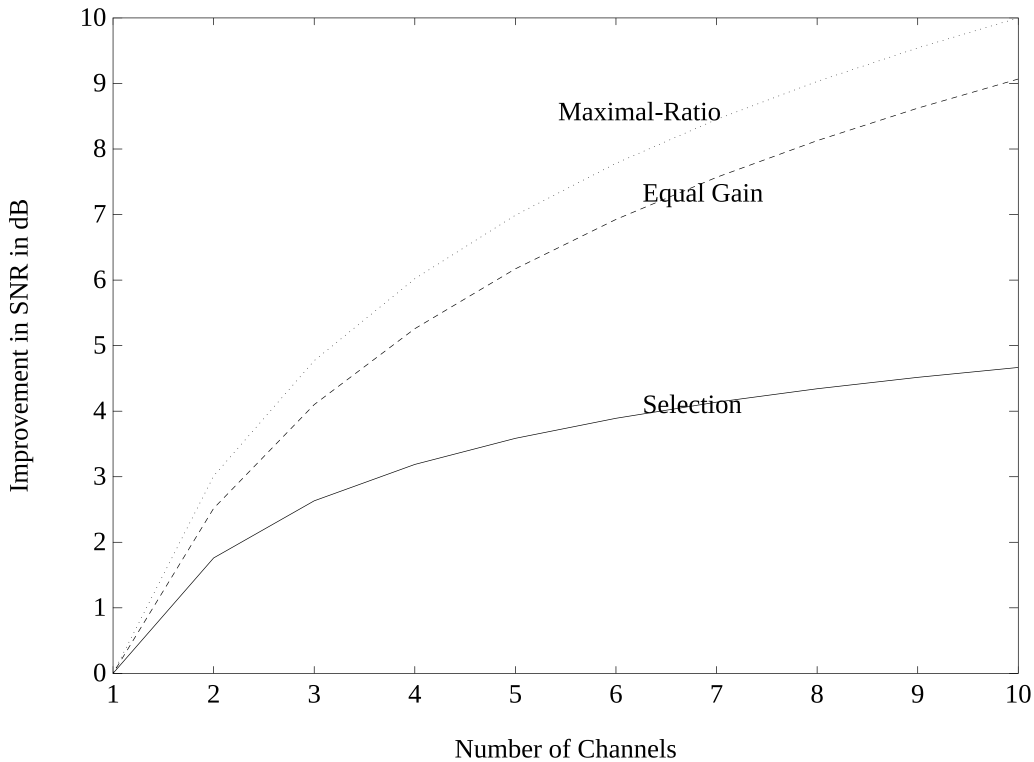


Figure 1.3: Improvement of the SNR from a diversity combiner compared to one channel.

diversity system when the channels are correlated by a certain amount. The amount of degradation experienced by the system will determine how critical is the design of independent channels. Analysis on binary signaling systems are described in [62, 31]. In addition, investigations of diversity combining with correlated channels are also performed by [34, 35, 36, 49, 71]. Each selects a different model to describe the channel correlation.

1.4 Interference Cancellation

1.4.1 Intersymbol Interference

As discussed in Section 1.1, the propagation channel will experience other forms of corruption besides fading. Delayed responses will cause ISI. Since the transmitter sends discrete-time symbols every T seconds, and the output of the receiver is sampled every T seconds, the channel is modeled as discrete-time. The received signal r_n at time n is a function the transmitted signal I_0 and the M previously transmitted signals. The transmitted signals are scaled by h_k, which is the channel impulse response. That is,

$$r_n = \sum_k I_{n-k} h_k + z_n \tag{1.5}$$

where z_n represents AWGN. We refer to this model as the *equivalent discrete-time white-noise filter model* [51]. The equalizer is the standard device used to compensate for ISI.

A major breakthrough in adaptive equalization was made by [40] in 1965 with the proposal of a zero-forcing algorithm to automatically adjust the tap weights of the linear equalizer. Subsequently the adaptive equalization problem was reformulated using a mean-square error solution in [21, 52] in 1969. A mathematical analysis of the convergence properties of an adaptive equalization with the least-mean square (LMS) algorithm was presented in [72] in 1972. The LMS algorithm, invented in 1960 by [79], is a scheme to iteratively adjust weights based on the steepest descent method.

The simplest structure used for equalization is the linear equalizer (LE). An illustration of an LE is given in Figure 1.4. The linear equalizer is comprised of a sequence of delays of length T, the symbol period, and tap weight coefficients c_j. The received signal, r_n, is multiplied by an equalizer weight and summed with previous r_n's which are multiplied by their corresponding equalizer weights. The result is an estimate of the transmitted symbol.

$$\hat{I}_n = \sum_{j=-N}^{N} c_j r_{n-j} \tag{1.6}$$

An example of a channel is given in Figure 1.5. The impulse and frequency response of the channels are illustrated. In Figure 1.6, the performance of the LE is demonstrated with the upper bound of the P_e versus SNR. With no equalization the performance is unacceptable. With the LE, the performance is only marginally

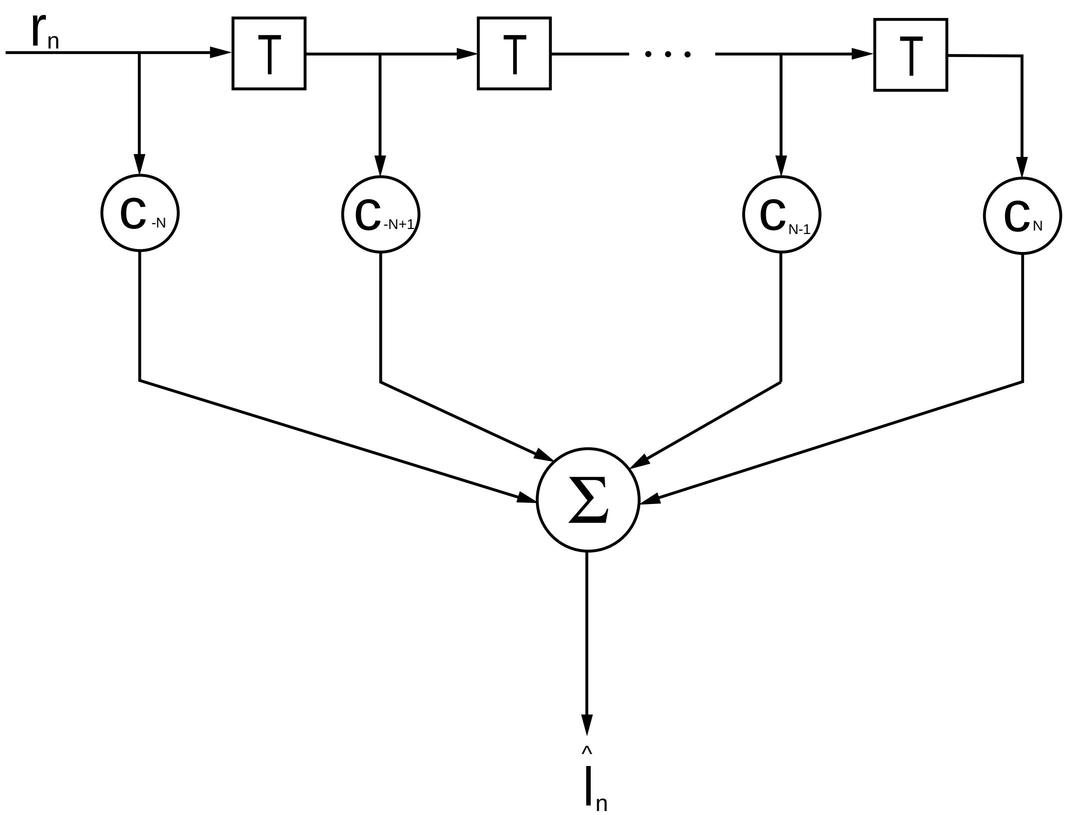


Figure 1.4: Linear equalizer.

worse than the case with no interference. Therefore with this channel, the LE almost completely compensates for the channel variations.

A second channel is illustrated in Figure 1.8. This channel has poor spectral characteristics. There is a null in the middle of the frequency spectrum. The performance of the LE for this channel is given in Figure 1.9. In this case the LE hardly provides any improvement in performance. The LE is an inadequate equalizer for channels with deep spectral nulls.

An alternative approach for equalization for channels with spectral nulls is to use nonlinear equalizers. The most common is the decision feedback equalizer (DFE). The DFE was first presented by [2] in 1967. Later in 1971, the DFE was optimized by minimizing the MSE in [50]. An illustration of a DFE is given in Figure 1.7. In a DFE, the output of the LE, the feedforward filter, is passed through a detector which selects the symbol which most closely matches the estimate of the transmitted symbol, $\tilde{I}_n$. This is fed back through another linear equalizer, the feedback filter, and the result is subtracted from the output of the feedforward filter. The output of the DFE is expressed as

$$\hat{I}_n = \sum_{j=-N_1}^{0} c_j r_{n-j} - \sum_{j=N_2}^{1} c_j \tilde{I}_{n-j} \tag{1.7}$$

The performance of the DFE for the channel of Figure 1.8 is also given in Figure 1.9. The DFE provides superior performance to the LE in eliminating the effects of the ISI.

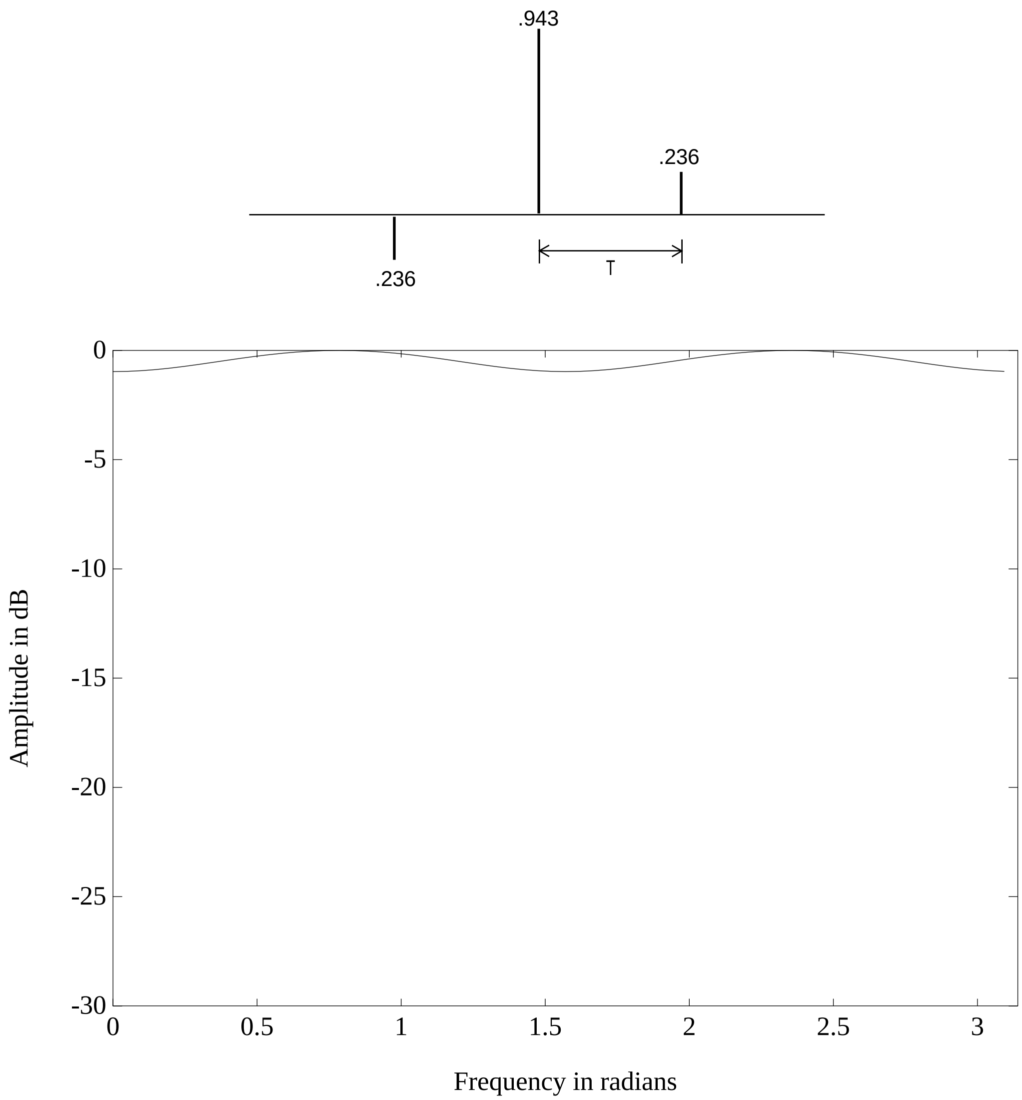


Figure 1.5: Example 1 of a channel impulse and frequency response.

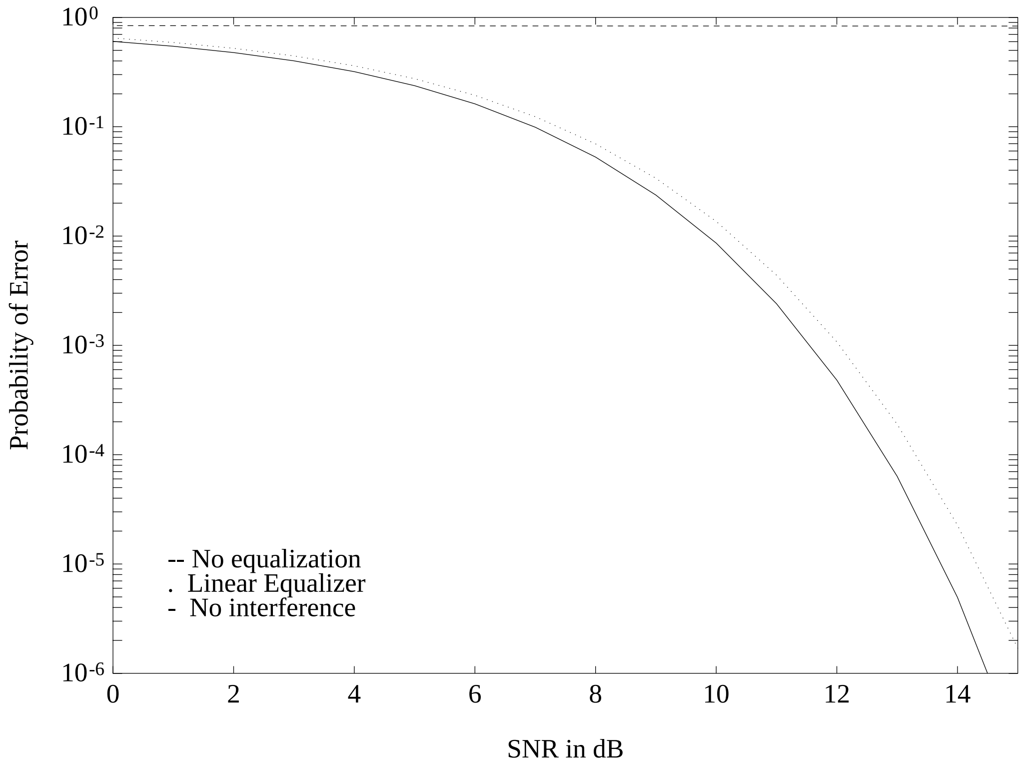


Figure 1.6: Performance of the linear equalizer.

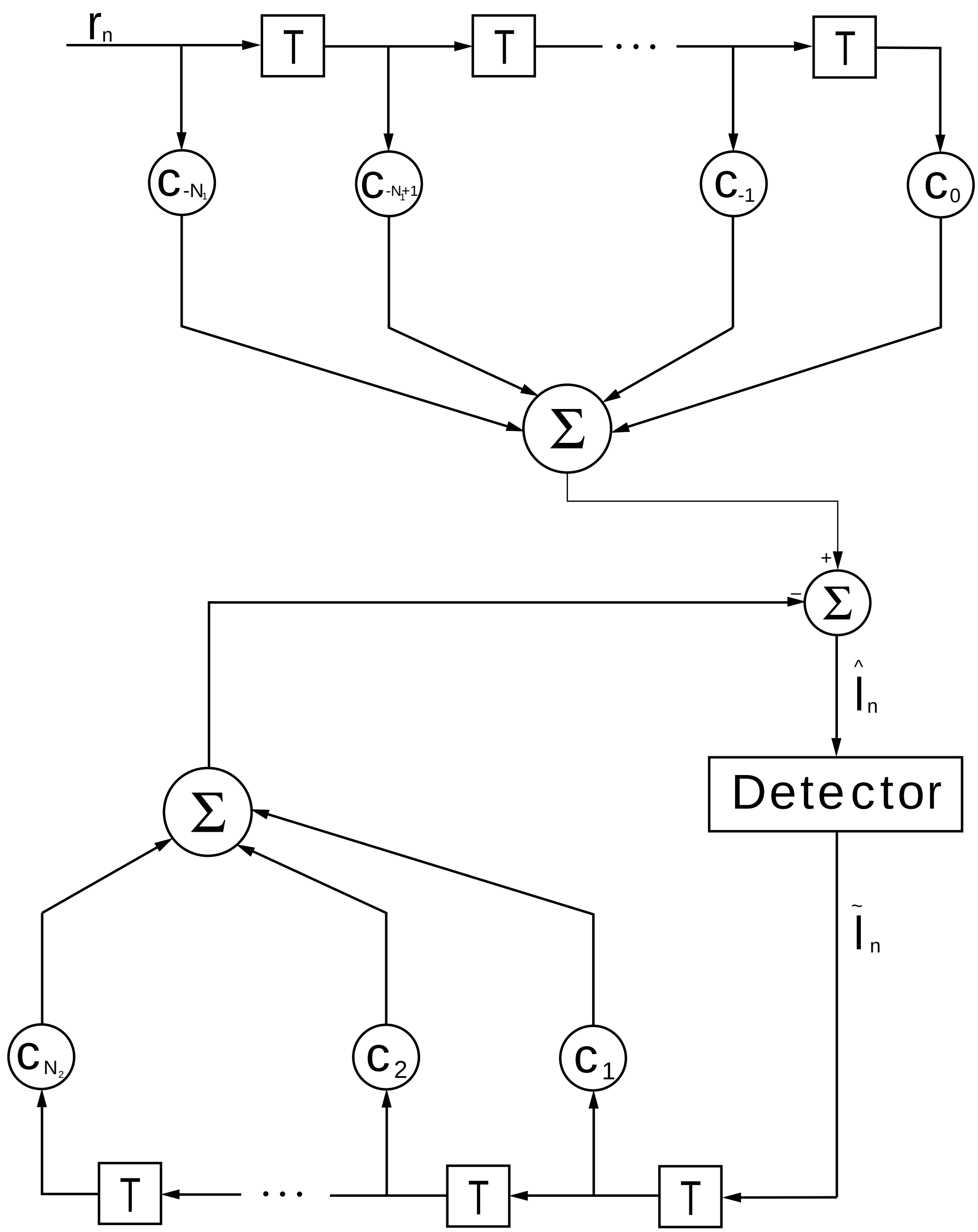


Figure 1.7: Decision feedback equalizer.

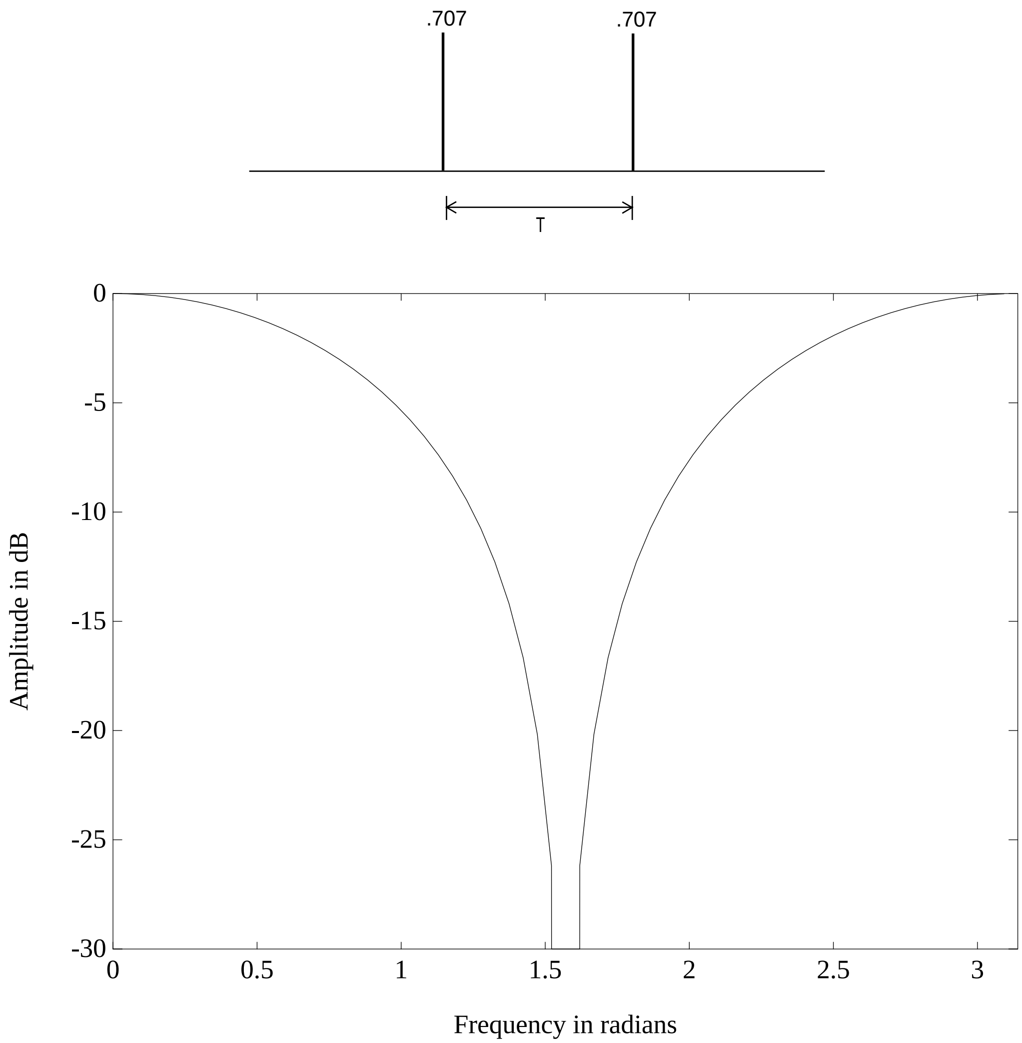


Figure 1.8: Example 2 of a channel impulse and frequency response.

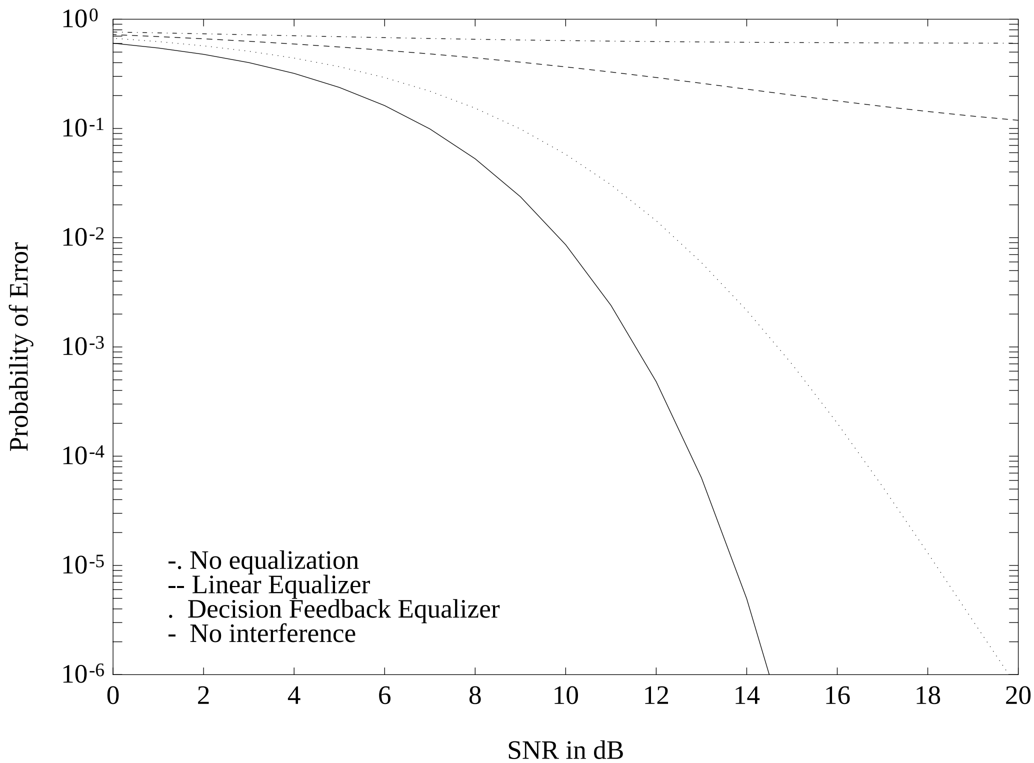


Figure 1.9: Performance of the linear equalizer and a decision feedback equalizer for a channel with a spectral null.

1.4.2 Co-Channel Interference

Users in a network transmitting on the same channel will cause CCI. An antenna array provides an effective means of reducing CCI. With a large number of antenna elements, we can attenuate the received signal in the directions of the interfering users, improving the ratio of the power of the desired signal to interference (SINR).

With adaptive beamforming, a set of antenna weights is determined such that the reception in the direction of the intended transmitter is optimized and the reception in the directions of the interferers is minimal. Initial development in adaptive beamforming is attributed to [28] with the side-lobe canceler in 1957. A subsequent major contribution [1] in 1966 was an algorithm based on maximizing the SNR at the antenna array output. Independently, another algorithm to adaptively adjust the weights based on the LMS algorithm was presented in [80] in 1967.

We present an example with a four element square antenna array and interference arriving at the receiver at angles of $-75°$, $-30°$, and $60°$. Antenna weights are calculated to minimize the MSE at the output of the antenna array. The results are illustrated in Figure 1.10. Nulls were created in the beampattern at $-75°$, $-30°$, and $60°$ to mitigate the effects of the interference. Similar techniques may be used with an adaptive transmitter to reduce the interference generated on transmission.

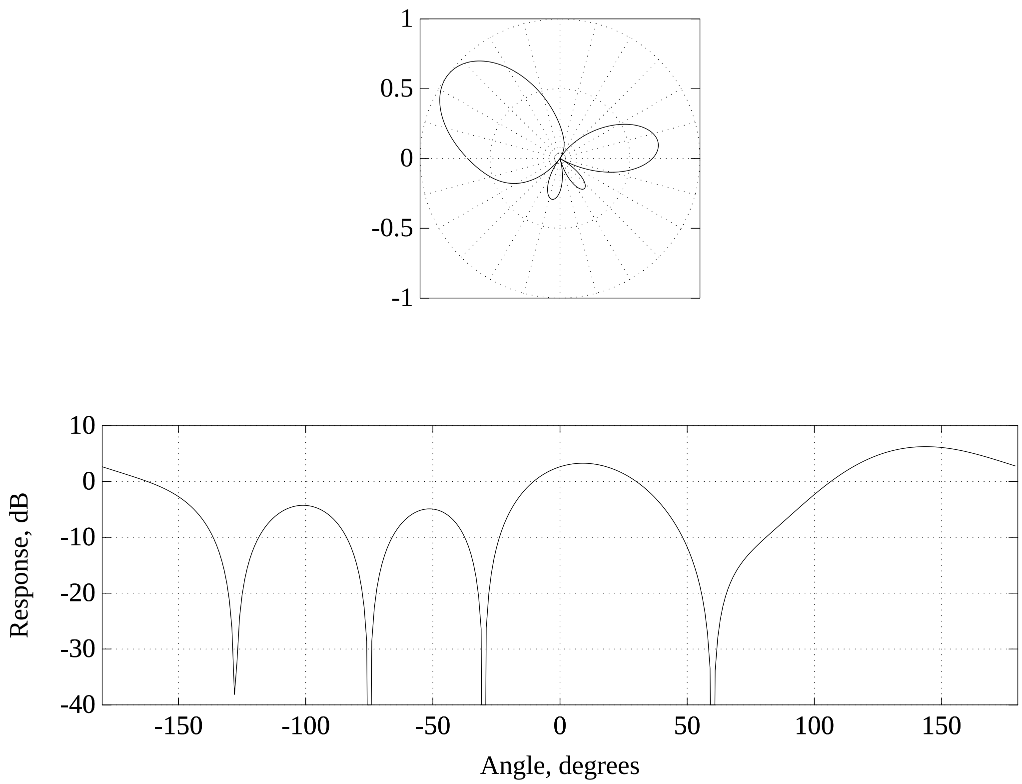


Figure 1.10: Antenna beampattern with interference from $-75°$, $-30°$, and $60°$.