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Abstract

An understanding of physics requires knowledge of mathematics. The converse is not true. By definition, pure mathematics contains no physics. Yet historically, mathematics has a rich history filled with physical applications. Mathematics was developed by people with the intent of making things work. As an engineer, I see these creators of early mathematics as budding engineers. This book is an attempt to tell their story, of the development of mathematical physics, as viewed by an engineer.

There are two distinct ways to learn mathematics: the traditional math way, by memorizing everything (the hard way), and the easy “physical” way, by associating every mathematical concept with its physical counterpart. These two methods have their pros and cons:

- Physicists and engineers choose to associate mathematical and physical concepts.
- Mathematicians choose to not make these associations.

We might conjecture why this is: Budding physicists and engineers may have come to mathematics while learning physics. Budding mathematicians may appreciate the beauty in the pure structures of mathematics.

Both methods have a chronological development, i.e., a history. Amazingly these histories heavily overlap. Why is that? Because most early mathematics evolved from attempts to understand the world, with the goal of navigating it. Pure mathematics followed when the corresponding physics was not yet discovered (or perhaps doesn’t exist).

This is not strictly a book on mathematics, physics and history: rather, it is a marriage of math and physics, presented roughly in chronological order, via its history. To teach mathematical physics in an orderly way, our treatment requires a step backwards in terms of the mathematics, but a step forward in terms of the physics. Historically speaking, mathematics was created by individuals, who by modern standards, would be viewed as engineers. This book contains the basic information that a well informed engineer needs to know.

The presentation is broken down around three streams, presented as five chapters: 1) Introduction, 2) Number Systems, 3) Algebra Equations, 4) Scalar Calculus, and 5) Vector Calculus. The material is delivered as 43 “Lectures,” spread out over a semester of 15 weeks, 3 lectures per week, with a three-lecture time-out for administrative duties. Eleven problem sets are provided for weekly assignments. These problems are written out in \LaTeX, with built-in solutions that may be compiled into the text by un-commenting one line of \LaTeX. Once the assignments are turned in, each student is given the solution.

Many students have rated these assignments as the most important part of the course. There is a built-in interplay between these assignments and the lectures. On many occasions I solved the problems in class, as motivation to come to every class.

There are four exams, one at the end of each of the three sections, plus the final. The first exam is in class, two others and the final are evening exams. Each exam and assignment is a \LaTeX file, with solutions encoded with a one line software switch. When the exam is returned by the student, the full solution is provided, while the exam is fresh in the students mind, providing a teaching moment. The exams are entirely based on the assignments. It is my philosophy that, in principle,
the students can see the exam in advance of taking it. In a real sense they do, since each exam is based on the assignments.

Author’s Personal Statement

An expert is someone who has made all possible mistakes in a small field. I don’t know if I would be called an expert, but I certainly have made my share of mistakes. I openly state that “I love making mistakes, because I learn so much from them.” One might call that the “expert’s corollary.”

This book has been written out of both my love for the topic of mathematical physics, and a desire to share many key concepts, and many new ideas on these basic concepts. Over the years I have developed a certain physical sense of math, along with a related mathematical sense of physics. While doing my research, I have come across what I feel are certain conceptual holes that need filling, and sense many deep relationships between math and physics that remain unidentified. While what we presently teach is not wrong, it is missing these relationships. What is lacking is an intuition for how math “works.” Good scientists “listen” to their data. In the same way we need to start listening to the language of mathematics. We need to let mathematics guide us toward our engineering goals.

History shows us, that over the millennia, many individuals have had deep insights. Such events define the history of mathematics. I hope the views offered here, small as they are, will be helpful in continuing to opening young minds.

As summarized in Fig. 1, this marriage of math, engineering and physics (MEP) will help us make progress in understanding the physical world. We must turn to mathematics and physics when trying to understand the universe. My views follow from a lifelong attempt to understand human communication, i.e., the perception and decoding of human speech sounds. This research arose from my 32 years at Bell Labs in the Acoustics Research Department. There such lifelong pursuits were not only possible, they were openly encouraged. The idea was that if you are successful at something, take it as far as you can. But on the other hand “you should not do something well that’s not worth doing.” People got fired for the latter. I should have left for a university after a mere 20 years, but the job was just too cushy.

In this text it is my goal to clarify conceptual errors while telling the story of physics and mathematics. My views have been inspired by classic works, as documented in the bibliography. This book was inspired by my reading of Stillwell (2002), through Chapter 21 (Fig. 2, p. 11). Somewhere in Chapter 22 I switched to the third edition (Stillwell, 2010), at which point I realized I had much more to master. It became clear that by teaching this material to first year engineers, I could absorb the advanced material at a reasonable pace. This book soon followed.

Summary

This is foremost a math book, but not the typical math book. First, this book is for the engineering minded, for those who need to understand math to do engineering, to learn how things work. In that

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1. https://auditorymodels.org/index.php/Main/Publications
2. MEP is a focused alternative to STEM.
3. I started around December 1970, fresh out of graduate school, and retired in December 5 2002.
Figure 1: There is a natural symbiotic relationship between Mathematics, Engineering, Physics (MEP), depicted in the Venn diagram. Mathematics provides the method and rigor. Engineering transforms the method into technology. Physics explores the boundaries. While these three disciplines work well together, there is poor communication, in part due to the different vocabulary. But style may more at issue. For example, Mathematics rarely uses a system of units, whereas Physics and Engineering depend critically on them. Mathematics strives to abstract the ideas into proofs. Physics rarely uses a proof. When they attempt rigor, they typically get into difficulty. Engineers blunder ahead, ignoring all the defects.

sense the book is more about physics and engineering than mathematics. Math skills are essential for making progress in building things, be it pyramids or computers, as clearly shown by the many great civilizations of the Chinese, Egyptians, Arabs (people of Mesopotamia), and Greeks and Romans.

Second, this is a book about the math that developed to explain physics, to allow people to engineer complex things. To sail around the world one needs to know how to navigate. This requires a model of the planets and stars. You can only know where you are, on earth, once you understand where earth is, relative to the sun, planets, Milky Way and the distant stars. The answers to such cosmic questions depends strongly on who you ask. Who is qualified to answer such question? It is best answered by those who study mathematics applied to the physical world. The utility and accuracy of that answer depends critically on the depth of understanding of the physics of the cosmic clock.

The English astronomer Edmond Halley (1656–1742) asked Newton (1643–1727) for the equation that describes the orbit of the planets. Halley was obviously interested in comets. Newton immediately answered “an ellipse.” It is said that Halley was stunned by the response (Stillwell, 2010, p. 176), as this was what had been experimentally observed by Kepler (c1619), and thus he knew Newton must have some deeper insight. Both were eventually Knighted.

When Halley asked Newton to explain how he knew, Newton responded “I calculated it.” But when challenged to show the calculation, Newton was unable to reproduce it. This open challenge eventually led to Newton’s grand treatise, *Philosophiae Naturalis Principia Mathematica* (July 5, 1687). It had a humble beginning, as a letter to Halley, explaining how to calculate the orbits of the planets. To do this Newton needed mathematics, a tool he had mastered. It is widely accepted that Isaac Newton and Gottfried Leibniz invented calculus. But the early record shows that perhaps
Bhāskara II (1114–1185 CE) had mastered the art well before Newton.\(^4\)

*Third,* the main goal of this book is to teach motivated engineers mathematics, in a way that it can be understood, mastered and remembered. How can this impossible goal be achieved? The answer is to fill in the gaps with *Who did what, and when?* Compared with the math, the historical record is easily mastered.

To be an expert in a field, one must know its history. This includes who the people were, what they did, and the credibility of their story. Do you believe the Pope or Galileo on the roles of the sun and the earth? The observables provided by science are clearly on Galileo’s side. Who were those first engineers? They are names we all know: Archimedes, Pythagoras, Leonardo da Vinci, Galileo, Newton, etc. All of these individuals had mastered mathematics. This book presents the tools taught to every engineer. Rather than memorizing complex formulas, make the relations “obvious” by mastering each simple underlying concept.

*Fourth,* when most educators look at this book, their immediate reactions are: *Each lecture is a topic we spend a week on (in our math/physics/engineering class).* And: *You have too much material crammed into one semester.* The first sentence is correct, the second is not. Tracking the students who have taken the course, looking at their grades, and interviewing them personally, demonstrate that the material presented here is appropriate for one semester.\(^5\)

To write this book I had to master the language of mathematics (John’s language). I had already mastered the language of engineering, and a good part of physics. One of my secondary goals is to bring down this scientific Tower of Babel, by unifying the terminology and removing the jargon.

**Acknowledgements**

Besides thanking my parents, I would like to credit John Stillwell for his constructive, historical summary of mathematics. My close friend and colleague Steve Levinson somehow drew me into this project, without my even knowing it. My brilliant graduate student Sarah Robinson was constantly at my side, grading homeworks and exams, and tutoring the students. Without her, I would never have survived the first semester the material was taught. Her proofreading skills are amazing. Thank you Sarah for your infinite help. Finally I would like to thank John D’Angelo for putting up with my many silly questions. When it comes to the heavy hitting, John was always there to provide a brilliant explanation that I could easily translate into Engineerese (Matheering?) (i.e., engineer language).

The services of my delightful friend Robert Fossum, emeritus professor of mathematics from the University of IL, who kindly pointed out my flawed use of mathematics. James (Jamie) Hutchinson’s precise use of the English language, thus dramatically raising the bar on my more than occasional-casual writing style. To each of you, thank you!

Finally I would like to thank my wife Sheau Feng Jeng, aka Patricia Allen, for her unbelievable support and love. She delivered constant peace of mind, without which this project could never have been started, much less finished. Many others played important roles, but they must remain anonymous.

–Jont Allen, Mahomet IL, Dec. 24, 2015 (Jan 1, 2018)

\(^4\)http://www-history.mcs.st-and.ac.uk/Projects/Pearce/Chapters/Ch8_5.html
\(^5\)http://www.istem.illinois.edu/news/jont.allen.html
Mathematics and its History (Stillwell, 2002)

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Figure 2: Table of contents of Stillwell (2002)
Preface

It is widely acknowledged that interdisciplinary science is the backbone of modern scientific investigation. This is embodied in the STEM (Science, Technology, Engineering, and Mathematics) programs. Contemporary research is about connecting different areas of knowledge, thus it requires an understanding of cross-disciplines. However, while STEM is being taught, interdisciplinary science is not, due to its inherent complexity and breadth. Furthermore there are few people to teach it. Mathematics, Engineering and Physics (MEP) are at the core of such studies.

STEM vs. MEP

Mathematics is about rigor. Mathematicians specifically attend to the definitions of increasingly general concepts. Thus mathematics advances slowly, as these complex definitions must be collectively agreed upon. Since it embraces rigor, mathematics shuns controversy. Physics explores the fringes of uncertainty, since physicists love controversy. Engineering addresses the advancement of technology. Engineers, much like mathematicians, are uncomfortable with uncertainty, but are trained to deal with it. Richard Hamming expressed this thought succinctly:

Great scientists tolerate ambiguity very well. They believe the theory enough to go ahead; they doubt it enough to notice the errors and faults so they can step forward and create the new replacement theory (Hamming, 1986).

To create an interdisciplinary STEM program, the MEP core is critical. In my view, this core should be based on mathematics, from the historical perspective, starting with Euclid or before (i.e., Chinese mathematics), up to modern information theory and logic. As a bare minimum, the fundamental theorems of mathematics (arithmetic, algebra, calculus, vector calculus, etc.) should be taught to every MEP student. The core of this curriculum is outlined in Table 1.5 (p. 27). Again paraphrasing Hamming:

Well, that book I’m trying to write [on Mathematics, Calculus, Probability and Statistics] says clearly . . . we can no longer tell you what mathematics you need, we must teach you how to find it for yourself. . . . We must give up the results, and teach the methods. You are only shown finished theorems and proofs. That is just plain wrong for teaching.

If first year students are taught a common MEP methodology and vocabulary, presented in terms of the history of mathematics, they will be equipped to

1. Exercise interdisciplinary science (STEM)

2. Communicate with other MEP trained (STEM) students and professors.
The goal should be a comprehensive understanding of the fundamental concepts of mathematics, defined in terms of the needs of the student. I contend that if students are given a comprehensive early conceptual understanding, they will end up at the top of their field. The goal should not be drilling on surface integrals, at least not in the first year. As suggested by Hamming, these details must be self-taught, at the time they are needed.

As identified by Hamming, a key problem is the traditional university approach, a five to eight semester sequence of: Calc I, II, III, Linear Algebra IV, DiffEq V, Real analysis VI, Complex analysis VII and given near-infinite stamina, Number theory VIII, over a time frame of three or more years (six semesters). This was the way I learned math. The process simply took too long, and the concepts spread too thin. After following this regime, I felt I had not fully mastered the material, so I started over. I consider myself to be largely self-taught.

We need a more effective teaching method. I am not suggesting we replace the standard six semester math curriculum of Calc I, II, III, etc. Rather, I am suggesting a broad unified introduction to all these topics, base on an historical approach. The present approach is driving the talent away from science and mathematics, by focusing too much on the details (as clearly stated by Hamming). One needs more than a high school education to succeed in college engineering courses. The key missing element in our present education system is teaching critical thought. Drilling facts does not do that.

By learning mathematics in the context of history, the student will fully and easily appreciate the underlying concepts. The history provides a uniform terminology for understanding the fundamentals of MEP. The present teaching method, using abstract proofs, with no (or few) figures or physical principles and units, by design, removes intuition and the motivation that was available to the creators of these fundamentals. The traditional six-semester initiation fee serves many students poorly, leaving them with little or no intuition, and an aversion for mathematics.

Postscript Dec 5, 2017 How to cram five semesters of math into one semester, and leave the students with something they can remember? Here are some examples:

Maxwell’s equations (MEs) are a fundamental challenging topic, presented in one lecture (Section 1.6.13, p. 173). Here is how it works:

1. The development starts with Sections 1.5.1-1.5.9 (pp. 108-125), which develop complex integration (p. 110) and the Laplace transform (pp. 123-124).

2. Kennelly’s (1893) 1893 complex impedance, as defined by Ohm’s Law, is the ratio of the force over the flow, the key elements being 1) capacitance (e.g., compliance) per unit area ($\epsilon_0 [F/d/m^2]$), and 2) inductance (e.g., mass) per unit area ($\mu_0 [H/m^2]$).

3. Section 1.6.1 (p. 128) develops analytic field theory, while Sect. 1.6.2 (p. 130) introduces $\nabla \cdot (\cdot)$, $\nabla \cdot (\cdot)$, and $\nabla \times (\cdot)$, starting from the scalar $A \cdot B$ and vector product $A \times B$ of two vectors.

4. On p. 168, second-order operators are introduced and given physical meanings (based on the physics of fluids), but most important they are given memorable names (DoC, CoG, Dog, God, and CoC: p. 130). Thanks to this somewhat quaint and gammy innovation, the students can both understand and easily remember the relationships between these confusing second-order vector calculus operations.

5. Exercises and examples are interspersed throughout the lectures.
6. The foregoing carefully sets the stage for ME (p. 173), introduced using proper names and units of the electrical and magnetic field intensity (strengths, and forces seen by charge) $E$, $H$, and electric and magnetic flux (flow) $\mathbf{D}$, $\mathbf{B}$, as summarized on page 174. The meanings of ME equations is next explored in integral form. After Sect. 1.6.2-1.6.13, the students are fully conversant with MEs. The conformation of this is in the final exam grade distributions.

Postscript Dec 15, 2017 As this book comes to completion, I’m reading and appreciating the Feynman lectures. We all know (I hope you know) that Feynman had a special lecture style, that was both entertaining, and informative. His communication skill was a result of his depth of understanding, and he was not afraid to question the present understanding of physics. He was always on a quest. He died in 1988, at the age of 70. Let us all be on his quest. Any belief that we have figured out the ways of the universe is absurd. We have a lot to learn. Major errors in our understanding must to be corrected. Its tautologic that one cannot understand the world around us until we understand its creation. That is, we cannot understand where we are going until we understand where we came from.

– Jont Allen
Chapter 1

Introduction

Much of early mathematics began before 1600 BCE, and centered around the love of art and music, due to the sensations of light and sound. Our psychological sense of color and pitch are determined by the frequencies (i.e., wavelengths) of light and sound. The Chinese and later the Pythagoreans are well known for their early contributions to early music theory. We are largely ignorant of exactly what the Chinese scholars knew. The best record comes from Euclid, who lived the 3rd century, after Pythagoras. Thus we can only trace the early mathematics back to the Pythagoreans in the 4th century (580-500 BCE), which is centered around the Pythagorean theorem and early music theory.

Pythagoras strongly believed that “all is number,” meaning that every number, and every mathematical and physical concept, could be explained by integral relationships, mostly based on either ratios, or the Pythagorean theorem. It is likely that his belief was based on Chinese mathematics from thousands of years earlier. It is also believed that his ideas, about the importance of integers, followed from the theory of music. The relationships between the musical notes (pitches) obey natural integral ratio relationships, based on the octave (a factor of two in frequency). The western 12-tone scale breaks the octave into 12 ratios. Today this has been rationalized to be the 12 root of 2, which is approximately equal to $18/17$. This innate sense of frequency ratios comes from the physiology the auditory organ (the cochlea).

As acknowledged by Stillwell (2010, p. 16), the Pythagorean view is relevant today:

*With the digital computer, digital audio, and digital video coding everything, at least approximately, into sequences of whole numbers, we are closer than ever to a world in which “all is number.”*

1.1 Early Science and Mathematics

While early Asian mathematics is not fully documented, it clearly defined the course for math for at least several thousand years. The first recorded mathematics were those of the Chinese (5000-1200 BCE) and the Egyptians (3,300 BCE). Some of the best early record were left by the people of Mesopotamia (Iraq, 1800 BCE)$^1$

The first 5,000 years of math not well documented, but the basic record is clear, as outlined in Fig. 1.1. Thanks to Euclid and later Diophantus (c250 CE), we have some vague understanding of Chinese mathematics. There is also the interesting modern studies of Yale science professor

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$^1$See Fig. 1.10, p. 49.
Joseph Needhan (Lin, 1995). For example, *Euclid’s formula* (Eq. 1.9, p. 48; Sec. 2.2.2, Fig. 2.3, p. 191) provides a method for computing Pythagorean triplets, a formula believed to be due to the Chinese (Stillwell, 2010, pp. 4-9).

Chinese bells and stringed musical instruments were exquisitely developed with tonal quality, as documented by ancient physical artifacts (Fletcher and Rossing, 2008). In fact this development was so rich that one must question why the Chinese failed to initiate the industrial revolution. Specifically, why did Europe eventually dominate with its innovation when it was the Chinese who did the extensive early invention?

It have been for the wrong reasons, but perhaps our best insight into the scientific history from China may have come from an American chemist and scholar from Yale, Joseph Needham, who learned to speak Chinese after falling in love with a chinese woman, and ended up researching early Chinese science and technology for the US government.

According to Lin (1995) this is known as the *Needham question*:

“Why did modern science, the mathematization of hypotheses about Nature, with all its implications for advanced technology, take its meteoric rise only in the West at the time of Galileo[, but] had not developed in Chinese civilization or Indian civilization?”

Needham cites the many developments in China:

“Gunpowder, the magnetic compass, and paper and printing, which Francis Bacon considered as the three most important inventions facilitating the West’s transformation from the Dark Ages to the modern world, were invented in China.” (Lin, 1995)

“Needham’s works attribute significant weight to the impact of Confucianism and Taoism on the pace of Chinese scientific discovery, and emphasizes what it describes as the ‘diffusionist’ approach of Chinese science as opposed to a perceived independent inventiveness in the western world. Needham held that the notion that the Chinese script had inhibited scientific thought was ‘grossly overrated’ ” (Grosswiler, 2004).

Lin was focused on military applications, missing the importance of non-military contributions. A large fraction of mathematics was developed to better understand the solar system, acoustics, musical instruments and the theory of sound and light. Eventually the universe became a popular topic, as it still is today.

Regarding the “Needham question,” I suspect the resolution is now clear. In the end, China withdrew from its several earlier expansions (Menzies, 2004, 2008).

1.1.1 **Lec 1 The Pythagorean theorem**

Thanks to Euclid’s Elements (c323 BCE) we have an historical record, tracing the progress in geometry, as established by the Pythagorean theorem, which states that for any right triangle

\[ c^2 = a^2 + b^2, \]  

having sides of lengths \((a, b, c)\) that are either positive real numbers, or more interesting, integers, such that \(c > [a, b]\) and \(a + b > c\). Integer solutions were likely found by trial and error rather than by Euclid’s formula.
1.1. EARLY SCIENCE AND MATHEMATICS

Chronological history pre 16th century

20th BCE Chinese (Primes; quadratic equation; Euclidean algorithm (GCD))
18th BCE Babylon (Mesopotamia/Iraq) (quadratic equation)
6th BCE Pythagoras (Thales) and the Pythagorean "tribe"
4th BCE Euclid (quadratic equation); Archimedes
3rd CE Diophantus c250CE
4th CE Alexandria Library destroyed 391CE
6th CE Brahmagupta (negative numbers; quadratic equation) 598-670CE
10th CE al-Khwārizmī (algebra) 830CE Hasan Ibn al-Haytham (Alhazen) 965-1040CE
12th CE Bhaskara (calculus) 1114-1183 Marco Polo 1254-1324
15th Leonardo da Vinci 1452-1519; Michelangelo 1475-1564 Copernicus 1473-1543
16th Tartaglia (cubic eqs); Bombelli 1526-1572 Galileo Galilei 1564-1642

Time-Line

![Time-LineBCE](image)

Figure 1.1: Mathematical time-line between 1500 BCE and 1650 CE. The western renaissance is considered to have occurred between the 15-17 centuries. However the Asian “renaissance” was likely well before the 1 century (i.e., 1500 BCE?). There is significant evidence that a Chinese ‘treasure ship’ visited Italy in 1434, initiating the Italian renaissance (Menzies, 2008). This was not the first encounter between the Italians and the Chinese, as documented in ‘The travels of Marco Polo’ (c1300 CE).

If $a, b, c$ are lengths, then $a^2, b^2, c^2$ are each the area of a square. Equation 1.1 says that the area $a^2$ plus the area $b^2$ equals the area $c^2$. Today a simple way to prove this is to compute the magnitude of the complex number $c = a + bj$, which forces the right angle

$$|c|^2 = (a + bj)(a - bj) = a^2 + b^2. \tag{1.2}$$

However, complex arithmetic was not an option for the Greek mathematicians, since complex numbers and algebra had yet to be discovered.

Almost 700 years after Euclid’s *Elements*, the Library of Alexandria was destroyed by fire (391 CE), taking with it much of the accumulated Greek knowledge. As a result, some of the best technical records remaining is Euclid’s Elements, along with some sparse mathematics due to Archimedes (c300 BCE) on geometrical series, computing the volume of a sphere, the area of the parabola, and elementary hydrostatics. In c1572 a copy of a book by Diophantus *Arithmetic* was discovered by Bombelli in the Vatican library (Stillwell, 2010, p. 51). This book became an inspiration for Galileo, Descartes, Fermat and Newton.

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2One might view Euclid significant role as a mathematical messenger.
Early number theory: Well before Pythagoras, the Babylonians (c.1800 BCE) had tables of triplets of integers \([a, b, c]\) that obey Eq. 1.1, such as \([3, 4, 5]\). However the triplets from the Babylonians were larger numbers, the largest being \(a = 12709, c = 18541\). A stone tablet (Plimpton-322) dating back to 1800 BCE (Fig. 1.10, p. 49) was found with integers for \([a, c]\). Given such sets of two numbers, which determined a third positive integer \(b = 13500\) such that \(b = \sqrt{c^2 - a^2}\), this table is more than convincing that the Babylonians were well aware of Pythagorean triplets (PTs), but less convincing that they had access to Euclid’s formula (Eq. 1.9 p. 48).

It seems likely that Euclid’s Elements was largely the source of the fruitful era due to the Greek mathematician Diophantus (215-285) (Fig. 1.1), who developed the concept of discrete mathematics, now known as Diophantine analysis. The term means that the solution, not the equation, are integers. The work of Diophantus was followed by fundamental change in mathematics, possibly leading to the development of algebra, but at least including the discovery of

1. negative numbers,
2. quadratic equation (Brahmagupta, 7th CE),
3. algebra (al-Khwārizmī, 9th CE), and
4. complex arithmetic (Bombelli, 15th CE).

These discoveries overlapped with the European middle (aka, dark) ages. While Europe went “dark,” presumably European intellectuals did not stop working during these many centuries.\(^4\)

1.1.2 What is mathematics?

Mathematics is a language, not so different from other languages. Today’s mathematics is a written language with an emphasis on symbols and glyphs, biased toward Greek letters. The specific evolution of these symbols is interesting (Mazur, 2014). Each symbol is dynamically assigned a meaning, appropriate for the problem being described. These symbols are then assembled to make sentences. It is similar to Chinese in that the spoken and written version are different across dialects. Like Chinese, the sentences may be read out loud in any language (dialect), while the mathematical sentence (like Chinese) is universal.

There is a second answer to this question, and that is told by studying its history, which starts with the earliest recorded record. This chronological view starts, of course, with the study of numbers. First there is the taxonomy of numbers. It took thousands of year to realize that numbers are more than the counting numbers \(\mathbb{N}\), and to create a symbol for nothing (i.e., zero), and negative numbers. With the invention of the abacus, a memory aid for the manipulation of complex sets of real integers, one could do very detailed calculations. But this required the discovery of algorithms, or procedures to follow to add, subtract, multiply (many adds of the same number) and divide (many subtracts of the same number), as in the Euclidean algorithm. Eventually it became clear to the experts (early mathematicians) that there were natural rules to be discovered, thus books (e.g., Euclid’s Elements) were written so summary this knowledge.

Mathematics turned out to be just like building a house. If you learn all the tools necessary to build a house, anyone can do it. But you need to learn and master the tool chain, from the bottom up. If you missing even one of the tools (if the chain is broken at any point), you will

\(^4\)It ould be interesting to search the archives of the monasteries, where the records were kept, to determine exactly what happened during this religious blackout.
not be able to build the house. There are many concepts that must be learned, and they form a long chain. All the concepts must be mastered, not simply learned. You need to learn about nails, screws, hammers, drills, lime, sand, water, concrete, you must know how to read drawings, down to the little marks, shovels, work with the labor force, lay a floor, bricks, ceramics, a compass, straight edge, Euclidean geometry, simple mathematics, ..., framing, siding, gutters, ..., drive ways, plumbing, sewers, and the list goes on.

A short list for mathematics is numbers \( \mathbb{N}, \mathbb{Z}, \mathbb{Q}, \mathbb{I}, \mathbb{C} \), algebra, derivatives, anti-derivatives (i., integration), differential equations, vectors and the spaces they define, matrices, matrix algebra, eigen values and vectors, solutions of systems of equations, matrix differential equations and their eigen solution. If you try to simply memorize these concepts, it is hopeless. They need to be learned in order, just as the case of building the house. Context is everything. You can’t build a house if you don’t know about screws or cement (plaster). Likewise in mathematics, you will not learn to integrate if you have failed to understand the difference between integers, complex numbers, and roots of polynomials.

Context is everything, and the most important context is physics. Without a physical problem to solve, at least at the beginning, there can be no mathematics, which began because people wanted to navigate the earth, or weigh things, namely understand the concept of gravity. Many questions were deep, such as “Where was the center of the universe?” Actually it was simple, ask the Pope and he will tell you. But church dogma only goes to the end of the block. Mathematics, along with a heavily dose of physics, finally answered this huge question. Some needed to perfect the telescope, and put satellites into space, and view the cosmos. Without mathematics, none of this would, or could have happened.

The point of all this blah, blah is that the more mathematics you learn, the easier it is to understand, because mathematics builds, from the bottom up. Its one continuum set of concepts, much like the construction of a house. If you try to learn calculus and differential equation, by skipping simple number theory, the lessons Will be much harder to understand. You will end up memorizing instead of understanding, and as a result you will forget all of it. When you truly understand something, it can never be forgotten. An nice example is the solution to a quadratic equation: If you learn how to complete the square (p. 59), you will never forget the quadratic formula. Complex numbers naturally follow.

**Math (the syntax) is a language:** It seems strange when people complain that they “can’t learn math,” but they claim to be good at languages. Pre-high-school students tend to confuse arithmetic with math. One does not need to be good at arithmetic to be good at math (but it doesn’t hurt).

Math is a language, with the symbols taken from various languages, with a bias toward Greek, obviously due to the popularity of Euclid’s Elements. Learning languages is an advanced social skill. However the social outcomes are very different between learning a language and math. Learning a new language is fun because it opens doors to other cultures.

Math is different due to the rigor of the rules of the language, along with the way it is taught (e.g., not as a language). A third difference between math and language is that math evolved from physics, with important technical applications.

The rules of math are defined by algebra. For example the sentence \( a = b \) means that the number \( a \) has the same value as the number \( b \). The sentence is spoken as “a equals b.” The numbers are nouns and the equal sign says they are equivalent, playing the role of a verb, or action symbol. Following the rules of algebra, this sentence may be rewritten as \( a - b = 0 \). Here the...
symbols for minus and equal indicate two types of actions.

Sentences can become arbitrarily complex, such as the definition of the integral of a function, or a differential equation. But in each case, the mathematical sentence is written down, may be read out loud, has a well defined meaning, and may be manipulated into equivalent forms following the rules of algebra and calculus. This language of mathematics is powerful, with deep consequences, known as theorems.

The writer of an equation should always translate (explicitly summarize the meaning of the expression), so the reader will not miss the main point, as a simply a matter of clear writing.

Just as math is a language, so may language be thought of as mathematics. To properly write correct English it is necessary to understand the construction of the sentence. It is important to identify the subject, verb, object, and various types of modifying phrases. Look up the interesting distinction between *that* and *which*. Thus, like math, language has rules. Most individuals use what “sounds right,” but if you’re learning English as a second language, it is necessary to understand the rules, which are arguably easier to master than the foreign speech sounds.

1.2 Early physics as mathematics: Back to Pythagoras

The role of mathematics is to summarize algorithms (i.e., sets of rules), and formalize the idea as a theorem. Pythagoras and his followers, the Pythagorean, believed that there was an fundamental relationship between mathematics and the physical world. The Pythagoreans may have been the first to capitalize on the relationship between science and mathematics, to use mathematics to design thing for profit. This may have been the beginning of capitalizing technology, based on the relationship between physics and math. This impacted commerce in many ways, such as map making, tools, implements of war (the wheel, gunpowder), art (music), water transport, sanitation, secure communication, food, etc. Of course the Chinese were the first to master many of these technologies.

Why is Eq. 1.1 called a *theorem*, and what exactly needs to be proved? We do not need to prove that \((a, b, c)\) obey this relationship, since this is a condition that is observed. We do not need to prove that \(a^2\) is the area of a square, as this is the definition of the area of a square. What needs to be proved is that the relation \(c^2 = a^2 + b^2\) holds *if, and only if*, the angle between the two shorter sides is \(90^\circ\). The Pythagoras’ theorem (Eq. 1.1) did not begin with Euclid or Pythagoras, rather they appreciated its importance, and documented it.

In the end the Pythagoreans, who instilled fear in the neighborhood, were burned out, and murdered. This may be the fate of mixing technology with politics:

“Whether the complete rule of number (integers) is wise remains to be seen. It is said that when the Pythagoreans tried to extend their influence into politics they met with popular resistance. Pythagoras fled, but he was murdered in nearby Mesopotamia in 497 BCE.”

–Stillwell (2010, p. 16)

\(^6\)https://en.oxforddictionaries.com/usage/that-or-which

\(^7\)It seems likely that the Chinese and Egyptians also did this, but more difficult to document.
1.2. EARLY PHYSICS AS MATHEMATICS

Chronological history 16-19th 3 centuries

<table>
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<th>1525</th>
<th>1596</th>
<th>1643</th>
<th>1700</th>
<th>1750</th>
<th>1800</th>
<th>1855</th>
</tr>
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<tbody>
<tr>
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<td>Galileo</td>
<td>Newton</td>
<td>Jacob Bernoulli</td>
<td>Euler</td>
<td>Gauss</td>
<td>Mozart</td>
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<tr>
<td>d'Alembert</td>
<td>Johann Bernoulli</td>
<td>Beethoven</td>
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Figure 1.2: Exploded time-line of the three centuries between the 16th and 19th CE. Given the time line, it seems likely that Bombelli’s discovery of Diophantus’ book on “Arithmetic” in the Vatican library, triggered many of the ideas presented by Galileo, Descartes and Fermat, followed by others (i.e., Newton). Bombelli’s discovery might be considered as a magic moment in mathematics. The vertical red lines connect mentor-student relationships. Figure 1.1 (p. 19) gives the time-line from 1500BCE to 1650CE. Figure 1.13 (p. 58) presents the 17-20 CE (Newton–Einstein) view from 1640–1950, and Fig. 1.25 (p. 107) outlines the full (Bombelli–Einstein, 1525-1925) period. [fig:TimeLine16_18]

1.2.1 Modern mathematics is born

Modern mathematics (what we practice today) was born in the 15-16th centuries, in the minds of Leonardo da Vinci, Bombelli, Galileo, Descartes, Fermat, and many others (Stillwell, 2010). Many of these early masters’ were, like the Pythagoreans, secretive to the extreme about how they solved problems. This soon changed due to Galileo, Mersenne, Descartes and Newton, causing mathematics to blossom. During this time the developments were hectic, seemingly disconnected. This is a wrong impression, as the development was dependent new technologies such as the telescope (optics) and more accurate time and frequency measurements, due to Galileo’s studies of the pendulum, and a better understanding of the relation \( f \lambda = c \), between frequency \( f \) and wavelength \( \lambda \).

1.2.2 Science meets mathematics

Galileo: In 1589 Galileo famously conceptualized an experiment where he suggested dropping two different weights from the Leaning Tower of Pisa, which suggested that they must take the same time to hit the ground. Conceptually this is an mathematically sophisticated experiment, driven by a mathematical argument in which he considered the two weights to be connected by an elastic cord (a spring). His studies resulted in the concept of conservation of energy, one of the cornerstones of physical theory since that time.

By joining them with an elastic cord, the masses become one. If the velocity were proportional to the mass, as believed by Archimedes, the sum of the two weights would necessarily fall even faster. This results in a logical fallacy: How can two masses fall faster than either? This also violate the concept of conservation of energy, as the total energy of two masses would be greater than that of the parts. In fact Galileo’s argument may have been the first time that the principle of conservation of energy was clearly stated.

It seems likely that Galileo was attracted to this model of two masses connected by a spring because he was also interested in planetary motion, which consist of masses (sun, earth, moon), also mutually attracted by gravity.

Galileo also performed related experiments on pendulums, where he varying the length \( l \), mass \( m \), and and angle \( \theta \) of the swing. By measuring the period (periods/unit time) he was able to formulate precise rules between the variables. This experiment also measured the force exerted...
CHAPTER 1. INTRODUCTION

Figure 1.3: Depiction of the argument of Galileo (unpublished book of 1638) as to why weights of different masses (i.e., weight) must fall with the same velocity, contrary to what Archimedes had proposed c250 BCE.

by gravity, so the experiments were related, but in very different ways. The pendulum served as the ideal clock, as it needed very little energy to keep it going, due to its very low friction (energy loss).

In a related experiment, Galileo measure the duration of a day by counting the number of swings of the pendulum in 24 hours, measured precisely by daily period of a star as it crossed the tip of a church steeple. The number of seconds in a day is precisely an integer (rounded to the nearest integer). It is the product of 24 [hr/day] \times 60 \times 60 [s/hr] = 86400 = 2^7 \cdot 3^2 [s/day]. This number may be reduced many ways, and remain precise, starting with seven factors of 2, 2 factors of 3 or 1 factor of 5. Factoring the number of days in a year (5*73) is impractical, thus cannot be easily treated as an integer.

Galileo also extended work on the relationship of wavelength and frequency of a sound wave in musical instruments. On top of these impressive accomplishments, Galileo greatly improved the telescope, which he needed for his observations of the planets.

Many of Galileo’s contributions resulted in new mathematics, leading to Newton’s discovery of the wave equation (c1687), followed 60 years later by its one-dimensional general solution by d’Alembert (c1747).

Mersenne: Mersenne (1588-1648) also contributed to our understanding of the relationship between the wavelength and the dimensions of musical instruments. At first Mersenne strongly disagreed with Galileo, partially due errors in Galileo’s reports of his results. But once Mersenne saw the significance of Galileo’s conclusion, he became Galileo’s strongest advocate, helping to spread the word (Palmerino, 1999).

Newton: With the closure of Cambridge University due to the plague of 1665, Newton returned home to Woolsthorpe-by-Colsterworth (95 [mi] north of London), to worked by himself, for over a year. It was during this time he did his most creative work.

While Newton (16421726) may be best known for his studies on light Newton (1687), he was the first to predict the speed of sound. However his theory was in error by \sqrt{c_p/c_v} = \sqrt{1.4} = 1.183. This famous error would not be resolved for over two hundred years, awaiting the formulation of thermodynamics and the equi-partition theorem, by Maxwell and Boltzmann, and others.

Just 11 years prior to Newton’s 1687 Principia, there was a basic understanding that sound and light traveled at very different speeds, due to the experiments of Ole Rømer.

Ole Rømer first demonstrated in 1676 that light travels at a finite speed (as opposed

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8The square root of the ratio of the specific heat capacity at constant pressure \(c_p\) to that at constant volume \(c_v\).
1.2. EARLY PHYSICS AS MATHEMATICS

Figure 1.4: Above: Jakob Bernoulli (1655-1705) and Johann (1667-1748) Bernoulli. Below: Leonhard Euler (1707-1783) and Jean le Rond d’Alembert (1717-1783). Euler was blind in his right eye, hence the left portrait view. The figure numbers are from Stillwell (2010).
to instantaneously) by studying the apparent motion of Jupiter’s moon Io. In 1865, James Clerk Maxwell proposed that light was an electromagnetic wave, and therefore traveled at the speed $c_o$ appearing in his theory of electromagnetism.\(^9\)

**Studies of vision and hearing:** Since light and music played such a key role in the development of science, it was important to understand the mechanism of our perception of light and sound. There are many outstanding examples where physiology impacted mathematics. Leonardo da Vinci is well known for his early studies of the human body. Exploring our physiological senses requires a scientific understanding of the physical processes of vision and hearing, first considered by Newton (1687), but first properly researched much later by Helmholtz (1863a)\(^10\) (Stillwell, 2010, p. 261). Helmholtz’s (1821-1894) studies and theories of music and the perception of sound are fundamental scientific contributions (Helmholtz, 1863a). His best known mathematical contribution is today known as the *Fundamental theorem of vector calculus*, or simply *Helmholtz theorem*.

**The amazing Bernoulli family:** The first individual who seems to have openly recognized the importance of mathematics, enough to actually teach it, was Jacob Bernoulli (16541705) (Fig. 1.4). Jacob worked on what is now viewed as the standard package of analytic “circular” (i.e., periodic) functions: $\sin(x)$, $\cos(x)$, $\exp(x)$, $\log(x)$.\(^11\) Eventually the full details were developed (for real variables) by Euler (Sections 1.4.1 p. 73 and 3.1.1, p. 198).

From Fig. 1.15 we see that Jacob was contemporary with Descartes, Fermat, and Newton. Thus it seems likely that he was strongly influenced by Newton, who in turn was influenced by Descartes,\(^12\) Vite and Wallis (Stillwell, 2010, p. 175).

Jacob Bernoulli, like all successful mathematicians of the day, was largely self-taught. Yet Jacob was in a new category of mathematicians, because he was an effective teacher. Jacob taught his sibling Johann (16671748), who then taught his sibling Daniel (17001782). But most importantly, Johann taught Leonhard Euler (1707-1783) (Figs. 1.4 and 1.25, p. 107), the most prolific (thus influential) of all mathematicians. This teaching resulted in an explosion of new ideas and understanding. It is most significant that all four mathematicians published their methods and findings. Much later, Jacob studied with students of Descartes\(^13\) (Stillwell, 2010, p. 268-9).

**Euler:** Euler went far beyond all the Bernoulli family (Stillwell, 2010, p. 315). A special strength of Euler was the degree to which he published. First he would master a topic, and then he would publish. His papers continued to appear long after his death (Calinger, 2015).

**d’Alembert:** Another individual of that time of special note, who also published extensively, was d’Alembert (Fig. 1.4). Some of the most creative ideas were first proposed by d’Alembert. Unfortunately, and perhaps somewhat unfairly, his rigor was criticized by Euler, and later by Gauss (Stillwell, 2010). But once the tools of mathematics were finally openly published, largely by Euler, mathematics grew exponentially.\(^14\)

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\(^10\)https://en.wikipedia.org/wiki/Acoustics
\(^11\)The $\log$ and $\tan$ functions are related by Eq. 1.83 p. 107.
\(^12\)https://en.wikipedia.org/wiki/Early_life_of_Isaac_Newton
\(^13\)It seems clear that Descartes was also a teacher.
\(^14\)There are at least three useful exponential scales: Factors of 2, factors of $e \approx 2.7$, and factors of 10. The decibel uses factors of 2 ($6 \text{[dB]}$) and factors of 10 ($20 \text{[dB]}$). Information theory uses powers of 2 (1 [bit]), 4 (2 [bits]). Circuit
1.2. EARLY PHYSICS AS MATHEMATICS

Gauss: It was one of the most creative times in mathematics. Figure 1.25 (p. 107) shows the time-line of the most famous mathematicians. I suspect that Gauss owed a great debt to Euler. Gauss was born at the end of Euler’s long and productive life. Surely he must have been a scholar of Euler. It is also interesting that Leonhard Euler was a contemporary of Mozart, and James Clerk Maxwell of Abraham Lincoln.

- The Pythagorean Theorem is the mathematical spring which bore the three streams.
- Several centuries per stream:

  1) Numbers:
     6th BCE $\mathbb{N}$ counting numbers, $\mathbb{Q}$ (Rationals), $\mathbb{P}$ Primes
     5th BCE $\mathbb{Z}$ Common Integers, $\mathbb{I}$ Irrationals
     7th CE zero $\in \mathbb{Z}$
  2) Geometry: (e.g., lines, circles, spheres, toroids, . . .)
     17th CE Composition of polynomials (Descartes, Fermat)
     Euclid’s Geometry + algebra $\Rightarrow$ Analytic Geometry
     18th CE Fundamental Theorem of Algebra
  3) Infinity: ($\infty \rightarrow \text{Sets}$)
     17-18th CE Taylor series, Functions, Calculus (Newton)
     19th CE $\mathbb{R}$ Real, $\mathbb{C}$ Complex 1851
     20th CE Set theory

Figure 1.5: Three streams followed from Pythagorean theorem: Number Systems (Stream 1), Geometry (Stream 2) and Infinity (Stream 3). /TABLE/TableRoadMap.tex

1.2.3 Three Streams from the Pythagorean theorem

From the outset of his presentation Stillwell (2010, p. 1) defines “three great streams of mathematical thought: Numbers, Geometry and Infinity” that flow from the Pythagorean theorem, as summarized in Table 1.5. This is a useful concept, based on reasoning not as obvious as one might think. Many factors are in play here. One of these is the strongly held opinion of Pythagoras that all mathematics should be based on integers. The rest are tied up in the long, necessarily complex history of mathematics, as best summarized by the fundamental theorems (Table 1.1, p. 38), each of which is discussed in detail in a relevant chapter. Stillwell’s concept of three streams, following from the Pythagorean theorem, is the organizing principle behind the this book:

1. Introduction Chapter 1 is intended to be a self-contained survey of basic pre-college mathematics, as a detailed overview of the fundamentals, presented as three streams:

   1.3 Number systems (Stream 1)
   1.4 Algebraic equations (Stream 2)
   1.5 Scalar calculus (Stream 3a)

theory uses factors of both $e$ and 10.
1.6 Vector calculus (Stream 3b)

If you’re a student, stick to Chapter 1.

Chapters 2-5 are disorganized rambling research ideas that have not yet found a home. Students, please stay out of these chapters.

2. Number Systems (Chapter 2: Stream 1) Some uncertain ideas of number systems, starting with prime numbers, through complex numbers, vectors and matrices.

3. Algebraic Equations (Chapter 3: Stream 2) Algebra and its development, as we know it today. The theory of real and complex equations and functions of real and complex variables. Complex impedance $Z(s)$ of complex frequency $s = \sigma + \omega j$ is covered with some care, developing the topic which is needed for engineering mathematics.


1.3 Stream 1: Number Systems

Number theory (discrete, i.e., integer mathematics) was a starting point for many key ideas. For example, in Euclid’s geometrical constructions the Pythagorean theorem for real $[a, b, c]$ was accepted as true, but the emphasis in the early analysis was on integer constructions, such as Euclid’s formula for Pythagorean triplets (Eq. 1.9, Fig. 1.9, p. 48).

As we shall see, the Pythagorean theorem is a rich source of mathematical constructions, such as composition of polynomials, and solutions of Pell’s equation by eigen-vector and recursive analysis methods. Recursive difference equation solutions predate calculus, going back at least to the Chinese (c2000 BCE). These are early (pre-limit) forms of differential equations, best analyzed using an eigen-vector or eigen-function expansion (a geometrical concept from linear algebra), as an orthogonal set of normalized (unit-length) vectors (Appendix D, p. 267),

The first use of zero and $\infty$:
It is hard to imagine that one would not appreciate the concept of zero and negative numbers when using an abacus. It does not take much imagination to go from counting numbers $\mathbb{N}$ to the set of all integers $\mathbb{Z}$, including zero. On an abacus, subtraction is obviously the inverse of addition. Subtraction, to obtain zero abacus beads, is no different than subtraction from zero, giving negative beads. To assume the Romans who first developed counting sticks, or the Chinese who then deployed the concept using beads, did not understand negative numbers, is impossible.

However, understanding the concept of zero (and negative numbers) is not the same as having a symbolic notation. The Roman number system has no such symbols. The first recorded use of a symbol for zero is said to be by Brahmagupta\textsuperscript{15} in 628 CE.\textsuperscript{16}\textsuperscript{17} Defining zero (c628 CE)

\begin{footnotesize}
\textsuperscript{15}http://news.nationalgeographic.com/2017/09/origin-zero-bakhshali-manuscript-video-spd/\textsuperscript{16}
\textsuperscript{16}The fall of the Roman Empire has been established as Sept. 4, 476 CE.\textsuperscript{17}
\textsuperscript{17}https://www.nytimes.com/2017/10/07/opinion/sunday/who-invented-zero.html
\end{footnotesize}
1.3. STREAM 1: NUMBER SYSTEMS (10 LECTURES)

depends on the concept of subtraction, which formally requires the creation of algebra (c830 CE, Fig. 1.1, p. 19). But apparently it takes more than 600 years, i.e., from the time Roman numerals were put into use, without any symbol for zero, to the time when the symbol for zero is first documented. Likely this delay is more about the political situation, such as government rulings, than mathematics.

The concept that caused much more difficulty was $\infty$, first solved by Riemann in 1851 with the development of the extended plane, which mapped the plane to a sphere (Fig. 1.20 p. 96). His construction made it clear that the point at $\infty$ is simply another point on the open plane, since rotating the sphere (extended plane) moves the point at $\infty$ to a finite point on the plane, thereby closing the plane.

1.3.1 Lec 2: The Taxonomy of Numbers: $\mathbb{N}, \mathbb{P}, \mathbb{Z}, \mathbb{Q}, \mathbb{F}, \mathbb{I}, \mathbb{R}, \mathbb{C}$

Once symbols for zero and negative numbers were accepted, progress could be made. To fully understand numbers, a transparent notation was required. First one must differentiate between the different classes (genus) of numbers, providing a notation that defines each of these classes, along with their relationships. It is logical to start with the most basic counting numbers, which we indicate with the double-bold symbol $\mathbb{N}$. For clarity we shall refer to the natural numbers as counting numbers, since natural here means integer. The mathematical sentence “$2 \in \mathbb{N}$” is read as $2$ is a member of the set of counting numbers. The word set is defined as the collection of any objects that share a specific property. Typically the set may be defined either as a sentence, or by example.

**Counting numbers $\mathbb{N}$:** These are known as the “natural numbers” $\mathbb{N} = 1, 2, 3, \cdots$, denoted by the double-bold symbol $\mathbb{N}$. For clarity we shall refer to the natural numbers as counting numbers, since *natural* here means *integer*. The mathematical sentence “$2 \in \mathbb{N}$” is read as 2 is a member of the set of counting numbers. The word *set* is defined as the collection of any objects that share a specific property. Typically the set may be defined either as a sentence, or by example.

**Primes $\mathbb{P}$:** A number is prime ($\pi_n \in \mathbb{P}$) if its only factors are 1 and itself. The set of Primes $\mathbb{P}$ are a subset of the counting numbers ($\mathbb{P} \subset \mathbb{N}$). A somewhat amazing fact, well known to the earliest mathematicians, is that every integer may be written as a unique product of primes. A second key idea is that the density of primes $\rho_\pi(N) \sim 1 / \log(N)$ ($\rho_\pi(N)$ is inversely proportional to the log of $N$ (Eq. 2.1, p. 184), an observation first made by Gauss (Goldstein, 1973). A third is that there is a prime between every integer $N$ and $2N$, excluding $2N$.

**Exercise:** Write out the first 10 to 20 integers in factored form. **Sol:** $1, 2, 3, 2^2, 5, 2 \cdot 3, 7, 2^3, 3^2, 2 \cdot 5, 11, 3 \cdot 2^2, 13, 2 \cdot 7, 3 \cdot 5, 2^4, 17, 2 \cdot 3^2, 19, 2^3 \cdot 5$.

**Exercise:** Write integers 2 to 20 in terms of $\pi_n$. Here is a table to assist you:

<table>
<thead>
<tr>
<th>$n$</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>8</th>
<th>9</th>
<th>10</th>
<th>11</th>
<th>12</th>
</tr>
</thead>
<tbody>
<tr>
<td>$\pi_n$</td>
<td>2</td>
<td>3</td>
<td>5</td>
<td>7</td>
<td>11</td>
<td>13</td>
<td>17</td>
<td>19</td>
<td>23</td>
<td>29</td>
<td>31</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>$n$</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>8</th>
<th>9</th>
<th>10</th>
<th>11</th>
<th>12</th>
<th>13</th>
<th>14</th>
<th>...</th>
</tr>
</thead>
<tbody>
<tr>
<td>$\prod \pi_n$</td>
<td>$\pi_1 \pi_2 \pi_3 \pi_4 \pi_5 \pi_6 \pi_7 \pi_8 \pi_9 \pi_{10} \pi_{11} \pi_{12} \pi_{13} \pi_{14} \cdots$</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

1840 primes are generated by the quadratic $P_k = k^2 - k + 41$, $k = 1 : 40$. But why?
We shall use the convenient notation $\pi_n$ for the prime numbers, indexed by $\mathbb{N}$. The first 12 primes ($n = 1, \ldots, 12$) are $\pi_n = 2, 3, 5, 7, 11, 13, 17, 19, 23, 29, 31, 37$. Since, $4 = 2^2$ and $6 = 2 \cdot 3$ may be factored, $4, 6 \notin \mathbb{P}$ (read as: $4$ and $6$ are not in the set of primes). Given this definition, multiples of a prime, i.e., $[2, 3, 4, 5, \ldots] \cdot \pi_k$ of any prime $\pi_k$, cannot be prime. It follows that all primes except 2 must be odd and every integer $N$ is unique in its factorization.

_Coprimed_ are two numbers with no common prime factors. For example, $4 = 2 \cdot 2$ and $6 = 2 \cdot 3$ are _not_ coprime, as they have 2 as a common factor, whereas $21 = 3 \cdot 7$ and $10 = 2 \cdot 5$ are. By definition all pairs of primes are coprime. We shall use the notation $m \perp n$ to indicate that $m, n$ are coprime. The ratio of two coprimed is reduced, as it has no factors to cancel. The ratio of two numbers that are not coprime may always be reduced by cancelling the common factors. This is called the _reduced form_, or a _irreducible fraction_. When doing numerical work, for computational accuracy it is always beneficial to work with the reduced form.

The _fundamental theorem of arithmetic_ states that each integer may be uniquely expressed as a product of primes. The _Prime Number Theorem_ estimates the mean density of primes over $\mathbb{N}$.

**Integers $\mathbb{Z}$:** These include positive and negative counting numbers and zero. Notionally we might indicate this using _set notation_ as $\mathbb{Z} = -\mathbb{N}, \{0\}, \mathbb{N}$. Read this as The integers are in the set composed of the negative of the natural numbers $-\mathbb{N}$, zero, and $\mathbb{N}$. Note that $\mathbb{N} \subset \mathbb{Z}$.

**Rational numbers $\mathbb{Q}$:** These are defined as numbers formed from the ratio of two integers. Given two numbers $n, d \in \mathbb{N}$, then $n/d \in \mathbb{Q}$. Since $d$ may be 1, it follows that the rationals include the counting numbers as a subset. For example, the rational number $3/1 \in \mathbb{N})$.

The main utility of rational numbers is that they can efficiently approximate any number on the real line, to any precision. For example the rational approximation $\pi \approx 22/7$, has a relative error of $\approx 0.04\%$.

**Fractional number $\mathbb{F}$:** A fractional number $\mathbb{F}$ is defined as the ratio of coprimed. If $n, d \in \mathbb{P}$, then $n/d \in \mathbb{F}$. Given this definition, $\mathbb{F} \subset \mathbb{Q} = \mathbb{Z} \cup \mathbb{F}$. Because of the powerful approximating power of rational numbers, the fractional set $\mathbb{F}$ has special utility. For example $\pi \approx 22/7$, $1/\pi \approx 7/22$ (to 0.04%), $e \approx 19/7$ to 0.15%, and $\sqrt{2} \approx 7/5$ to 1%.

**Irrational numbers $\mathbb{I}$:** Every real number that is not rational is _irrational_ ($\mathbb{Q} \perp \mathbb{I}$). Irrational numbers include $\pi, e$ and the square roots of primes. These are decimal numbers that never repeat, thus requiring infinite precision in their representation. Such numbers cannot be represented on a computer, as they would require an infinite number of bits (precision).

The rationals $\mathbb{Q}$ and irrationals $\mathbb{I}$ split the reals ($\mathbb{R} = \mathbb{Q} \cup \mathbb{I}$), thus each is a subset of the reals ($\mathbb{Q} \subset \mathbb{R}$, $\mathbb{I} \subset \mathbb{R}$). This relation is analogous to that of the integers $\mathbb{Z}$ and fractions $\mathbb{F}$, which split the rationals ($\mathbb{Q} = \mathbb{Z} \cup \mathbb{F}$, $\mathbb{Z} \perp \mathbb{F}$) (thus each is a subset of the rationals ($\mathbb{Z} \subset \mathbb{Q}$, $\mathbb{F} \subset \mathbb{Q}$)).

Irrational numbers ($\mathbb{I}$) were famously problematic for the Pythagoreans, who incorrectly theorized that all numbers were rational. Like $\infty$, irrational numbers required mastering a new and difficult concept before they could even be defined: It was essential to understand the factorization of counting numbers into primes (i.e., the fundamental theorem of arithmetic) before the concept of irrationals could be sorted out. Irrational numbers could only be understood once limits were mastered.

As discussed in Sect. Lec. 1.3.5 (p. 45), fractions can approximate any irrational number with arbitrary accuracy. Integers are also important, but for a very different reason. All numerical
computing today is done with \( \mathbb{Q} = \mathbb{F} \cup \mathbb{Z} \). Indexing uses integers \( \mathbb{Z} \), while the rest of computing (flow dynamics, differential equations, etc.) is done with fractionals \( \mathbb{F} \) (i.e., IEEE-754). Computer scientists are trained on these topics, and computer engineers need to be at least conversant with them.

**Real numbers** \( \mathbb{R} \): Real numbers are the union of rational and irrational numbers, namely \( \mathbb{R} = \mathbb{Z} \cup \mathbb{F} \cup \mathbb{I} \). Lengths in Euclidean geometry are reals. Many people assume that IEEE 754 floating point numbers (c1985) are real (i.e., \( \mathbb{R} \)). In fact they are rational (\( \mathbb{Q} = \{ \mathbb{F} \cup \mathbb{Z} \} \)) approximations to real numbers, designed to have a very large dynamic range. The hallmark of fractional numbers \( \mathbb{F} \) is their power in making highly accurate approximations of any real number.

Using Euclid’s compass and ruler methods, one can make line length proportionally shorter or longer, or (approximately) the same. A line may be made be twice as long, an angle bisected. However, the concept of an integer length in Euclid’s geometry was not defined.\(^{19}\) Nor can one construct an imaginary or complex line, as all lines are assumed to be real.

Real numbers were first fully accepted only after set theory was developed by Cantor (1874) (Stillwell, 2010, pp. 461). At first brush, this seems amazing, given how widely accepted real numbers are today. In some sense they were accepted by the Greeks, as lengths of real lines.

**Complex numbers** \( \mathbb{C} \): Complex numbers are best defined as ordered pairs of real numbers.\(^{20}\) The word “complex,” as used here, does not mean that the numbers are complicated or difficult. They are also know as “imaginary” numbers, but this does not mean the numbers disappear. Complex numbers are quite special in engineering mathematics, as roots of polynomials. The most obvious example is the quadratic formula, for the roots of polynomials of degree 2, having coefficients \( \mathbb{C} \). All real numbers have a natural order on the real line. Complex numbers do not have a natural order. For example, \( j > 1 \) makes no sense.

Today the common way to write a complex number is using the notation \( z = a + bj \in \mathbb{C} \), where \( a, b \in \mathbb{R} \). Here \( 1j = \sqrt{-1} \). We also define \( 1j = -1j \) to account for the two possible signs of the square root. Accordingly \( 1j^2 = 1j^2 = -1 \).

Cartesian multiplication of complex numbers follows the basic rules of real algebra, for example, the rules of multiplying two polynomials. Multiplication of two first-degree polynomials (i.e., monomials) gives

\[
(a + bx)(c + dx) = ac + (ad + bc)x + bdx^2
\]

If we substitute \( 1j \) for \( x \), and use the definition \( 1j^2 = -1 \), we obtain the cartesian product of the two complex numbers

\[
(a + bj)(c + dj) = ac - bd + (ad + bc)j.
\]

Thus multiplication and division of complex numbers obey the usual rules of algebra. However there is a critical extension, cartesian multiplication only holds when the angles sum to less than \( \pm \pi \), namely the range of the complex plane. When the angles add to more that \( \pm \pi \), one must use polar coordinates, where the angles add, for angles outside of of \( \pm \pi \) (Boas, 1987, p. 8). This is particularly striking for the Laplace transform of a delay (Table F.3, p. 278).

Complex number can be challenging, providing unexpected results. For example it is not obvious that \( \sqrt{3 + 4j} = \pm(2 + j) \).

---

\(^{19}\) As best I know.

\(^{20}\) A polynomial \( a + bx \) and a 2-vector \( [a, b]^T = \begin{bmatrix} a \\ b \end{bmatrix} \) are also examples of ordered pairs.
Exercise: Verify. Sol: Squaring both sides $\sqrt{3 + 4j^2} = \pm(2 + j)^2 = 4 - j^2 + 4j = 3 + 4j$.

An alternative to cartesian multiplication of complex numbers is to work in polar coordinates. The polar form of complex number $z = a + bj$ is written in terms of its magnitude $\rho = \sqrt{a^2 + b^2}$ and angle $\theta = \angle z = \tan^{-1} z = \arctan z$, as

$$z = \rho e^{\theta j} = \rho (\cos \theta + j \sin \theta).$$

From the definition of the complex natural log function

$$\ln z = \ln \rho e^{\theta j} = \ln \rho + \theta j,$$

which is important, even critical, in engineering calculations.\(^{21}\) When the angles of two complex numbers are greater that $\pm \pi$, one must use polar coordinates. It follows that when computing the phase, this is much different than the single- and double-argument $\angle \theta = \arctan(z)$ function.

The polar representation make clear the utility of a complex number: Its magnitude scales while its angle $\Theta$ rotates. The property of scaling and rotating is what makes complex numbers useful in engineering calculations. This is especially obvious when dealing with impedances, which have complex roots with very special properties, as discussed in Sect. 1.4.7 (p. 79).

Matrix representation: An alternative way to represent complex numbers is in terms of 2x2 matrices. This relationship is defined by the mapping from a complex number to a 2x2 matrix

$$a + bj \leftrightarrow \begin{bmatrix} a & -b \\ b & a \end{bmatrix}, \quad 1 \leftrightarrow \begin{bmatrix} 1 & 0 \\ 0 & 1 \end{bmatrix}, \quad 1j \leftrightarrow \begin{bmatrix} 0 & -1 \\ 1 & 0 \end{bmatrix}, \quad e^{\theta j} \leftrightarrow \begin{bmatrix} \sin(\theta) & -\cos(\theta) \\ \cos(\theta) & \sin(\theta) \end{bmatrix} \quad (1.3)$$

The conjugate of $a + bj$ is then defined as $a - bj \leftrightarrow \begin{bmatrix} a & b \\ -b & a \end{bmatrix}$. By taking the inverse of the 2x2 matrix (assuming $|a + bj| \neq 0$), one can define the ratio of one complex number by another. Until you try out this representation, it may not seem obvious, or even that it could work.

This representation proves that $1j$ is not necessary when defining a complex number. What $1j$ can do is to conceptually simplify the algebra. It is worth your time to become familiar with the matrix representation, to clarify any possible confusions you might have about multiplication and division of complex numbers. This matrix representation can save you time, heartache and messy algebra. Once you have learned how to multiply two matrices, it’s a lot simpler than doing the complex algebra. In many cases we will leave the results of our analysis in matrix form, to avoid the algebra altogether.\(^{22}\) Thus both representations are important. More on this topic may be found in Chapter 2.

History of complex numbers: It is notable how long it took for complex numbers to be accepted (1851), relative to when they were first introduced by Bombelli (16th century CE). One might have thought that the solution of the quadratic, known to the Chinese, would have settled this question. It seems that complex integers (aka, Gaussian integers) were accepted before non-integral complex numbers. Perhaps this was because real numbers ($\mathbb{R}$) were not accepted (i.e., proved to exist, thus mathematically defined) until the development of set theory in the late 19th century to sort out a proper definition of the real number, due to the existence of irrational numbers.

---

\(^{21}\) Chapter 2 discusses the definition of the phase, i.e., how is it computed (i.e., $\arctan(e^{\theta j})$, $\arctan2(x, y)$), and the importance of the unwrapped phase, as in the example (Table F.3, p. 278) $\delta(t - \tau) \leftrightarrow e^{-\tau j}$.

\(^{22}\) Sometimes we let the computer do the final algebra, numerically, as 2x2 matrix multiplications.
Exercise: Using Matlab/Octave, verify that

\[
\frac{a + bj}{c + dj = \frac{ab + bd + (bc - ad)j}{c^2 + d^2}} \leftrightarrow \left[ \begin{array}{cc}
 a & -b \\
 b & a
\end{array} \right] \left[ \begin{array}{cc}
 c & -d \\
 d & c
\end{array} \right]^{-1} = \left[ \begin{array}{cc}
 a & -b \\
 b & a
\end{array} \right] \left[ \begin{array}{cc}
 c & d \\
 -d & c
\end{array} \right] \frac{1}{c^2 + d^2}. \quad (1.4)
\]

Sol: The best way may be using numbers. Below is symbolic code, independent of numerics:

```matlab
syms a b c d A B
A=[a -b;b a];
B=[c -d;d c];
C=A*inv(B)
```

Numerical taxonomy:

A simplified taxonomy of numbers is given by the mathematical sentence

\[
\pi_k \in \mathbb{P} \subset \mathbb{N} \subset \mathbb{Z} \subset \mathbb{Q} \subset \mathbb{R} \subset \mathbb{C}.
\]

This sentence says:

1. Every prime number \( \pi_k \) is in the set of primes \( \mathbb{P} \),
2. which is a subset of the set of counting numbers \( \mathbb{N} \),
3. which is a subset of the set of integers \( \mathbb{Z} = -\mathbb{N}, 0, \mathbb{N} \),
4. which is a subset of the set of rationals \( \mathbb{Q} \) (ratios of counting numbers \( \mathbb{N} \)),
5. which is a subset of the set of reals \( \mathbb{R} \),
6. which is a subset of the set of complex numbers \( \mathbb{C} \).

The rationals \( \mathbb{Q} \) may be further decomposed into the fractionals \( \mathbb{F} \) and the integers \( \mathbb{Z} = \mathbb{Q} \cup \mathbb{F} \), and the reals \( \mathbb{R} \) into the rationals \( \mathbb{Q} \) and the irrationals \( \mathbb{I} \) (\( \mathbb{R} = \mathbb{I} \cup \mathbb{Q} \)). This classification nicely defines all the numbers used in engineering and physics.

The taxonomy structure may be summarize with the single compact sentence, starting with the prime numbers \( \pi_k \) and ending with complex numbers \( \mathbb{C} \):

\[
\pi_k \in \mathbb{P} \subset \mathbb{N} \subset \mathbb{Z} \subset (\mathbb{Z} \cup \mathbb{F} = \mathbb{Q}) \subset (\mathbb{Q} \cup \mathbb{I} = \mathbb{R}) \subset \mathbb{C}.
\]

As discussed in Appendix A (p. 251), all numbers may be viewed as complex. Namely every real number is complex if we take the imaginary part to be zero (Boas, 1987). For example, \( 2 \in \mathbb{P} \subset \mathbb{C} \). Likewise every purely imaginary number (e.g., \( 0 + 1j \)) is complex with zero real part.

Finally, note that complex numbers \( \mathbb{C} \), much like vectors, do not have “rank-order,” meaning one complex number cannot be larger or smaller than another. It makes no sense to say that \( j > 1 \) or \( j = 1 \) (Boas, 1987). The real and imaginary parts and the magnitude and phase, have order. If time \( t \) were complex, there could be no yesterday and tomorrow.\(^{23}\)

\(^{23}\)One can define \( \xi = x + 1j \) \( ct \) to be complex \((x, t \in \mathbb{R})\), with \( x \) in meters [m], \( t \) is in seconds [s], and the speed of light \( c_o \) [m/s].
Applications of integers

The most relevant question at this point is “Why are integers important?” First we count with them, so we can keep track of “how much.” But there is much more to numbers than counting: We use integers for any application where absolute accuracy is essential, such as banking transactions (making change), the precise computing of dates (Stillwell, 2010, p. 70) and location (“I’ll meet you at 34 and Vine at noon on Jan. 1, 2034.”), building roads or buildings out of bricks (objects built from a unit size).

To navigate we need to know how to predict the tides, the location of the moon and sun, etc. Integers are important because they are precise: Once a month there is a full moon, easily recognizable. The next day its slightly less than full. If one could represent our position as integers in time and space, we would know exactly where we are at all times. But such an integral representation of our position or time is not possible.

The Pythagoreans claimed that all was integer. From a practical point of view, it seems they were right. Today all computers compute floating point numbers as fractionals. However, in theory they were wrong. The difference is a matter of precision.

Numerical Representations of \( \mathbb{I}, \mathbb{R}, \mathbb{C} \): When doing numerical work, one must consider how we may compute within the set of reals (i.e., which contain irrationals). There can be no irrational number representation on a computer. The international standard of computation, IEEE floating point numbers,\(^{24}\) are actually rational approximations. The mantissa and the exponent are both integers, having sign and magnitude. The size of each integer depends on the precision of the number being represented. An IEEE floating-point number is rational because it has a binary (integer) mantissa, multiplied by 2 raised to the power of a binary (integer) exponent. For example \( \pi \approx a2^b \) with \( a, b \in \mathbb{Z} \). In summary, IEEE floating-point numbers are not, and cannot, be irrational, because numerical representations would imply an infinite number of bits.

True floating point numbers contain irrational numbers, which must be approximated by rational numbers. This leads to the concept of fractional representation, which requires the definition of the mantissa, base and exponent, where both the mantissa and the exponent are signed. Numerical results must not depend on the base. One could dramatically improve the resolution of the numerical representation by the use of the fundamental theorem of arithmetic (Section 1.3.2, page 37). For example one could factor the exponent into its primes and then represent the number as \( a2^35^57^7 \) \((a, b, c, d, e \in \mathbb{Z})\), etc. Such a representation would improve the resolution of the representation. But even so, the irrational numbers would be approximate. For example, base ten\(^{25}\) is natural using this representation since \( 10^n = 2^n5^n \). Thus

\[
\pi \cdot 10^5 \approx 314159.27 \ldots = 3 \cdot 2^55^5 + 1 \cdot 2^45^4 + 4 \cdot 2^35^3 + \ldots + 9 \cdot 2^05^0 + 2 \cdot 2^{53}5^{-1} \ldots.
\]

If we approximate \( \pi \) by \( 22/7 \), then according to the Matlab/Octave \texttt{DEC2BIN()} routine, the binary representation is

\[
\pi \cdot 2^{17} \approx 13107210 \cdot 22/7 = 110,0100,1001,0010,01012,
\]

where 1 and 0 are multipliers of powers of 2, which are then added together as follows:

\[
2^{18} + 2^{17} + 2^{14} + 2^{11} + 2^8 + 2^5 + 2^2 + 2^0.
\]

\(^{24}\text{IEEE 754: http://www.h-schmidt.net/FloatConverter/IEEE754.html.}\)

\(^{25}\text{Base 10 is the natural world-wide standard simply because we have 10 fingers.}\)
In base 16 (i.e., hexadecimal) $2^{17} \cdot 22/7 = 2^{18} \cdot 8_{16}/7_{16}$.

Computers keep track of the decimal point using the exponent, which in this case is the factor $2^{17} = 131072_{10}$. The concept of the decimal point is replaced by an integer, having the desired precision, and a scale factor of any base (radix). This scale factor may be thought of as moving the decimal point to the right (larger number) or left (smaller number). The mantissa “fine-tunes” the value about a scale factor (the exponent).

Example: $x = 2^{17} \times 22/7$, using IEEE 754 double precision, $x = 411, 940.5625_{10} = 2^{54} \times 1198372 = 0, 10010, 00, 110010, 010010, 010010_2 = 0x48c92492_{16}$. The exponent is $2^{18}$ and the mantissa is $4, 793, 490_{10}$. The commas in the binary (0,1) string are to help visualize the quasi-periodic nature of the bit-stream. The numbers are stored in a 32 bit format, with 1 bit for sign, 8 bits for the exponent and 23 bits for the mantissa. Perhaps a more instructive number is $x = 4, 793, 490.0 = 0, 100, 1010, 100, 100, 100, 100, 100_2 = 0x4a924924_{16}$, which has a repeating binary bit pattern of $((100))_3$, broken by the scale factor $0x4a$. Even more symmetrical is $x = 0x24, 924, 924_{16} = 00, 100, 100, 100, 100, 100, 100, 100_2 = 6.344, 131, 191, 146, 9 \times 10^{-17}$ In this example the repeating pattern is clear in the hex representation as a repeating $((942))_3$, as represented by the double brackets, with the subscript indicating the period, in this case, three digits. As before, the commas are to help with readability and have no other meaning.

The representation of numbers in not unique. For example, irrational complex numbers have approximate rational representations (i.e., $\pi \approx 22/7$). A better example is complex numbers $z \in \mathbb{C}$, which have many representations, as a pair of reals (i.e., $z = (x, y)$), or by Euler’s formula, and matrices ($\theta \in \mathbb{R}$)
\[
 e^{j\theta} = \cos \theta + j \sin \theta \leftrightarrow \begin{bmatrix}
 \cos \theta & -\sin \theta \\
 \sin \theta & \cos \theta 
\end{bmatrix}.
\]

At a higher level, differentiable functions, aka, analytic functions, may be represented by a Taylor series, limited by the RoC.

**Pythagoreans, Integers**  The integer is the cornerstone of the Pythagorean doctrine, so much so that it caused a fracture within the Pythagoreans when it was discovered that not all numbers are rational. The famous proof of irrational numbers is known as the spiral of Theodorus: the short side is 1 and the long side is recursively incremented by one, using a simple compass-ruler construction to maintain the right triangle.

To form triangles with perfect $90^\circ$ angles, the lengths need to satisfy Eq. 1.1. Such triangles are useful in constructing buildings or roads made from bricks of uniform size.

**Public-key Security:** An important application of prime numbers is public-key encryption, essential for internet security applications (e.g., online banking). Most people assume encryption is done by a personal login and passwords. Passwords are fundamentally insecure, for many reasons. Deencryption depends on factoring large integers, formed from products of primes having thousands of bits.\(^{27}\) The security is based on the relative ease of multiplying large primes, along with the virtual impossibility of factoring their products.

\(^{26}\)http://www.h-schmidt.net/FloatConverter/IEEE754.html

\(^{27}\)It would seem that public-key encryption could work by having two numbers with a common prime, and then using the Euclidean algorithm, the greatest common divisor (GCD) could be worked out. One of the integers could be the public-key and the second could be the private key.
When a computation is easy in one direction, but its inverse is impossible, it is called a trapdoor function. We shall explore trapdoor functions in Chapter 2. If everyone were to switch from passwords to public-key encryption, the internet would be much more secure.

Puzzles: Another application of integers is imaginative problems that use integers. An example is the classic Chinese four stone problem: “Find the weight of four stones that can be used with a scale to weigh anything (e.g., salt, gold) between 0, 1, 2, . . . , 40 [gm].” As with the other problems, the answer is not as interesting as the method, since the problem may be easily recast into a related one. This type of problem can be found in airline magazines as amusement on a long flight. This puzzle is best cast as a linear algebra problem, with integer solutions. Again, once you know the trick, it is “easy.”

1.3.2 Lec 3: The role of physics in mathematics

Bells, chimes and eigen-modes: Integers naturally arose in art, music and science. Examples include the relations between musical notes, the natural eigen-modes (tones) of strings and other musical instruments. These relations were so common and well studied, it appeared that to understand the physical world (aka, the Universe), one needed to understand integers. This was a seductive view, but not actually correct. As will be discussed in Sections 1.4.1 (p. 56) and 3.1.1 (p. 197), it is best to view the relationship between acoustics, music and mathematics as historical, since these topics played such an important role in the development of mathematics. Also interesting is the role that integers play in quantum mechanics, also based on eigen-modes, but in this case, those of atoms. Eigen-modes follow from solutions of the wave equation, which has natural delay due to d’Alembert’s solution, along with reflecting boundary conditions (Eq. 1.176, p. 160), resulting in roots of the characteristic equation.

Engineers are so accustomed to working with real (or complex) numbers, the distinction between real (i.e., irrational) and fractional numbers is rarely acknowledged. Integers on the other hand arise in many contexts. One cannot master programming computers without understanding integer, hexadecimal, octal, and binary representations, since all numbers in a computer are represented in numerical computations in terms of rationals \( \mathbb{Q} = \mathbb{Z} \cup \mathbb{F} \).

As discussed in Section 1.3.1 (p. 29), the primary reason integers are so important is their absolute precision. Every integer \( n \in \mathbb{Z} \) is unique,\(^29\) and has the indexing property, which is essential for making lists that are ordered, so that one can quickly look things up. The alphabet also has this property (e.g., a book’s index).

Because of the integer’s absolute precision, the digital computer overtook the analog computer, once it was practical to make logic circuits that were fast. From 1946 the first digital computer was thought to be the University of Pennsylvania’s Eniac. We now know that the code-breaking effort in Bletchley Park, England, under the guidance of Alan Turing, created the first digital computer (The Colossus) used to break the WWII German “Enigma” code. Due to the high secrecy of this war effort, the credit was only acknowledged in the 1970s when the project was finally declassified.

There is zero possibility of analog computing displacing digital computing, due to the importance of precision (and speed). But even with binary representation, there is a non-zero probability of error, for example on a hard drive, due to physical noise. To deal with this, error correcting codes

\(^{28}\)Whenever someone tells you something is “easy,” you should immediately appreciate that it is very hard, but once you learn a concept, the difficulty evaporates.

\(^{29}\)Check out the history of \(1729 = 1^3 + 12^3 = 9^3 + 10^3\).
have been developed, reducing the error by many orders of magnitude. Today error correction is a science, and billions of dollars are invested to increase the density of bits per area, to increasingly larger factors. A few years ago the terabyte drive was unheard of; today it is the standard. In a few years petabyte drives will certainly become available. It is hard to comprehend how these will be used by individuals, but they are essential for on-line (cloud) computing.

The role of mathematics in physics

Modern mathematics is built on a hierarchical construct of fundamental theorems, as summarized in Table 1.1. The importance of such theorems cannot be overemphasized. Gauss’ and Stokes’ laws play a major role in understanding and manipulating Maxwell’s equations. Every engineering student needs to fully appreciate the significance of these key theorems. If necessary, memorize them. But that will not do over the long run, as each and every theorem must be fully understood. Fortunately most students already know several of these theorems, but perhaps not by name. In such cases, it is a matter of mastering the vocabulary.

The theorems are naturally organized and may be thought of in terms of Stillwell’s three streams. For Stream 1 there is the fundamental theorem of arithmetic and the prime number theorem. For Stream 2 there is the fundamental theorem of algebra while for Stream 3 there are a host of theorems on calculus, ordered by their dimensionality. Some of these theorems verge on trivial (e.g., the fundamental theorem of arithmetic). Others are more challenging, such as the fundamental theorem of vector calculus and Green’s theorem.

Complexity should not be confused with importance. Each of these theorems is, as stated, fundamental. Taken as a whole, they are a powerful way of summarizing mathematics.

Stream 1: Prime number theorems

There are three fundamental theorems about primes,

1. **The fundamental theorem of arithmetic**: This states that every integer \( n \in \mathbb{Z} \) may be uniquely factored into prime numbers. This raises the question of the meaning of factor (split into a product). The product of two integers \( m, n \in \mathbb{Z} \) is \( mn = \sum_m n = \sum_n m \). For example \( 2 \times 3 = 2 + 2 + 2 = 3 + 3 \).

2. **The prime number theorem**: One would like to know how many primes there are. That is easy: \( |\mathbb{P}| = \infty \) (the size of the set of primes is infinite). One way of asking this questions is *What is the average density of primes, in the limit as \( n \to \infty \)?* This question was answered, for all practical purposes, by Gauss, who in his free time, computed the first three million primes by hand. He discovered that, to a good approximation, the primes are equally likely on a log scale. This is nicely summarized by the jingle, attributed to the mathematician Pafnuty Chebyshev (Stillwell, 2010, p. 585)

   Chebyshev said, and I say it again: **There is always a prime between \( n \) and \( 2n \)**

   which nicely summarizes theorem 3.

3. **There an infinite number of primes**.

When the ratio of two frequencies (pitch) is 2, the relationship is called an octave. Thus we might say, with a slight stretch of terminology, there is at least one prime per octave. Thus one might
1. Fundamental theorems of:
   
   (a) **Number systems: Stream 1**
   - arithmetic
   - prime number

   (b) **Geometry: Stream 2**
   - algebra

   (c) **Calculus: Stream 3\(^a\)**
   - Leibniz $\mathbb{R}^1$
   - complex $\mathbb{C} \subset \mathbb{R}^2$
   - vectors $\mathbb{R}^3, \mathbb{R}^n, \mathbb{R}^\infty$
     - Gauss’s law (divergence theorem)
     - Stokes’s law (curl theorem, or Green’s theorem)
     - Vector calculus (Helmholtz’s theorem)

2. Other key concepts:
   
   - Complex analytic functions (complex roots are finally accepted!)
     - Complex Taylor series (complex analytic functions)
     - Region of convergence (RoC) of complex analytic series (p. 71)
     - Laplace transform, and its inverse
     - Causal time $\Rightarrow$ complex frequency $s$
     - Cauchy integral theorem
     - Residue integration (i.e., Green’s thm in $\mathbb{R}^2$)

   - Riemann mapping theorem (Gray, 1994; Walsh, 1973)
   - Complex impedance (Ohm’s law) (Kennelly, 1893)

---

1.3. **STREAM 1: NUMBER SYSTEMS (10 LECTURES)**

W wonder about the maximum number of primes per octave. In modern music the octave is further divided into 12 intervals called semitones (factors), equal to the $\sqrt[12]{2}$. The product of 12 semitones is an octave. Thus one might ask how many primes there are per semitone? In the end, it is a question of the density of primes on a log (i.e., ratio) scale.

**Stream 2: Fundamental theorem of algebra**

This theorem states that every polynomial in $x$ of degree $N$

$$P_N(x) = \sum_{k=0}^{N} a_k x^k \tag{1.5}$$

has at least one root (Section 3.2.1, p. 200). When a common root is factored out, the degree of the polynomial is reduced by 1. Applied recursively, a polynomial of degree $N$ has $N$ roots. Note there are $N + 1$ coefficients (i.e., $[a_N, a_{N-1}, \ldots, a_0]$).

**Stream 3: Fundamental theorems of calculus**

In Sections 1.6.11 and 1.6.12 we will deal with each of the theorems for Stream 3, where we consider the several fundamental theorems of integration, starting with Leibniz’s formula for integration on the real line ($\mathbb{R}$), then progressing to complex integration in the complex plane ($\mathbb{C}$) (*Cauchy’s theorem*), which is required for computing the inverse Laplace transform. Gauss’ and Stokes’ laws for $\mathbb{R}^2$ require closed and open surfaces, respectively. One cannot manipulate Maxwell’s equations, fluid flow, or acoustics without understanding these theorems. Any problem that deals with the wave equation in more than one dimension requires an understanding of these theorems, thus are the basis of the derivation of the Kirchhoff voltage and current laws. The $\nabla$ symbol is pronounced as “del” (preferred) or “nabla.”

Finally we define the four basic vector operations based on the $\nabla$ “operator:” The gradient $\nabla()$, divergence $\nabla \cdot ()$ curl $\nabla \times ()$ and the Laplacian $\nabla^2() = \nabla \cdot \nabla()$. The first three operations are defined in terms of integral operations on a surface in 1, 2 or 3 dimensions, by taking the limit as that surface, and the volume contained within, goes to zero. These three differential operators are essential, to fully understand Maxwell’s equations, the crown jewel of mathematical physics. Hence mathematics plays a key role in physics, as does physics in math.

**Other key concepts**

Besides the widely recognized fundamental theorems for the three streams, there are a number of equally important that have not yet been labeled as “fundamental.”

The widely recognized *Cauchy integral theorem* is an excellent example, since it is a stepping stone to Green’s theorem and the fundamental theorem of complex calculus. In Section ?? (p. ??) we clarify the contributions of each of these special theorems.

Once these fundamental theorems of integration (Stream 3) have been mastered, the student is ready for the complex frequency domain, which takes us back to Stream 2 and the complex frequency plane ($s = \sigma + \omega j \in \mathbb{C}$). While the Fourier and Laplace transforms are taught in mathematics courses, the concept of complex frequency is rarely mentioned. The complex frequency domain (p. 1) and causality are fundamentally related (Sects. 1.5.6–1.5.8, p. 122–124), and

$^{30}$It is not clear what it takes to reach this more official sounding category.
are critical for the analysis of signals and systems, and especially for the concept of impedance (Sect. 1.5.3, p. 112).

Without the concept of time and frequency, one cannot develop an intuition for the Fourier and Laplace transforms, especially within the context of engineering and mathematical physics. The Fourier transform covers signals, while the Laplace transform describes systems. Separating these two concepts, based on their representations as Fourier and Laplace transforms, is an important starting place for understanding physics and the role of mathematics. However, these methods, by themselves, do not provide the insight into physical systems, necessary to be productive, or better, creative with these tools. One needs to master the tools of differential equations, and then partial differential equations, to fully appreciate the world that they describe. Electrical and mechanical networks, composed of inductors, capacitors and resistors, are isomorphic to mechanical systems composed of masses, springs and dash-pots. Newton’s laws are analogous to those of Kirchhoff, which are the rules needed to analyze simple physical systems composed of linear (and nonlinear) sub-components. When lumped-element systems are taken to the limit, in several dimensions, we obtain Maxwell’s partial differential equations, or the laws of continuum mechanics, and beyond.

The ultimate goal of this book is to make you aware of and productive in using these tools. This material can be best absorbed by treating it chronologically through history, so you can see how this body of knowledge came into existence, through the minds and hands of Galileo, Newton, Maxwell and Einstein. Perhaps one day you too can stand on the shoulders of the giants that went before you.

1.3.3 Lec 4: Prime numbers

If someone asked you for a theory of counting numbers, I suspect you would laugh and start counting. It sounds like either a stupid question, or a bad joke. Yet integers are a rich topic, so the question is not even slightly dumb. It is somewhat amazing that even birds and bees can count. While I doubt birds and bees can recognize primes, cicadas and other insects only crawl out of the ground in prime number cycles, (e.g., 13 or 17 year cycles). If you have ever witnessed such an event (I have), you will never forget it. Somehow they know. Finally, there is an analytic function, first introduced by Euler, based on his analysis of the sieve, now known as the Riemann zeta function $\zeta(s)$, which is complex analytic, with its poles at the logs of the prime numbers. The exact relationship between the primes and the poles will be discussed in Section 3.4.2 (p. 215). The properties of this function are truly amazing, even fun.$^{31}$ Many of the questions and answers about primes go back to at least the early Chinese (c1500 BCE) (Stillwell, 2010).

The importance of prime numbers: Likely the first insight into the counting numbers started with the sieve, shown in Fig. 1.6. A sieve answers the question “How can one identify the prime numbers?” The answer comes from looking for irregular patterns in the counting numbers, by playing the counting numbers against themselves.

A recursive sieve method for finding primes was first devised by the Greek Eratosthenes$^{32}$ (276-194, BCE), and summarized in Fig. 1.6.

For example, starting from $\pi_1 = 2$ one strikes out $2 \cdot 2, 2 \cdot 3, 2 \cdot 4, 2 \cdot 5, \ldots, 2 \cdot \sqrt{N}$. By definition the multiples are products of the target prime (2 in our example) and every another integer ($n \geq 2$).

$^{31}$The Riemann zeta function is known as the million dollar equation as there is a cash reward for a proof of the Riemann Hypothesis.

$^{32}$https://en.wikipedia.org/wiki/Sieve_of_Eratosthenes\#Euler.27s_Sieve
1.3. **STREAM 1: NUMBER SYSTEMS (10 LECTURES)**

In this way all the even numbers are removed in this first iteration. The next remaining integer (3 in our example) is identified as the next (second) prime \( \pi_2 \). Then all the \((N - 2)/2\) multiples of \( \pi_2 = 3 \) are removed. The next remaining number is \( \pi_3 = 5 \), so all multiples of \( \pi_3 = 5 \) are removed (i.e., \( 5, 15, 25 \) etc., \( \cdots \)). This process is repeated until all the numbers of the list have either been canceled or identified as prime.

1. Write \( N \) integers from 2 to \( N - 1 \). Let \( k = 1 \). The first element \( \pi_1 = 2 \) is a prime. Cross out \( n \cdot \pi_n \) (4, 8, 16, 32, \( \cdots \)).

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2. Let \( k = 2 \), \( \pi_2 = 3 \). Cross out \( n\pi_k \) (9, 15, 21, 27, 33, 39, \( \cdots \)).

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3. Let \( k = 3 \), \( \pi_3 = 5 \). Cross out \( n\pi_3 \) (25, 35).

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4. Finally let \( k = 4 \), \( \pi_4 = 7 \). Cross out \( n\pi_4 \) (49). Thus there are 15 primes less than \( N = 50 \): \( \pi_k = 2, 3, 5, 7, 11, 13, 17, 19, 23, 29, 31, 37, 41, 43, 47 \) (highlighted in red).

*Figure 1.6: Sieve of Eratosthenes for \( N = 50 \).*

As the word *sieve* implies, this process takes a heavy toll on the integers, rapidly pruning the non-primes. In four iterations of the sieve algorithm, all the primes below \( N = 50 \) are identified in red. The final set of primes is displayed in step 4 of Fig. 1.6.

Once a prime greater than \( N/2 \) has been identified, the recursion stops, since twice that prime is greater than \( N \), the maximum number under consideration. Once \( \sqrt{N} \) has been reached all the primes have been identified (this follows from the fact that the next prime \( \pi_n \) is multiplied by an integer \( n = 1, \ldots N \)).

When using a computer, memory efficiency and speed are the main considerations. There are various schemes for making the sieve more efficient. For example the recursion \( n\pi_k = (n - 1)\pi_k + \pi_k \) will speed up the process by replacing the multiply with an addition of \( \pi_k \).
Two fundamental theorems of primes: Early theories of numbers revealed two fundamental theorems (there are many more than two), as discussed in Section 1.3.2 and 2.1.1 (p. 183). The first of these is the fundamental theorem of arithmetic, which says that every integer greater than 1 may be uniquely factored into a product of primes

$$n = \prod_{k=1}^{K} \pi_k^{\beta_k}$$

where \( k = 1, \ldots, K \) indexes the integer’s \( K \) prime factors \( \pi_k \in \mathbb{P} \). Typically prime factors appear more than once, for example \( 25 = 5^2 \). To make the notation compact we define the multiplicity \( \beta_k \) of each prime factor \( \pi_k \). For example \( 2312 = 2^3 \cdot 17^2 = \pi_1^3 \pi_7^2 \) (i.e., \( \pi_1 = 2, \beta_1 = 3; \pi_7 = 17, \beta_7 = 2 \)) and \( 2313 = 3^2 \cdot 257 = \pi_2^3 \pi_{55} \) (i.e., \( \pi_2 = 3, \beta_3 = 2; \pi_{55} = 257, \beta_{55} = 1 \)). Our demonstration of this is empirical, using the Matlab/Octave factor \((N)\) routine, which factors \( N \).

What seems amazing is the unique nature of this theorem. Each counting number is uniquely represented as a product of primes. No two integers can share the same factorization. Once you multiply the factors out, the result is unique \((N)\). Note that it’s easy to multiply integers (e.g., primes), but expensive to factor them. And factoring the product of three primes is significantly more difficult than factoring two.

Factoring is much more expensive than division. This is not due to the higher cost of division over multiplication, which is less than a factor of 2. Dividing the product of two primes, given one, is trivial, slightly more expensive that multiplying. Factoring the product of two primes is nearly impossible, as one needs to know what to divide by. Factoring means dividing by some integer and obtaining another integer with remainder zero. Thus one could factor a product of primes \( N = \pi_k \pi_l \) by doing \( M \) divisions, where \( M \) is the number of primes less than \( N \). This assumes the list of primes less than \( N \) are known.

But the utility has to do with the density of primes (the prime number theorem, i.e., Gauss’ hypothesis). If we were simply looking up a few numbers from a short list of primes, it would be easy, but the density of primes among the integers, is logarithmic (>1 per octave, Section 2.1.1, p. 185).

This brings us to the prime number theorem (PNT). The security problem is the reason why these two theorems are so important: 1) Every integer has a unique representation as a product of primes, and 2) the density of primes is high (Section 2.1.1, p. 183 and discussion on p. 37). Security reduces to the “needle in the haystack problem,” the cost of a search. A more formal way to measure the density is known as Shannon entropy, couched in terms of the expected value of the log-probability of events: “What is the probability of finding a prime between \( N \) and \( 2N \)?”

1.3.4 Lec 5: Greatest common divisor (Euclidean algorithm)

The Euclidean algorithm is a method to find the greatest common divisor (GCD) \( k \) between two integers \( n, m \), denoted \( k = \text{gcd}(n, m) \), where \( n, m, k \in \mathbb{N} \). For example \( 15 = \text{gcd}(30, 105) \) since when factored \((30, 105) = (2 \cdot 3 \cdot 5, 7 \cdot 3 \cdot 5) = 3 \cdot 5 \cdot (2, 7) = 15 \cdot (2, 7) \). The Euclidean algorithm was known to the Chinese (i.e., not discovered by Euclid) (Stillwell, 2010, p. 41).
Why is the GCD important? The GCD is important precisely because of the fundamental difficulty of factoring large integers into their primes. Computing the GCD using the Euclidean algorithm is easy compared to factoring, which is extremely expensive. This utility surfaces when the two numbers are composed of very large primes. When two integers have no common factors they are said to be coprime, thus their GCD is 1. The ratio of two integers which are coprime is automatically in reduced form (they have no common factors).

For example, $4/2 \in \mathbb{Q}$ is not reduced since $2=\gcd(4,2)$ (with no remainder). Canceling out the common factor 2, gives the reduced form $2/1=2 \in \mathbb{N}$. Thus if we wish to form the ratio of two integers, first compute the gcd, and remove it from the two numbers, to form the ratio. This assures the rational number is in its reduced form. If the GCD were very large, it is obvious that the common factor must be removed to greatly simplify further computation.

Exercise: Divide 10 into 99, and discuss where the GCD is used. Sol: When we divide a smaller number into a larger one, we must find the GCD and a remainder. For example $99/10 = 9 + 9/10$ has a gcd of 9 and a remainder of 9/10. Thus we all learned how to compute the GCD in grade school, when we learned long division.

An example: Take the two integers $[873, 582]$. In factored form these are $[\pi_{25} \cdot 3^2, \pi_{25} \cdot 3 \cdot 2]$. Given the factors, we see that the largest common factor is $\pi_{25} \cdot 3 = 291$ ($\pi_{25} = 97$). When we take the ratio of the two numbers this common factor cancels

$$\frac{873}{582} = \frac{\pi_{25} \cdot 3 \cdot 3}{\pi_{25} \cdot 3 \cdot 2} = \frac{3}{2} = 1.5.$$

Of course if we divide 582 into 873 this we will numerically obtain the answer $1.5 \in \mathbb{F}$.

Exercise: Show that in Matlab $\text{rat}(873/582) = 1+1/(-2)$ gives the wrong answer, since $\text{factor}(873) = 3 \cdot 3 \cdot 97$ and $\text{factor}(582) = 2 \cdot 3 \cdot 97$, $3/2 = 1 + 1/2$ is the correct answer. But due to rounding methods, it is not $3/2$. As an example, in Matlab/Octave $\text{rat}(3/2) = 2+1/(-2)$. One would expect $\text{rat}(3/2) = 1+1/2$. Matlab’s rat command using rounding rather than the floor function, which explains the difference. When the rat function produces negative numbers, rounding is being used to find the integer part.

Euclidean algorithm: The algorithm is best explained by a trivial example: Let the two numbers be 6, 9. At each step the smaller number (6) is subtracted from the larger (9) and the difference (the remainder) and the smaller numbers are saved. This process continues until the two resulting numbers are equal, at which point the GCD equals that final number. If we were to take one more step, the final numbers would be the gcd and zero. For our example step 1 gives 9-6=3, leaving 6 and 3. Step 2 gives 6-3=3 and 3. Since the two numbers are the same, the GCD=3. If we take one more difference we obtain (3,0). We can easily verify this result since this example is easily factored (e.g., $3 \cdot 3 \cdot 2 = 3(3,2)$). It may be numerically verified using the Matlab/Octave GCD command $\text{gcd}(6,9)$, which returns 3.

Matrix method: The GCD may be written a matrix recursion given the starting vector $(m_0, n_0)^T$. The recursion is then

$$\begin{bmatrix} m_{k+1} \\ n_{k+1} \end{bmatrix} = \begin{bmatrix} 1 & -1 \\ 0 & 1 \end{bmatrix} \begin{bmatrix} m_k \\ n_k \end{bmatrix}.$$
CHAPTER 1. INTRODUCTION

Greatest Common Divisor: $k = \gcd(m, n)$

- Examples ($m, n, k \in \mathbb{Z}$):
  - $\gcd(13 \cdot 5, 11 \cdot 5) = 5$ (The common 5 is the gcd)
  - $\gcd(13 \cdot 10, 11 \cdot 10) = 10$ (The gcd(130,110) = 10 = $2 \cdot 5$, is not prime)
  - $\gcd(1234, 1024) = 2$ ($1234 = 2 \cdot 617, 1024 = 2^{10}$)
  - $\gcd(\pi m, \pi n) = \pi k$ $k = \gcd(m, n)$ is the part that cancels in the fraction $m/n \in \mathbb{F}$
  - $m/gcd(m,n) \in \mathbb{Z}$

- Coprimes ($m \perp n$) are numbers with no distinct common factors: i.e., $\gcd(m,n)=1$
  - The gcd of two primes is always 1: $\gcd(13,11) = 1, \gcd(\pi m, \pi n) = 1$ ($m \neq n$)
  - $m = 7 \cdot 13, n = 5 \cdot 19 \Rightarrow (7 \cdot 13) \perp (5 \cdot 19)$
  - If $m \perp n$ then $\gcd(m,n) = 1$
  - If $\gcd(m,n) = 1$ then $m \perp n$

- The GCD may be extended to polynomials: e.g., $\gcd(ax^2 + bx + c, \alpha x^2 + \beta x + \gamma)$
  - $\gcd((x - 3)(x - 4), (x - 3)(x - 5)) = (x - 3)$
  - $\gcd(ax^2 - 7x + 12, 3(x^2 - 8x + 15)) = 3(x - 3)$
  - $\gcd(ax^2 - 7x + 12, (3x^2 - 24x + 45)) = 3(x - 3)$
  - $\gcd((x - 2\pi)(x - 4), (x - 2\pi)(x - 5)) = (x - 2\pi)$ (Needs long division)

**Figure 1.7:** The Euclidean algorithm for finding the GCD of two numbers is one of the oldest algorithms in mathematics, and is highly relevant today. It is both powerful and simple. It was used by the Chinese during the Han dynasty (Stillwell, 2010, p. 70) for reducing fractions. It may be used to find pairs of integers that are coprime (their gcd must be 1), and it may be used to identify factors of polynomials by long division. It has an important sister algorithm called the continued fraction algorithm (CFA), that is so similar in concept that Gauss referred to the Euclidean algorithm as the “continued fraction algorithm” (Stillwell, 2010, p. 48).

This recursion continues until $m_{k+1} < n_{k+1}$, at which point $m$ and $n$ must be swapped. Because the operation depends on the output, this is a nonlinear recursion (Postulate P1 (Linear/nonlinear) of Section 3.5.1, p. 224).

The direct method is inefficient because in recursively subtract $n$ many times until the resulting $m$ is less than $n$, as shown in Fig. 2.1 (p. 188). It also must test for $m < n$ at each iteration, and then swap $m$ and $n$ once that condition is met. This recursion is repeated until $m_{k+1} = 0$. At that stage the GCD is $n_{k+1}$. Figure 2.1 (p. 188), along with the above matrix relation, gives insight into the Euclidean Algorithm.

Below is a simple Matlab/Octave code to find $k = \gcd(m, n)$ based on the strict definition of the EA as described by Stillwell (2010):

```matlab
function k = gcd(m,n)
    while m ~= 0
        A=m; B=n;
        m=max(A,B); n=min(A,B); %m>n
        m=m-n;
    end
    k = m;
end
```

This program looping until $m = 0$. An much more efficient method is described in Section 2.1.3, p. 186, using the `floor()` function, which is called **division with rounding**.
1.3. STREAM 1: NUMBER SYSTEMS (10 LECTURES)

1.3.5 Lec 6: Continued fraction algorithm

In its simplest form, the Continued fraction algorithm (CFA) starts from a single real decimal number $x \in \mathbb{R}$, and recursively expands it as a fraction $x \in \mathbb{F}$. Thus the CFA is used in finding rational approximations to any real number. For example, $\pi \approx 22/7$, which was well known by the Chinese mathematicians. The GCD (the Euclidean algorithm) on the other hand operates on a pair of integers $m, n \in \mathbb{N}$ and finds their greatest common divisor $k \in \mathbb{N}$. Thus $m/k, n/k \in \mathbb{N}$, reducing the ratio to its irreducible form, since $m/k \perp n/k$.

Despite this seeming large difference in the two algorithms, apparently the CFA is closely related to the Euclidean algorithm (the GCD), so closely in fact, that Gauss referred to the Euclidean algorithm as the Continued fraction algorithm (Stillwell, 2010, P. 48). At first glance it is not clear why Gauss would call the CFA the Euclidean algorithm. One must assume that Gauss had some deeper insight into this relationship. If so, that insight would be valuable to understand. 36

In the following we refine the description of the CFA and give examples that go beyond the simple cases of expanding numbers. The CFA of any positive number, say $x_o \in \mathbb{R}^+$, is defined as follows:

1. Start with $n = 0$ and a positive input target $x_o \in \mathbb{R}^+$.

2. Define $a_n = \text{round}(x_n)$, which rounds to the nearest integer.

3. Define $r_n = x_n - a_n$, thus $-0.5 \leq r_n \leq 0.5$. If $r_n = 0$, the recursion terminates.

4. Define $x_{n+1} = 1/r_n$ and return to step 2, with $n = n + 1$.

The recursion may continue to any desired accuracy, since convergence is guaranteed.

An example: Let $x_o \equiv \pi \approx 3.14159 \ldots$. Thus $a_o = 3, r_o = 0.14159, x_1 = 7.065 \approx 1/r_o$, and $a_1 = 7$. If we were to stop here we would have

$$\hat{\pi}_2 \approx 3 + \frac{1}{7 + 0.0625 \ldots} \approx 3 + \frac{1}{7} = \frac{22}{7}. \quad (1.7)$$

This approximation of $\pi \approx 22/7$ has a relative error of 0.04%

$$\frac{22/7 - \pi}{\pi} = 4 \times 10^{-4}.$$  

For the next approximation we continue by reciprocating the remainder $1/0.0625 \approx 15.9966$ which rounds to 16 giving a negative remainder of $\approx -1/300$, resulting in the second approximation

$$\hat{\pi}_3 \approx 3 + 1/(7 + 1/16) = 3 + 16/(7 \cdot 16 + 1) = 3 + 16/113 = 355/113.$$  

Note that if we had truncated 15.9966 to 15, the remainder would have been much larger, but always positive, resulting in a much less accurate rational approximation for the same number of terms. It follows that there can be a dramatic difference depending on the rounding scheme, which, for clarity, is best specified rather than inferred.

36The resolution of this interrelationship is still unresolved.
Rational approximation examples

\[ \hat{\pi}_2 = \frac{22}{7} = [3., 7] \approx \hat{\pi}_2 + O(1.3 \times 10^{-3}) \]

\[ \hat{\pi}_3 = \frac{355}{113} = [3., 7, 16] \approx \hat{\pi}_3 + O(2.7 \times 10^{-7}) \]

\[ \hat{\pi}_4 = \frac{104348}{33215} = [3., 7, 16, -249] \approx \hat{\pi}_4 + O(3.3 \times 10^{-10}) \]

Figure 1.8: The expansion of \( \pi \) to various orders, using the CFA, along with the order of the error of each rational approximation, with rounding. For example \( \hat{\pi}_2 = 22/7 \) has an absolute error \( |22/7 - \pi| \) of about 0.13%.

**Notation:** Writing out all the fractions can become tedious. For example, expanding \( e = 2.7183 \cdots \) using the Matlab/Octave command `rat(exp(1))` gives the approximation

\[
\exp(1) = 3 + 1/((-4 + 1/(2 + 1/(5 + 1/(-2 + 1/(-7))))) - O \left(1.75 \times 10^{-6}\right),
\]

\[ = [3. - 4, 2, 5, -2, -7] - O(1.75 \times 10^{-6}) .\]

Since many entries are negative, we may deduce that rounding arithmetic is being used by Matlab (but not documented). Note that the leading integer part may be noted by an optional decimal point or semicolon.\(^{37}\) If the process is carried further, the values of \( a_n \in \mathbb{N} \) give increasingly more accurate rational approximations.

**Rounding schemes:** In Matlab/Octave there are five different rounding schemes (i.e., mappings): `round(c)`, `fix(c)`, `floor(c)`, `ceil(c)`, `roundb(c)` with \( c \in \mathbb{C} \). If the rounding-down (`floor()`) is used \( \hat{\pi}_{12} = [3., 7, 15, 1, 292, 1, 1, 1, 2, 1, 3, 1] \), whereas true rounding to the nearest integer (`round()`) gives \( \hat{\pi}_8 = [3., 7, 16, -294, 3, -4, 5, -15] \). Thus `round()` introduces negative coefficients when a number rounds up to the nearest integer.

**Exercise:** Based on several examples, which rounding scheme is the most accurate? Explain why. **Sol:** Rounding will give a smaller remainder at each iteration, resulting in a smaller net error and thus faster convergence.

When the CFA is applied and the expansion terminates \( (r_n = 0) \), the target is rational. When the expansion does not terminate (which is not always easy to determine, as the remainder may be ill-conditioned due to small numerical rounding errors), the number is irrational. Thus the CFA has important theoretical applications regarding irrational numbers. You may explore this using Matlab’s `rats(pi)` command.

One of the useful things about the procedure, besides its being so simple, are its generalizations, one of which will be discussed in Section 2.1.3 (p. 186).

Besides these five basic rounding schemes, there are two other important \( \mathbb{R} \to \mathbb{N} \) functions (i.e., mappings): `mod(x, y)`, `rem(x, y)` with \( x, y \in \mathbb{R} \), which will be needed later. The base 10 numbers may be generated from the counting numbers using \( y = \text{mod}(x, 10) \).

\(^{37}\)Unfortunately Matlab/Octave does not support the bracket notation.
Exercise:

1. Show how to generate a base-10 real number \( y \in \mathbb{R} \) from the counting numbers \( \mathbb{N} \) using the 
   \[ m = \text{mod} ( n, 10 ) + k10 \] 
   with \( n, k \in \mathbb{N} \). **Sol:** Every time \( n \) reaches 10, \( m = 0 \). Keep track of this by adding 1 to \( k \).

2. How would you generate binary numbers (base 2) using the \( \text{mod} ( x, b ) \) function? **Sol:** Use the same method as in the first example above, but with \( b = 2 \).

3. How would you generate hexadecimal numbers (base 16) using the \( \text{mod} ( x, b ) \) function? **Sol:** Use the same method as in the first example above, but with \( b = 16 \).

4. Write out the first 19 numbers in hex notation, starting from zero. **Sol:** 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F, 10, 11, 12, \ldots

5. What is \( FF_{16} \) in decimal notation? **Sol:** Using the Matlab/octave function \( \text{hex2dec} ( 'ff' ) \), is 255.

**Symmetry:** A continued fraction expansion can have a high degree of symmetry. For example, the CFA of

\[
R_1 = \frac{1 + \sqrt{5}}{2} = 1 + \frac{1}{1 + \frac{1}{1 + \frac{1}{1 + \cdots}}} = 1.618033988749895 \ldots \quad (1.8)
\]

Here \( a_n \) in the CFA is always 1 \( (R_1 \equiv [1., 1, 1, \cdots]) \), thus the sequence will not terminate, proving that \( \sqrt{5} \in \mathbb{I} \). A related example is \( R_2 \equiv \text{rat} \left( 1+\sqrt{2} \right) \), which gives \( R_2 = [2., 2, 2, 2, \cdots] \).

When expanding a target irrational number \( (x_o \in \mathbb{I}) \), and the CFA is truncated, the resulting rational fraction approximates the irrational target. For the example above, if we truncate at three coefficients ([1; 1, 1]) we obtain

\[
1 + \frac{1}{1 + \frac{1}{1+0}} = 1 + 1/2 = 3/2 = 1.5 = \frac{1 + \sqrt{5}}{2} + 0.118 \ldots
\]

Truncation after six steps gives

\[
[1., 1, 1, 1, 1, 1, 1] = 13/8 \approx 1.6250 = \frac{1 + \sqrt{5}}{2} + .0070 \ldots
\]

Because all the coefficients are 1, this example converges very slowly. When the coefficients are large (i.e., remainder small), the convergence will be faster. The expansion of \( \pi \) is an example of faster convergence.
In summary: Every rational number $m/n \in \mathbb{F}$, with $m > n > 1$, may be uniquely expanded as a continued fraction, with coefficients $a_k$ determined using the CFA. When the target number is irrational ($x_o \in \mathbb{Q}$), the CFA does not terminate, thus each step produces a more accurate rational approximation, converging in the limit as $n \to \infty$.

Thus the CFA expansion is an algorithm that can, in theory, determine when the target is rational, but with an important caveat: one must determine if the expansion terminates. This may not be obvious. The fraction $1/3 = 0.33333 \cdots$ is an example of such a target, where the CFA terminates yet the fraction repeats. It must be that $1/3 = 3 \times (10^{-1} + 10^{-2} + 10^{-3} + \cdots)$.

Here $3 \times 3 = 9$. As a second example

$1/7 = 0.142857142857142857142857 \cdots = 142857 \times (10^{-6} + 10^{-12} + \cdots)$

There are several notations for repeating decimals such as $1/7 = 0.142857$ and $1/7 = 0.1(142857)$. Note that $142857 = 999999/7$. Related identities include $1/11 = 0.090909 \cdots$ and $11 \times 0.090909 = 999999$. When the sequence of digits repeats, the sequence is predictable, it must be rational. But it is impossible to be sure that it repeats, because the length of the repeat can be very arbitrarily long.

WEEK 3

1.3.6 Lec 7: Pythagorean triplets (Euclid’s formula)

Euclid’s formula is a method for finding three integer lengths $[a, b, c] \in \mathbb{N}$, that satisfy Eq. 1.1. It is important to ask “Which set are the lengths [a,b,c] drawn from?” There is a huge difference, both practical and theoretical, if they are from the real numbers $\mathbb{R}$, or the counting numbers $\mathbb{N}$. Given $p, q \in \mathbb{N}$ with $p > q$, the three lengths $[a, b, c] \in \mathbb{N}$ of Eq. 1.1 are given by

$$a = p^2 - q^2, \quad b = 2pq, \quad c = p^2 + q^2. \quad (1.9)$$

This result may be directly verified, since


or

$$p^4 + q^4 + 2p^2q^2 = p^4 + q^4 - 2p^2q^2 + 4p^2q^2.$$

Thus, Eq. 1.9 is easily proven, once given. Deriving Euclid’s formula is obviously more difficult.

A well known example is the right triangle depicted in Fig. 1.9, defined by the integers $[3, 4, 5] \in \mathbb{N}$, having angles $[0.54, 0.65, \pi/2]$ [rad], which satisfies Eq. 1.1 (p. 18). As quantified by Euclid’s formula (Eq. 1.9), there are an infinite number of Pythagorean triplets (PTs). Furthermore the seemingly simple triangle, having angles of $[30, 60, 90] \in \mathbb{N}$ [deg] (i.e., $[\pi/6, \pi/3, \pi/2] \in \mathbb{I}$ [rad]), has one irrational ($\mathbb{I}$) length ($[1, \sqrt{3}, 2]$).

38Taking the Fourier transform of the target number, represented as a sequence, could help to identify an underlying periodic component. The number $1/7 \leftrightarrow [1, 4, 2, 8, 5, 7]$ has a 50 [dB] notch at 0.8$\pi$ [rad] due to its 6 digit periodicity, carried to 15 digits (Matlab/Octave maximum precision), Hamming windowed, and zero padded to 1024 samples.
1.3. STREAM 1: NUMBER SYSTEMS (10 LECTURES)

Figure 1.9: Beads on a string form perfect right triangles when the numbers of unit lengths between beads for each side satisfies Eq. 1.1. For example, when $p = 2$, $q = 1$, the sides are $[3, 4, 5]$.

<p>| | |</p>
<table>
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<tbody>
<tr>
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<td>119</td>
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<td>3367</td>
<td>4825</td>
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<td>4601</td>
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<tr>
<td>1771</td>
<td>3229</td>
</tr>
<tr>
<td>56</td>
<td>106</td>
</tr>
</tbody>
</table>

Figure 1.10: “Plimpton-322” is a stone tablet from 1800 BCE, displaying $a$ and $c$ values of the Pythagorean triplets $[a, b, c]$, with the property $b = \sqrt{c^2 - a^2} \in \mathbb{N}$. Several of the $c$ values are primes, but not the $a$ values. The stone is item 322 (item 3 from 1922), from the collection of George A. Plimpton. —Stillwell (2010)

The technique for proving Euclid’s formula for PTs $[a, b, c] \in \mathbb{Q}$, derived in Fig. 2.3 (p. 191) of Section 2.2.1, is much more interesting than the PTs themselves.

The set from which the lengths $[a, b, c]$ are drawn was not missed by the Indians, Chinese, Egyptians, Mesopotamians, Greeks, etc. Any equation whose solution is based on integers is called a *Diophantine equation*, named after the Greek mathematician Diophantus of Alexandria (c250 CE).

A stone tablet having the numbers engraved on it, as shown in Table 1.10, was discovered in Mesopotamia, from the 19th century BCE, and cataloged in 1922 by George Plimpton.\(^{39}\)\(^{40}\) These numbers are $a$ and $c$ pairs from PTs $[a, b, c]$. Given this discovery, it is clear that the Pythagoreans were following those, long before them. Recently a second similar stone, dating between 350 and 50 BCE has been reported, that indicates early calculus on the orbit of Jupiter’s moons.\(^{41}\)

40 https://en.wikipedia.org/wiki/Plimpton_322
1.3.7 **Lec 8: Pell’s Equation**

There is a venerable history for Pell’s equation

\[ x_n^2 - Ny_n^2 = (x_n - \sqrt{N}y_n)(x_n + \sqrt{N}y_n) = 1, \]  

(1.10)

with non-square \( N \in \mathbb{N} \) specified and \( x, y \in \mathbb{N} \) unknown. It is suspected that Pell’s equation is directly related to the Euclidean algorithm, as applied to polynomials having integer coefficients (Stillwell, 2010, 48). For example, with \( N = 2 \), one solution is \( x = 17, y = 12 \) \((17^2 - 2 \cdot 12^2 = 1)\).

A 2x2 matrix recursion algorithmic, likely due to the Chinese, was used by the Pythagoreans to investigate the \( \sqrt{N} \)

\[ \begin{bmatrix} x_n \\ y_n \end{bmatrix} = \begin{bmatrix} 1 & N \\ 1 & 1 \end{bmatrix} \begin{bmatrix} x_{n-1} \\ y_{n-1} \end{bmatrix}. \]  

(1.11)

Starting with \( [x_o, y_o]^T = [1, 0]^T \), results in solutions of Pell’s equations (Stillwell, 2010, p. 44). Their approach was likely motivated by the Euclidean algorithm (GCD, p. 42), since \( y_n/x_n \rightarrow \sqrt{2} \) (Stillwell, 2010, p. 37, 55).

Note that this is a composition method, of 2x2 matrices, since the output of one matrix multiply is the input to the next. They key question what is the relationship between Pell’s equation and the linear recursion? Is it that Pell’s equation may be trivially factored? There must be some simple way to prove that Eqs. 1.11 and 1.10 are equivalent, as demonstrated on Section 2.2.3 (p. 193).

**Asian solutions:** The first solution of Pell’s equation was published by Brahmagupta (c628), who independently discovered the equation (Stillwell, 2010, p. 46). Brahmagupta’s novel solution introduced a different composition method (Stillwell, 2010, p. 69), and like the Greek result, these solutions were incomplete.

Then in 1150CE, Bhâskara II obtained solutions using Eq. 1.11 (Stillwell, 2010, p.69). This is the solution method we shall explore here, as summarized in Fig. 1.11.

The best way to see how this recursion results in solutions to Pell’s equation, is by example. Initializing the recursion with the trivial solution \( [x_o, y_o]^T = [1, 0]^T \), gives

\[
\begin{align*}
[x_1] &= [1] = [1 \ 2] [1] \\
[y_1] &= [1] = [1 \ 1] [0] \\
[x_2] &= [3] = [1 \ 2] [1] \\
[y_2] &= [2] = [1 \ 1] [1] \\
[x_3] &= [7] = [1 \ 2] [3] \\
[y_3] &= [5] = [1 \ 1] [2] \\
[x_4] &= [17] = [1 \ 2] [7] \\
[y_4] &= [12] = [1 \ 1] [5] \\
[x_5] &= [41] = [1 \ 2] [17] \\
[y_5] &= [29] = [1 \ 1] [12]
\end{align*}
\]

Thus the recursion results in a modified version of Pell’s equation

\[ x_n^2 - 2y_n^2 = (-1)^n, \]  

(1.12)

where only even values of \( n \) are solutions. This sign change had no effect on the Pythagoreans goal, since they only cared about the ratio \( y_n/x_n \rightarrow \pm \sqrt{2} \).
The key to the analysis of such equations is called the **eigen-analysis**, or **modal-analysis** method. Eigen-modes describe the naturally occurring “modes” found in all physical wave-dominated boundary value problems, and each modes “eigen-value” quantifies the modes frequency. Complex eigen-values result in damped modes, which decay in time, due to energy...
losses. Cauchy’s residue theorem is used to find the time-domain response of each frequency-domain complex eigen-mode. Thus eigen-analysis and eigen-modes of physics are the same thing (see Sect. 1.5.3, p. 112), but are described using an different (i.e., unrecognizable) notional methods.\textsuperscript{42} The eigen-method method is summarized in Appendix D, p. 267.

Taking a simple example of a 2x2 matrix $T \in \mathbb{C}$, we start from the definition of the two eigen-equations
\[
Te_\pm = \lambda_\pm e_\pm,
\] (1.13)
corresponding to two eigen-values $\lambda_\pm \in \mathbb{C}$ and two 2x1 eigen-vectors $e_\pm \in \mathbb{C}$. The eigen-values $\lambda_\pm$ may be merged into a 2x2 diagonal eigen-value matrix
\[
\Lambda = \begin{bmatrix}
\lambda_+ & 0 \\
0 & \lambda_-
\end{bmatrix},
\] while the two eigen-vectors $e_+$ and $e_-$ are merged into a 2x2 eigen-vector matrix
\[
E = [e_+, e_-] = \begin{bmatrix}
e_1^+ & e_2^- \\
e_1^- & e_2^+
\end{bmatrix},
\] (1.14)
corresponding to the two eigen-values. Using matrix notation, this may be compactly written as
\[
TE = E\Lambda.
\] (1.15)

Note that while $\lambda_\pm$ and $E_\pm$ commute, $E\Lambda \neq \Lambda E$. From Eq. 1.15 we may obtain two very important forms,

1. the diagonalization of $T$
\[
\Lambda = E^{-1}TE,
\] (1.16)

and

2. the eigen-expansion of $T$
\[
T = E^{-1}\Lambda E^{-1},
\] (1.17)

which is useful for computing power of $T$ (i.e., $T^{100} = E^{-1}\Lambda^{100}E$).

Example: If we take
\[
T = \begin{bmatrix}
1 & 1 \\
1 & -1
\end{bmatrix}
\] then the eigen-values are given by $(1 - \lambda_\pm)(1 + \lambda_\pm) = -1$, thus $\lambda_\pm = \sqrt{2}$. This method of eigen-analysis is discussed in Section 2.2.2 (p. 190) and Appendix E (p. 271).

The key idea of the 2x2 matrix solution, widely used in modern engineering, can be traced back to Brahmagupta’s solution of Pell’s equation, for arbitrary $N$. Brahmagupta’s recursion, identical to that of the Pythagoreans’ $N = 2$ case (Eq. 1.11), eventually lead to the concept of linear algebra, defined by the simultaneous solutions of many linear equations. The recursion by the Pythagoreans

\textsuperscript{42}During the discovery or creation of quantum mechanics, two methods were developed, Schrödinger’s method and Heisenberg’s method. Schrödinger used differential equations while Heisenberg used a matrix method. Eventually it was recognized there were equivalent methods.
(6th BCE) predated the creation of algebra by al-Khwārizmī (9th CE century) (Fig. 1.1) (Stillwell, 2010, p. 88).

## WEEK 4

### 1.3.8 Lec 9: Fibonacci sequence

Another classic problem, formulated by the Chinese, was the Fibonacci sequence, generated by the relation

$$f_{n+1} = f_n + f_{n-1}. \quad (1.18)$$

Here the next number $f_{n+1} \in \mathbb{N}$ is the sum of the previous two. If we start from $[0, 1]$, this difference equation leads to the Fibonacci sequence $f_n = [0, 1, 1, 2, 3, 5, 8, 13, 21, 34, \ldots]$. Alternatively, if we define $y_{n+1} = x_n$, then Eq. 1.18 may be compactly represented by a 2x2 matrix equation recursion

$$
\begin{bmatrix}
x \\
y
\end{bmatrix}_{n+1} =
\begin{bmatrix}
1 & 1 \\
1 & 0
\end{bmatrix}
\begin{bmatrix}
x \\
y
\end{bmatrix}_n.
$$

(We have moved the index outside the vector to save typing.)

The correspondence of Eqs. 1.18 and 1.19 is easily verified. Starting with $[x, y]^T_0 = [0, 1]^T$ we obtain for the first few steps

$$
\begin{bmatrix}
1 \\
0
\end{bmatrix}_1 =
\begin{bmatrix}
1 & 1 \\
1 & 0
\end{bmatrix}
\begin{bmatrix}
0 \\
1
\end{bmatrix}_0, \\
\begin{bmatrix}
1 \\
2
\end{bmatrix}_2 =
\begin{bmatrix}
1 & 1 \\
1 & 0
\end{bmatrix}
\begin{bmatrix}
0 \\
1
\end{bmatrix}_1, \\
\begin{bmatrix}
2 \\
3
\end{bmatrix}_3 =
\begin{bmatrix}
1 & 1 \\
1 & 0
\end{bmatrix}
\begin{bmatrix}
1 \\
2
\end{bmatrix}_2, \\
\begin{bmatrix}
3 \\
5
\end{bmatrix}_4 =
\begin{bmatrix}
1 & 1 \\
1 & 0
\end{bmatrix}
\begin{bmatrix}
2 \\
3
\end{bmatrix}_3, \ldots
$$

From the above $x_n = [0, 1, 1, 2, 3, 5, \ldots]$ is the Fibonacci sequence since the next $x_n$ is the sum of the previous two and the next $y_n$ is $x_n$.

Figure 1.12: This construction is called the Fibonacci spiral. Note how it is constructed out of squares having areas given by the square of the Fibonacci numbers. In this way, the spiral is smooth and the radius increases as the Fibonacci numbers (e.g., $8 = 3 + 5$, $13 = 5 + 8$, etc.). (Adapted from https://en.wikipedia.org/wiki/Golden_spiral.)

### The mean-Fibonacci sequence:

Suppose that the Fibonacci sequence recursion is replaced by the mean of the last two values, namely let

$$f_{n+1} = \frac{f_n + f_{n-1}}{2}. \quad (1.20)$$

This seems like a small change. But how does the solution differ? Two answer this question it is helpful to look at the corresponding 2x2 matrix.
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Exercise: Find the 2x2 matrix corresponding to Eq. 1.20. Sol: The 2x2 matrix may be found using the companion matrix method (p. 66), giving

\[ \begin{bmatrix} x \\ y \end{bmatrix}_{n+1} = \frac{1}{2} \begin{bmatrix} 1 & 1 \\ 2 & 0 \end{bmatrix} \begin{bmatrix} x \\ y \end{bmatrix}_n. \] (1.21)

The eigen-values of this matrix are \([1, -1/2]\) the roots of the binomial equation \(\lambda^2 - \lambda/2 - 1/2 = 0\). Thus \([x_n, y_n] = [1, 1]^T\) and \([x_n, y_n] = [1, 1]^T\) are each solutions.

Exercise: Starting from \([x_n, y_n]^T = [1, 0]^T\) compute the first 5 values of \([x_n, y_n]^T\). Sol: Here is a Matlab/Octave code for computing \(x_n\)

\begin{verbatim}
x(1:2,1)=[1;0];
A=[1 1;2 0]/2;
for k=1:10;
x(k+1)=A*x(:,k);
end
\end{verbatim}

which gives the rational \((x_n \in \mathbb{Q})\) sequence: 1, 1/2, 3/4, 5/8, 11/2^4, 21/2^5, 43/2^6, 85/2^7, 171/2^8, 341/2^9, 683/2^{10}, \ldots.

Exercise: Show that the solution to Eq. 1.20 is bounded, unlike that of the Fibonacci sequence, which diverges. Explain what is going on. Sol: Because the next value is the mean of the last two, the sequence is bounded. To see this one needs to compute the eigen-values of the matrix of Eq. 1.20.

Exercise: Use the formula for the generalized diagonalization of a matrix to find the general solution of the mean-Fibonacci sequence. Sol: The eigen-values are given by the roots of

\[ 0 = -\lambda(1/2 - \lambda) - 1/2 = (\lambda - 1/4)^2 - 9/16 \]

which are \([1, -1/2]\).

By studying the eigen-values of Eq. 1.21 one finds that the steady state solution approaches 1. Namely \(f_n \rightarrow 1 = (f_{n-1} + f_{n-2})/2\) is the solution, as \(n \rightarrow \infty\). Namely the average of the last two values must approach 1 for large \(n\).

Exercise: Show that the geometric series formula hold for 2x2 matrices. Starting with the 2x2 identity matrix \(I_2\) and \(a \in \mathbb{C}\), with \(|a| < 1\), show that

\[ I_2(I_2 - aI_2)^{-1} = I_2 + aI_2 + a^2I_2^2 + a^3I_2^3 + \cdots. \]

Sol: Since \(a^kI_2^k = a^kI_2\), we may multiply both sides by \(I_2 - aI_2^k\) to obtain

\[ I_2 = I_2 + aI_2 + a^2I_2^2 + a^3I_2^3 + \cdots - aI_2(aI_2 + a^2I_2^2 + a^3I_2^3 + \cdots) \]
\[ = [1 + (a + a^2 + a^3 + \cdots) - (a + a^2 + a^3 + a^4 + \cdots)]I_2 \]
\[ = I_2 \]
Summary: The GCD (Euclidean algorithm), Pell’s equation and the Fibonacci sequence may all be written as compositions of 2x2 matrices. Thus Pell’s equation and the Fibonacci sequence are special cases of 2x2 matrix composition

\[
\begin{bmatrix}
x \\
y
\end{bmatrix}_{n+1} = \begin{bmatrix}
a & b \\
c & d
\end{bmatrix} \begin{bmatrix}
x \\
y
\end{bmatrix}_n.
\]

This is an important and common thread of these early mathematical findings. It will turn out that this 2x2 matrix recursion plays a special role in physics, mathematics and engineering, because such equations are solved using the eigen-analysis method. The first several thousands of years of mathematical trial and error research, set the stage for this breakthrough, but this took a long time to fully appreciate.

1.3.9 Lec 10: Exam I (In class)
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1.4 Algebraic Equations: Stream 2

The era from 1600-1850 (Fig. 1.25, p. 107) produced a stream of fundamental theorems. A few of the individuals who played a notable role in this development, in chronological (birth) order, include Galileo, Mersenne, Newton, d’Alembert, Fermat, Huygens, Descartes and Helmholtz. These individuals were some of the first to develop the basic ideas, in various forms, that were then later reworked into the proofs, that today we recognize as the fundamental theorems of mathematics.

1.4.1 Lec 11 The physics behind nonlinear Algebra (Euclidean geometry)

Following Stillwell’s history of mathematics, Stream 2 is geometry, which led to the merging of Euclid’s geometrical methods and the 9th century development of algebra by al-Khwarizmi (830 CE). This integration of ideas lead Descartes and Fermat to develop analytic geometry. While not entirely a unique and novel idea, it was late in coming, given what was known at that time.

The mathematics up to the time of the Greeks, documented and formalized by Euclid, served students of mathematics for more than two thousand years. Algebra and geometry were, at first, independent lines of thought. When merged, the focus returned to the Pythagorean theorem, generalized as analytic conic sections rather than as geometry in Euclid’s Elements. With the introduction of Algebra, numbers, rather than lines, could be used to represent a geometrical length. Thus the appreciation for geometry grew, given the addition of the rigorous analysis using numbers.

Physics inspires algebraic mathematics: The Chinese used music, art, and navigation to drive mathematics. Unfortunately much of our knowledge has been handed down either as artifacts, such as musical bells and tools, or mathematical relationships documented, but not created by, scholars such as Euclid, Archimedes, Diophantus, and perhaps Brahmagupta. With the invention of algebra by al-Khwārizmī (830 CE) this paradigm did not shift, rather it expanded. During the 16th and 17th century, it had becoming clear that differential equations (DEs), such as the wave equation, can characterize a law of nature, at a single point in space and time. This principle was not obvious. A desire to understand motions of objects and planets precipitated many new discoveries. This period is organized in Fig. 1.15. Galileo investigated gravity and invented the telescope. For example the law of gravity (first formulated by Galileo to explain the dropping to two objects of different masses) must obey conservation of energy. Kepler investigated the motion of the planets. While Kepler was the first to appreciate that the planets were described by ellipses, it seems he under-appreciate the significance of this finding, and continued with his epicycle model of the planets. Newton (c1687) went on to show that there must be a gravitational potential between two masses \( m_1, m_2 \), of the form

\[
\phi_g(r(t)) \propto \frac{m_1 m_2}{r(t)},
\]

where \( r = |x_1 - x_2| \) is the Euclidean distance between the two point masses at locations \( x_1 \) and \( x_2 \). Using algebra and his calculus, Newton formalized the equation of gravity, forces and motion (Newton’s three laws) and showed that Kepler’s discovery of planetary elliptical motion naturally follows from these laws.

With the discovery of Uranus (1781) “Kepler’s theory was ruined.” (Stillwell, 2010, p. 23).

Once Newton proposed the basic laws of gravity, he proceed to calculate, for the first time, the speed of sound. This required some form of the pressure wave equation

\[
\frac{\partial^2}{\partial x^2} \rho(x, t) = \frac{1}{c_0^2} \frac{\partial^2}{\partial t^2} \rho(x, t),
\]

(1.23)
a key equation in mathematical physics. The speed of sound is

\[ c_o = \sqrt{\frac{\gamma P_o}{\rho_o}} = 343, \text{ [m/s]} \]

which is a function of the density \( \rho_o = 1.12 \text{ [kg/m}^3\text{]} \) and the dynamic stiffness \( \eta P_o \) of air.\(^{43}\)

What was needed was the concept of constant-heat, or an adiabatic process. For audio frequencies (0.02-20 [kHz]), the small temperature gradients cannot diffuse the distance of a wavelength in one cycle (Pierce, 1981; Boyer and Merzbach, 2011), “trapping” the heat energy in the wave. There were several other physical enigmas, such as the observation that sound disappears in a vacuum and that a vacuum cannot draw water up a column by more than 34 feet.

If we substitute for the pressure

\[ \varrho(x, t) = e^{2\pi f(t \pm kx)}, \quad (1.24) \]

where \( t \) is time and \( x \) is position, we find that \( k = 2\pi / \lambda = 2\pi f / c_o \), because \( f \lambda = c_o \).

This classic relation \( \lambda f = c \) is deceivingly simple, thus confusing, because \( k = 2\pi / \lambda \) becomes complex (has both real and imaginary parts) in dispersive media (e.g., acoustic waves in tubes when losses are considered (Kirchhoff, 1868); electron waves in silicon crystals (Brillouin, 1953). In these more general cases, \( k(f) = 2\pi f / c \) must be replaced with a complex analytic function \( \kappa(s) \) of \( s \), i.e.,

\[ p(x, t) = P_o(s)e^{st}e^{\pm \kappa(s)x}, \quad (1.25) \]

such that the wave number \( \kappa(s) \) is a complex analytic function of the Laplace frequency \( s = \sigma + \omega j \), used with the Laplace transform. This is because electron “waves” in a dispersive medium (e.g., a semi-conductor, such as silicon), are “filtered” in their magnitude and phase. This filter acts as a transmission line, or a complex “wave-filter,” forcing the wavelength to be a complex function of frequency. This view is elegantly explained by Brillouin (1953, Chap. 1), in an historical context. Acoustics provides a rich source of examples (Morse, 1948; Beranek, 1954; Beranek and Mellow, 2012).

Newton’s Principia was finally published in 1687, and the general solution to Newton’s wave equation [i.e., \( p(x, t) = G(t \pm x/c) \)], where \( G \) is any function, was first published 60 years later by d’Alembert (c1747), which showed that for sounds of a single frequency, the wavelength \( \lambda \) and frequency \( f \) were related by

\[ f \lambda = c. \quad (1.26) \]

Today d’Alembert’s analytic wave solution can be written as Eq. 1.24 with a real wave number \( k = 1 / \lambda \text{ [m}^{-1}\text{]} \). This formulation led to the frequency domain concept of Fourier analysis, based on the linearity (i.e., superposition) property of the wave equation (Postulate P2: Lec. 1.4.17, p. 103).

While Newton’s value for the speed of sound in air \( c_o \) was incorrect by the thermodynamic constant \( \sqrt{\gamma} \), a problem that would take more than two hundred years to resolve, his success was important because it quantified the physics behind the speed of sound, and demonstrated that momentum \( mv \), not mass \( m \), was transported by the wave. His concept was correct, and his formulation using algebra and calculus represented a milestone in science. If the speed of sound in-vacuo is zero. In air, assuming no visco-elastic losses, it is constant (i.e., \( c_o = \sqrt{\eta P_o / P_o} \)). When including losses the wave number becomes a complex function of frequency, leading to Eq. 1.25.

\(^{43}\eta = C_p/C_v = 1.4 \) is the ratio of two thermodynamic constants and \( P_o = 10^5 \text{ [Pa]} \) is the barometric pressure of air.
In periodic structures, again the wave number becomes complex due to diffraction, as commonly observed in optics (e.g., diffraction gratings) and acoustics. Thus Eq. 1.26 only holds for the most simple cases, but in general it must be considered as a complex analytic function of $s$, as $\kappa(s)$ in Eq. 1.25.

The corresponding discovery for the formula for the speed of light was made 174 years after Principia, by Maxwell (c1861). Maxwell’s formulation also required great ingenuity, as it was necessary to hypothesize an experimentally unmeasured term in his equations, to get the mathematics to correctly predict the speed of light.

### Chronological history post 17th century

17th  Newton 1642-1727, Bernoulli, Johann 1667-1748

18th  Bernoulli, Daniel, Cauchy 1789-1857, Euler 1707-83, d’Alembert 1717-83, Gauss 1777-1855

19th  Kirchhoff 1824-87, Helmholtz 1821-1894, Riemann 1826-1866, Maxwell 1831-1879, Rayleigh 1842-1919, Heaviside 1850-1925, Poincare 1854-1912,


### Time-Line

Figure 1.13: Time-line of the three centuries from the mid 17th to 20th CE. This was one of the most productive times of all because people were sharing information and working together. Figure 1.15 (p. 82) (Newton–Gauss) gives a closer look at the 15-18 CE, and Fig. 1.25 (p.107) (Bombelli–Einstein) for the full view from 16-20 CE.

### The first Algebra:

Prior to the invention of algebra, people worked out problems as sentences using an obtuse description of the problem (Stillwell, 2010, p. 93). Algebra solved this problem, resulting in a compact language of mathematics, where numbers are represented as abstract symbols (e.g., $x$ and $\alpha$). The problem to be solved could be formulated in terms of sums of powers of smaller terms, the most common being powers of some independent variable (i.e., time or frequency). Today we call such
an expression a \textit{polynomial} of degree \( n \)

\[
P_N(z) \equiv z^n + a_{n-1}z^{n-1} + \cdots + a_0 = \sum_{k=0}^{n} a_k z^k = \prod_{k=0}^{n} (z - z_k).
\] (1.27)

Here we have let \( a_n = 1 \). The coefficient \( a_n \) cannot be zero, or the polynomial would not be of degree \( n \). A solution is to force \( a_n = 1 \), since this simplifies the expression, and does not change the roots.

The key question is “What values of \( z = z_k \) result in \( P_N(z_k) = 0 \).” In other words, what are the roots \( z_k \) of the polynomial? The answer to this question consumed thousands of years, with intense efforts by many aspiring mathematicians. In the earliest attempts, it was a competition to evaluate mathematical acumen. Results were held as a secret to the death bed. It would be fair to view this effort as an obsession. Today the roots of any polynomial may be found by numerical methods, to very high accuracy. The limits to finding roots is limited by the numerical limits of the representation, namely by IEEE-754 (p. 35). There are also a number of important theorems.

Of particular interest is composing a circle with a line, when the line does not touch the circle, and finding the roots. There was no solution to this problem using geometry. This question is addressed in the assignments.

\subsection*{Finding roots of polynomials}

The problem of factoring polynomials has a history more than a millennium in the making. While the quadratic (degree \( N = 2 \)) was solved by the time of the Babylonians (i.e., the earliest recorded history of mathematics), the cubic solution was finally published by Cardano in 1545. The same year, Cardano’s student solved the quartic (\( N = 4 \)). In 1826 (281 years later) it was proved that the quintic (\( N = 5 \)) could not be factored by analytic methods.

As a concrete example we begin with trivial but important case of the quadratic

\[
P_2(x) = ax^2 + bx + c.
\] (1.28)

First note that if \( a = 0 \), the quadratic reduces to the monomial \( P_1(x) = bx + c \). Thus we have the necessary condition that \( a \neq 0 \). The best way to proceed is to divide \( a \) out and work directly with the normalized quadratic \( \hat{P}_2(x) = \frac{1}{a} P_2(x) \). In this way we do not need to worry about the exception, \( a = 0 \).

The \textit{roots} are those values of \( x \) such that \( \hat{P}_2(x_k) = 0 \). One of the first results (recorded by the Babylonians, c2000 BCE) was the factoring of this equation by \textit{completing the square} (Stillwell, 2010, p. 93). One may isolate \( x \) by rewriting Eq. 1.28 as

\[
\frac{1}{a} P_2(x) = \left(x + \frac{b}{2a}\right)^2 - \left(\frac{b}{2a}\right)^2 + c/a.
\] (1.29)

This is easily verified by expanding the squared term and canceling \( (b/2a)^2 \)

\[
\frac{1}{a} P_2(x) = [x^2 + (b/a)x + (b/2a)^2] - \left(\frac{b}{2a}\right)^2 + c/a.
\]

Setting Eq. 1.29 to zero and solving for the two roots \( x_\pm \), gives the \textit{quadratic formula}

\[
x_\pm = -\frac{b \pm \sqrt{b^2 - 4ac}}{2a}.
\] (1.30)
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The roots of $\hat{P}_2(x)$, with $a = 1$, greatly simplify to

$$x_\pm = -\frac{1}{2} b \pm \sqrt{(b/2)^2 - c}. \tag{1.31}$$

This can simplify even further. The term $b^2 - c > 0$ under the square root is called the discriminant, and in physics and engineering problems, 99.9% of the time it is negative. Finally $b/2 \ll \sqrt{c}$, thus the most natural way (i.e., corresponding to the most common physical cases) of writing the solution is

$$x_\pm = -\frac{1}{2} b \pm j \sqrt{c} - \frac{b}{2} \approx -b/2 \pm j \sqrt{c}. \tag{1.32}$$

This form separates the real and imaginary parts of the solution in a natural way. The term $b/2$ is called the damping, which accounts for losses in a resonant circuit, while the term $\sqrt{c}$, for mechanical, acoustical and electrical networks, is called the resonant frequency, typically written as $\omega_0$. The last last approximation ignores the (typically) minor correction to the resonant frequency, which in engineering practice is typically always ignored. Knowing that there is a correction is highlighted by this formula, making one aware of its exists, which is a final feature of this approach.

**Summary:** The quadratic equation and its solution is ubiquitous in physics and engineering. It seems obvious, that instead of memorizing the meaningless Eq. 1.30, one should learn the physically meaningful solution, Eq. 1.32, obtained via Eq. 1.29, with $a = 1$. Arguably, the factored and normalized form (Eq. 1.29) is easier to remember, as a method (completing the square), rather than as a formula to be memorized.

Additionally, the real ($b/2$) and imaginary $\sqrt{c}$ parts of the roots, have physical significance as the damping and resonant frequency. Equation 1.30 has none.

No insight is gained by memorizing the quadratic formula. To the contrary, an important concept is gained by learning how to complete the square, which is typically easier than identifying $a, b, c$ and blindly substituting them into Eq. 1.30. Thus its worth learning the alternating solution (Eq. 1.32), since is more common in practice, and requires less algebra to interpret the final answer.

**Exercise:** By direct substitution demonstrate that Eq. 1.30 is the solution of Eq. 1.28. Hint: Work with $\hat{P}_2(x)$. **Sol:** Setting $a = 1$ the quadratic formula may be written

$$x_\pm = \frac{-b \pm \frac{b}{2} \sqrt{b^2 - 4c} + (b^2 - 4c)}{2}. \tag{1.33}$$

Substituting this into $\hat{P}_2(x)$ gives

$$\hat{P}_\pm(x_\pm) = (x_\pm^2 + bx_\pm + c)$$

$$= \left( \frac{-b \pm \sqrt{b^2 - 4c}}{2} \right)^2 + b \left( \frac{-b \pm \sqrt{b^2 - 4c}}{2} \right) + c$$

$$= \frac{1}{4} \left( b^2 \pm 2b \sqrt{b^2 - 4c} + (b^2 - 4c) + \frac{1}{4} (-2b^2 \pm 2b \sqrt{b^2 - 4c}) + \phi \right)$$

$$= 0.$$

---

44This is the case for mechanical and electrical circuits having small damping. Physically $b > 0$ is the damping coefficient and $\sqrt{c} > 0$ is the resonant frequency. One may then simplify the form as $x^2 + 2bx + c^2 = (x + b + \sqrt{c})(x + b - \sqrt{c})$. 
Exercise: By direct substitution, show that Eq. 1.32 satisfies Eq. 1.30. 

\[ \hat{P}_2(x) = (x - x_+)(x - x_-) \]
\[ = \left( x + \frac{1}{2} b + \sqrt{c - (b/2)^2} \right) \left( x + \frac{1}{2} b - \sqrt{c - (b/2)^2} \right) \]
\[ = \left( x + \frac{b}{2} \right)^2 + (c - b^2/4) \]
\[ = x^2 + bx + \frac{b^2}{4} + \left( c - \frac{b^2}{4} \right) \]
\[ = x^2 + bx + c. \]

In third grade I learned the times-table trick for 9:
\[ n \cdot 9 = (n - 1) \cdot 10 + (10 - n). \] (1.33)

With this simple rule I did not need to depend on my memory for the 9 times tables. By expanding the above, one can see why it works:
\[ 9 \cdot 7 = (7 - 1) \cdot 10 + (10 - 7) = 60 + 3 \] and \[ 9 \cdot 3 = (3 - 1) \cdot 10 + (9 - 3) = 20 + 7. \] As a check, note that the two terms \((n - 1)\) and \((10 - n)\), add to 9.
With each step we are looking for a sequence \( s_n \) that converges to the root, so that \( P_N(s_n) = 0 \) as \( n \to \infty \). Thus we replace \( s \) by \( s_n \) and \( s_0 \) with \( s_{n-1} \), then the formula becomes a recursion for the root \( s_n \) as \( n \to \infty \) \(^{46}\)

\[
(s_n - s_{n+1})P_N'(s_{n+1}) + P_N(s_{n+1}) = P_N(s_n) \to 0.
\]

With every step the expansion point moves closer to the root, converging to the root in the limit. As it comes closer, the linearity assumption becomes more accurate, ultimately resulting in the convergence to the root. Solving for \( s_n \) gives the key formula behind Newton’s famous root-finding method

\[
s_n = s_{n-1} - \frac{P_N(s_{n-1})}{P_N'(s_{n-1})}.
\]

Here \( s_{n-1} \) is the old expansion point and \( s_n \) is the next approximation to the root.

This expression is related to the log-derivative since \( \frac{d \log P(x)}{dx} = \frac{P'(x)}{P(x)} \). It follows that Newton’s method should converge even for cases where fractional derivatives of roots are involved, since the log function will linearize them.

![Figure 1.14: Newton's method applied to the polynomial having roots [1, 2, 3, 4]. A random starting point was chosen, and each curve shows the values of \( s_n \) as Newton’s method converges to the root. Different random starting points converge to different roots. The method always results in convergence to a root. Claims to the contrary seem overblown. For convergence, one must work with \( s_n \in \mathbb{C} \).](https://en.wikipedia.org/wiki/Newton%27s_method)

Newton believed that imaginary roots and numbers had no meaning (p. 108) and only sought out real roots. In this case Newton’s relation may be explored as a graph, which puts Newton’s method in the realm of analytic geometry. The function \( P_N'(x) \) is the slope of the polynomial \( P_N(x) \) at \( x_n \). The value of \( x_n \in \mathbb{R} \) is the estimate of the root after \( n \) iterations, with \( x_0 \) is the initial guess.

Example: When the polynomial is \( P_2 = 1 - x^2 \), so \( P_2'(x) = -2x \) Newton’s iteration becomes

\[
x_{n+1} = x_n + \frac{1 - x_n^2}{2x_n}.
\]

To start the iteration (\( n = 0 \)) we need an initial guess for \( x_0 \), which is a “best guess” of where the root will be. If we let \( x_0 = 1/2 \), then

\[
x_1 = x_0 - \frac{1 - x_0^2}{2x_0} = x_0 + \frac{1}{2} (x_0 - 1/x_0).
\]

\(^{46}\)https://en.wikipedia.org/wiki/Newton’s_method
1. Let $P_2(x) = 1 - x^2$, and $x_0 = 1/2$. Draw a graph describing the first step of the iteration. 

**Sol:** Start with an $(x, y)$ coordinate system and put points at $x_0 = (1/2, 0)$ and the vertex of $P_2(x)$, i.e., $(0, 1)$ ($P_2(0) = 1$). Then draw $1 - x^2$, along with a line from $x_0$ to $x_1$.

2. Calculate $x_1$ and $x_2$. What number is the algorithm approaching? Is it a root of $P_2$?

**Sol:** First we must find $P_2'(x) = -2x$. Thus the equation we will iterate is

$$x_{n+1} = x_n + \frac{1 - x_n^2}{2x_n} = \frac{x_n^2 + 1}{2x_n}.$$ 

By hand

$$x_0 = 1/2$$
$$x_1 = \frac{(1/2)^2 + 1}{2(1/2)} = \frac{1}{4} + 1 = 5/4 = 1.25$$
$$x_2 = \frac{(5/4)^2 + 1}{2(5/4)} = \frac{(25/16) + 1}{10/4} = \frac{41}{40} = 1.025$$

These answers are approaching $x = 1$, the positive (real) root of $P_2(x)$.

3. Write a Matlab script to check your answer for part (a). **Sol:**

```matlab
x=1/2;
for n = 1:3
    x = x+(1-x*x)/(2*x);
end
```

(a) For $n = 4$, what is the absolute difference between the root and the estimate, $|x_r - x_4|$? **Sol:** $4.6E-8$ (very small!)

(b) What happens if $x_0 = -1/2$? **Sol:** You converge on the negative root, $x = -1$.

4. Does Newton’s method work for $P_2(x) = 1 + x^2$? Why? Hint: What are the roots in this case? **Sol:** In this case $P_2'(x) = +2x$ thus the iteration gives

$$x_{n+1} = x_n - \frac{1 + x_n^2}{2x_n}.$$ 

In this case the roots are $x_{\pm} = \pm 1j$, namely purely imaginary. Obviously *Newton’s method fails*, because there is no way for the answer to become complex. Real in, real out.

---

47 https://en.wikipedia.org/wiki/Newton%27s_method#Complex_functions
5. What if you let \( x_0 = (1 + j)/2 \) for the case of \( P_2(x) = 1 + x^2 \)? **Sol:** By starting with a complex initial value, we fix the Real in = Real out problem.

**Fractal diagrams:** When the initial guess is real, but the nearest root is complex, Newton’s iteration fails, as discussed in the exercises above. This failure to converge gives rise to fractal patterns, called the *Mandelbrot set.*

**Example:** Assume that polynomial \( P_3(s) = (s - a)^2(s - b)\pi \). Then

\[
\ln P_3(s) = 2 \ln s - a + \pi \ln s - b
\]

and

\[
\frac{d}{ds} \ln P_3(s) = \frac{2}{s - a} + \frac{\pi}{s - b}.
\]

**Reduction by logarithmic derivative to simple poles:** As shown by the above trivial example, any polynomial, having zeros of arbitrary degree (i.e., \( \pi \) in the example), may be reduced to the ratio of two polynomials, by taking the logarithmic derivative, since

\[
Y_N(s) = \frac{N(s)}{D(s)} = \frac{d}{ds} \ln P_N(s) = \frac{P'_N(s)}{P_N(s)}.
\] (1.34)

Here the starting polynomial is the denominator \( D(s) = P_n(s) \) while the numerator \( N(s) = P'_N(s) \) is the derivative of \( D(s) \). Thus the logarithmic derivative can play a key role in analysis of complex-analytic functions, as it reduces higher order poles, even those of irrational degree, to simple poles.

The logarithmic derivative \( Y_N(s) \) has a number of special properties:

1. \( Y_N(s) \) has simple poles \( s_p \) and zeros \( s_z \).
2. The poles of \( Y_N(s) \) are the zeros of \( P_N(s) \).
3. The zeros of \( Y_N(s) \) (i.e., \( P'_N(s_z) = 0 \)) are the zeros of \( P'_N(s) \).
4. \( Y_N(s) \) is analytic everywhere other than its poles.
5. Since the zeros of \( P_N(s) \) are simple (no second order poles), it is obvious that the zeros of \( Y_N(s) \) always lie close to the line connecting the two poles. One may easily demonstrate the truth of the statement numerically, and has been quantified by the *Gauss-Lucas theorem* which specifies the relationship between the roots of a polynomial and those of its derivative.

6. Newton’s method may be expressed in terms of the logarithmic derivative, since

\[
s_{k+1} = s_k + \epsilon_o Y_N(s),
\]

where we call \( \epsilon_o \) the *step size*, which may be used to control the rate of convergence of the algorithm to the zeros of \( P_n(s) \). If the step size is too large, the root finding can jump to a different domain of convergence, thus a different root of \( P_n(s) \).

---

1.4.2 Matrix formulation of the polynomial

There is a one-to-one relationship between polynomials and matrix analysis. These are best describe in terms of two methods from mathematical physics, the Vandermonde determinant and the companion matrix.

1.4.3 Vandermonde determinant

Consider the polynomial of degree $n$, with $a_n = 1$ (a.k.a., monic\(^49\))

$$P_n(x) = a_0 + a_1 x + a_2 x^2 + \cdots + x^n,$$

(1.35)

which we wish to evaluate at $x = \{x_k\} \in \mathbb{C}$, with $k = 1, \cdots, n$. This leads to an $n \times n$ system of linear equations in $a_k, k = 1, \cdots, n$,

$$\begin{bmatrix} P_n(x_1) \\ P_n(x_2) \\ P_n(x_3) \\ \vdots \\ P_n(x_n) \end{bmatrix} = \mathbf{A} \mathbf{a} = \begin{bmatrix} 1 & x_1 & x_1^2 & x_1^3 & \cdots & x_1^n \\ 1 & x_2 & x_2^2 & x_2^3 & \cdots & x_2^n \\ 1 & x_3 & x_3^2 & x_3^3 & \cdots & x_3^n \\ \vdots & \vdots & \vdots & \vdots & \ddots & \vdots \\ 1 & x_n & x_n^2 & x_n^3 & \cdots & x_n^n \end{bmatrix} \begin{bmatrix} a_0 \\ a_1 \\ a_2 \\ \vdots \\ a_{n-1} \\ 1 \end{bmatrix}.$$

The existence of the solution $\mathbf{a}^* = \mathbf{A}^{-1} \mathbf{y}$ depends on the Vandermonde determinant $|\mathbf{A}| \neq 0$.

We wish to show that

$$P_n(x) = \begin{vmatrix} 1 & x_1 & x_1^2 & \cdots & x_1^n \\ 1 & x_2 & x_2^2 & \cdots & x_2^n \\ 1 & x_3 & x_3^2 & \cdots & x_3^n \\ \vdots & \vdots & \vdots & \ddots & \vdots \\ 1 & x_n & x_n^2 & \cdots & x_n^n \end{vmatrix} = \prod_{i<j=2}^n (x_j - x_i) = (x_2 - x_1) \cdots (x_3 - x_1)(x_3 - x_2) \cdots (x_n - x_n),$$

namely that the set $\{x_k\}$ are the roots of Eq. 1.35, namely that $P_n(x_k) = 0$. In otherwords, when $x \in \{x_k\}$ are the roots, the matrix is singular since then equation becomes $\mathbf{A}(x) = 0$

To do:

1. Show that if the determinant of the Vandermonde matrix is zero, $x_k$ are the roots of $P_{n-1}(x)$.

**Sol:** To make sense of the Vandermonde determinant, replace $x_n$ with $t$

$$\Delta_n = \begin{vmatrix} 1 & x_1 & x_1^2 & x_1^3 & \cdots & x_1^n \\ 1 & x_2 & x_2^2 & x_2^3 & \cdots & x_2^n \\ 1 & x_3 & x_3^2 & x_3^3 & \cdots & x_3^n \\ \vdots & \vdots & \vdots & \vdots & \ddots & \vdots \\ 1 & t & t^2 & t^3 & \cdots & t^n \end{vmatrix},$$

and expand in cofactors along the bottom row. The Vandermonde polynomial (Eq. 1.35) follows. Furthermore it trivially follows that coefficients $a_2, a_1, a_0$ only depend on $x_k$ (they are independent of $t$).

\(^{49}\)https://en.wikipedia.org/wiki/Monic_polynomial
Here is the magic step: If any \( x_k = t \), that row will be identical to the bottom row. If any two rows are the same, the determinant must be zero. It follows that the set \( \{ x_j \} \) are the roots of \( P_n(x) \), namely
\[
P_n(x) = (x - x_1)(x - x_2) \cdots (x - x_{n-1}).
\]

2. Give examples for degrees \( n = 1 \) and \( n = 2 \). **Sol:**

For \( n = 1 \)
\[
P_1(x) = \begin{vmatrix}
1 & x \\
1 & t
\end{vmatrix} = x - t
\]
Thus \( x \) is a root of \( P_1(x) \).

For \( n = 2 \)
\[
\begin{bmatrix}
P_2(x_1) \\
P_2(x_2)
\end{bmatrix} = \begin{vmatrix}
1 & x_1 & x_1^2 \\
1 & x_2 & x_2^2 \\
1 & t & t^2
\end{vmatrix}
\]
Expanding in cofactors
\[
P_2(t) = (x_2 - x_1)t^2 - (x_2^2 - x_1^2)t + (x_1x_2^2 - x_1^2x_2)
\]
\[
= (x_2 - x_1)t^2 - (x_2 - x_1)(x_1 + x_2)t + x_1x_2(x_2 - x_1)
\]
\[
= (x_2 - x_1)[t^2 - (x_1 + x_2)t + x_1x_2]
\]
If we set \( t = x_1 \) we find
\[
P_2(x_1) = (x_2 - x_1)[x_1^2 - (x_1 + x_2)x_1 + x_1x_2]
\]
\[
= (x_2 - x_1)[(x_1^2 - x_2^2) - x_2x_1 + x_1x_2]
\]
\[
= 0.
\]

1.4.4 **Companion Matrix**

The companion matrix
\[
C_N = \begin{bmatrix}
-c_{N-1} & c_{N-2} & \cdots & \cdots & -c_0 \\
1 & 0 & \cdots & \cdots & 0 \\
0 & 1 & \cdots & \cdots & \vdots \\
\vdots & \vdots & \ddots & \ddots & \vdots \\
0 & 0 & \cdots & 1 & 0
\end{bmatrix}
\]
(1.36)
is derived from the monic polynomial
\[
P_N(s) = s^N + c_{N-1}s^{N-1} \cdots + c_2s^2 + c_1s + c_0
\]
\[
= s^N + \sum_{n=0}^{N-1} c_n s^n.
\]
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having coef vector having coefficients

\[ c_{N-1}^T = [c_{N-1}, c_{N-2}, \cdots, c_0]^T. \]

An alternate form is (Horn and Johnson, 1988, p. 146)

\[
C_N' = \begin{bmatrix}
0 & -c_0 \\
1 & 0 & 0 & -c_1 \\
0 & 1 & 0 & -c_2 \\
\vdots & 0 & 1 & \ddots & \vdots \\
\vdots & \ddots & \ddots & \ddots & \vdots \\
0 & \cdots & 0 & 1 & \cdots & 0 & -c_{N-2} \\
0 & \cdots & 0 & 1 & -c_{N-1}
\end{bmatrix}.
\]

(1.37)

The Companion matrix has the same eigen-values as the roots of the monic polynomial \( P_N(s) \). That is, the roots of monic polynomial of degree \( N \) are the eigen-values of the companion matrix \( C_N \).

**Exercise:** Show that the eigen-values of the 3x3 companion matrix

\[
C_3 = \begin{bmatrix}
-\lambda & 0 & -c_0 \\
1 & -\lambda & -c_1 \\
0 & 1 & -(c_2 + \lambda)
\end{bmatrix}
\]

are the same as the roots of \( P_3(s) \). **Sol:** Expanding the determinant of \( C - \lambda I_3 \) along the right-most column gives the characteristic polynomial, which is equal to \( P_3(s) \).

**Exercise:** Find the companion matrix for the Fibonacci sequence, defined by the difference equation

\[ f_{n+2} = f_{n+1} + f_n \]

with \( f_0 = 1 \). **Sol:** Taking the Z transform gives the polynomial \( z^2 - z^1 - z^0 = 0 \), having the coefficient vector \( C = [1, -1, -1] \), resulting in the Fibonacci companion matrix

\[
C = \begin{bmatrix} 1 & 1 \\ 1 & 0 \end{bmatrix},
\]

first discussed on page 53.

Example: Matlab/Octave: A polynomial is represented in Matlab/Octave in terms of its coefficient vector. When the polynomial vector for the poles of a differential equation is

\[
c_N = [1, c_{N-1}, c_{N-2}, \cdots, c_0]^T,
\]

the coefficient \( c_N = 1 \). This normalization guarantees that the lead term is not zero, and the number of roots \( (N) \) is equal to the degree of the monic polynomial.
CHAPTER 1. INTRODUCTION

Working with polynomials in Matlab/Octave:

In Matlab/Octave there are eight functions you need to become familiar with, that work together:

1. \texttt{R=\textbf{r}oot(A)}: Vector \( A = [a_N, a_{N-1}, \ldots, a_0] \in \mathbb{C} \) are the complex coefficients of polynomial \( P_N(z) = \sum_{n=0}^{N} a_n z^n \in \mathbb{C} \), where \( N \in \mathbb{N} \) is the degree of the polynomial. It is convenient to force \( a_N = 1 \), corresponding to dividing the polynomial by this value, when it is not 1, guaranteeing it cannot be zero, as mentioned above. Further \( R \) is the vector of roots, \( [z_1, z_2, \cdots, z_n] \in \mathbb{C} \) such that \( \text{polyval}(A, z_k) = 0 \).

2. \( y=\text{polyval}(A, x) \): This evaluates the polynomial defined by vector \( A \in \mathbb{C}^N \) evaluated at \( x \in \mathbb{C} \), returning vector \( y(x) \in \mathbb{C} \).

3. \( P=\text{poly}(R) \): This is the inverse of \( \text{root}() \), returning a vector of polynomial coefficients \( P \in \mathbb{C}^N \) of the corresponding characteristic polynomial, starting from either a vector of roots \( R \), or a matrix \( A \), for example, defined with the roots on the diagonal. The characteristic polynomial is defined as the determinant of \( |A - \lambda I| = 0 \) having roots \( R \).

Due to IEEE-754 scaling issues, this can give strange results, that are numerically correct, but only within the limits of IEEE-754 accuracy.

4. \( R=\text{polyder}(N) \) This routine takes the coefficients of a polynomial \( N \) and returns the coefficients of the derivative of \( N \). This is useful with working with Newton’s method, since each step is proportional to \( P_N(x) / P'_N(x) \).

5. \( [K, R]=\text{residue}(N, D) \): Given the ratio of two polynomials \( N, D \), \( \text{residue}(N, D) \) returns vectors \( K, R \) such that

\[
\frac{N(s)}{D(s)} = \sum_{k} \frac{K_k}{s - s_k},
\]

where \( s_k \in \mathbb{C} \) are the roots of the denominator \( D \) polynomial and \( K \in \mathbb{C} \) is a vector of residues, which characterize the roots of the numerator polynomial \( N(s) \). The use of \( \text{residue}() \) will be discussed in Sect. 1.4.7 (p. 79), and in greater detail in Sect. 1.6.6 (p. 146), and in Appendix F.1, p. 279.

6. \( C=\text{conv}(A, B) \): Vector \( C \in \mathbb{C}^{N+M-1} \) contain the polynomial coefficients of the convolution of the two vector of coefficients of polynomials \( A, B \in \mathbb{C}^N \) and \( B \in \mathbb{C}^M \). For example \( [1, 2, 1] = \text{conv}([[1, 1], [1, 1]]) \).

7. \( [C, R]=\text{deconv}(N, D) \): Vectors \( C, N, D \in \mathbb{C} \). This operation uses long division of polynomials to find \( C(s) = N(s) / D(s) \) with remainder \( R(s) \), where \( N = \text{conv}(D, C) + R \), namely

\[
C = \frac{N}{D} \quad \text{remainder} \quad R
\]

8. \( A=\text{compan}(D) \): Vector \( D = [1, d_{N-1}, d_{N-2}, \cdots, d_0]^T \) contains the coefficients of polynomial

\[
D(s) = s^N + \sum_{k=1}^{N} d_{N-k} s^k,
\]

and \( A \) is the companion matrix, of vector \( D \) (Eq. F.3, p. 280). The eigen-values of \( A \) are the roots of monic polynomial \( D(s) \).
Exercise: Practice the use of Matlab’s/Octave’s related functions, which manipulate roots, polynomials and residues: `root()`, `conv()`, `deconv()`, `poly()`, `polyval()`, `polyder()`, `residue()`, `companion()`.

Sol: Try Newton’s method for various polynomials. Use $N = \text{poly}(R)$ to provide the coefficients of a polynomial given the roots $R$. Then use `root()` to factor the resulting polynomial. Then use Newton’s method and show that the iteration converges to the nearest root.\(^{50}\)

### 1.4.5 Taylor series

The definition of an analytic function is that it may be expanded in a series (called a Taylor series expansion)

$$P(x) = \sum_{n=0}^{\infty} c_n (x - x_0)^n,$$  \hspace{1cm} (1.40)

that converges for $|x - x_0| < 1$, called the RoC, with coefficients $c_n$ that are determined by the derivatives of $P(x)$, evaluated at the expansion point $x = x_0$. The Taylor series representation of $P(x)$ has special applications for solving differential equations, because 1) it is single valued, and 2) all its derivatives are uniquely defined. A serious limitation of the Taylor series expansion is that it is not valid outside of the RoC. One method for avoiding this limitation is to move the expansion point. But this is easier said than done, because it requires working with the derivatives of $P(x)$ at the new expansion point, where $P(x)$ may not even have derivatives, due to a singularity, for example.

The Taylor series coefficients $c_n$ are defined by taking derivatives of $P(x)$ and evaluating them at the expansion point $x_0$, namely

$$c_n = \frac{1}{n!} \left. \frac{d^n}{dx^n}P(x - x_0) \right|_{x=x_0}. \hspace{1cm} (1.41)$$

The Taylor formula is a prescription for how to uniquely define the coefficients $c_n$. Without the Taylor series formula, we would have no way of determining $c_n$. The proof of the Taylor formula is transparent, simply by taking successive derivative of Eq. 1.40, and then evaluating the result at the expansion point. If $P(x)$ is analytic then this procedure will always work. If $P(x)$ fails to have a derivative of any order, then the function is not analytic and Eq. 1.40 does not valid for $P(x)$. For example if $P(x)$ has a pole at $x_0$ then it is not analytic at that point.

Example: The trivial case to see this is the Taylor series expansion of the geometric series $P(x) = 1/(1 - x)$, about the expansion point $x = 0$. The function $P(x)$ is defined everywhere, except at the singular point $x = 1$, whereas the geometric series is only valid for $|x| < 1$.

Exercise: Verify that $c_0$ and $c_1$ of Eq. 1.40 follow from Eq. 1.41. Sol: To obtain $c_0$, for $n = 0$, there is no derivative ($d^0/dx^0$ indicates no derivative is taken), so we must simply evaluate $P(x - x_0) = c_0 + c_1(x - x_0) + \cdots$ at $x = x_0$, leaving $c_0$. To find $c_1$ we take one derivative which results in $P'(x) = c_1 + 2c_2(x - x_0) + \cdots$. Evaluating this at $x = x_0$ leaves $c_1$. Each time we take a derivative we reduce the degree of the series by 1, leaving the next constant term.

\(^{50}\)A Matlab/Octave program that does this may be downloaded from [http://jontalle.web.engr.illinois.edu/uploads/493/M/NewtonJPD.m](http://jontalle.web.engr.illinois.edu/uploads/493/M/NewtonJPD.m).
The Taylor series plays an important role in mathematics, as the coefficients of the series uniquely determine the analytic series representation via its derivatives. The implications, and limitations of the power series representation are very specific. First, if the series fails to converge (i.e., outside the RoC), it is essentially meaningless. The Taylor series does not need to be infinite to converge to the function it represents, since it obviously works for any polynomial \( P_N(x) \) of degree \( N \). But in the finite case \( N < \infty \), the RoC is infinite, and the series is the function \( P_N(x) \) exactly, everywhere. Of course \( P_N(x) \) is called a polynomial, of degree \( N \). When \( N \to \infty \), the Taylor series is only valid within the RoC, and it is (typically) the representation of the reciprocal of a polynomial.

These properties are both the curse and the blessing of the analytic function. On the positive side, analytic functions are the ideal starting point for solving differential equations, which is exactly how they were used by Newton and others. Analytic functions are “smooth” since they are infinitely differentiable, with coefficients given by Eq. 1.41. They are single valued, so there can be no ambiguity in their interpretation.

Two well known analytic functions are the geometric series

\[
\frac{1}{1 - x} = 1 + x + x^2 + x^3 + \ldots = \sum_{n=0}^{\infty} x^n
\]

and exponential series

\[
e^x = 1 + \frac{1}{2}x^2 + \frac{1}{3 \cdot 2}x^3 + \frac{1}{4 \cdot 3 \cdot 2}x^4 + \ldots = \sum_{n=0}^{\infty} \frac{1}{n!}x^n.
\]

**Brune impedance:** A third very special family of functions are ratios of polynomials, typically used to define Impedances. Impedance is special because it must have a positive real part, so that they obey conservation of energy. A physical impedance cannot have a negative resistance (the real part), or it would act like a source of power, which violates conservation of energy. Most impedances are in the class of Brune impedances, defined by the ratio of two polynomials, of degrees \( M \) and \( N \)

\[
Z_{\text{Brune}}(s) = \frac{P_N(s)}{P_M(s)} = \frac{s^N + a_1 s^{N-1} \ldots a_0}{s^M + b_1 s^{M-1} \ldots b_0},
\]

where \( M = N \pm 1 \) (i.e., \( N = M \pm 1 \)). This fraction of polynomials is sometimes known as a “Pade approximation,” but more specifically this ratio is a Brune impedance, with poles and zeros,” defined as the complex roots of the two polynomials. The key property of the Brune impedance is that the real part of the impedance is non-negative (positive or zero) in the right-hand \( s \) plane

\[
\Re Z(s) = \Re \left[ R(\sigma, \omega) + j X(\sigma, \omega) \right] = R(\sigma, \omega) \geq 0 \quad \text{for} \quad \Re s = \sigma \geq 0.
\]

Since \( s = \sigma + \omega j \), the right-hand plane complex frequency \( (s) \) plane (RHP) corresponds to \( \Re s = \sigma \geq 0 \). This condition is called the class of positive-real functions, and the above condition, and is called the Brune condition, which is frequently written in the abbreviated form as \( \Re Z(\Re s \geq 0) \geq 0 \).

As a result of this positive-real constraint on impedance functions, the subset of Brune impedances (those given by Eq. 1.44 satisfying Eq. 1.45), must be complex analytic in the entire right-hand \( s \) plane. This is a powerful constraint, that places strict limitations on the locations of both the poles and the zeros of every Brune impedance.
Exercise: Find the RoC of the following by application of Eq. 1.41.

1. \( w(x) = \frac{1}{1-x^j} \) \[ \text{Sol: From a straightforward expansion we know the coefficients are} \]

\[
\frac{1}{1-x^j} = 1 + x^j + (x^j)^2 + (x^j)^3 \ldots = 1 + x^j - x^2 - jx^3 \ldots.
\]

Working this out using Eq. 1.41 is more work:

\[
c_0 = \left. \frac{1}{2\pi i} w \right|_0 = 1; \quad c_1 = \left. \frac{1}{\pi} \frac{dw}{dx} \right|_0 = -\frac{j}{(1-x^j)^2} \bigg|_{x=0} = j; \quad c_2 = \left. \frac{1}{2\pi} \frac{d^2w}{dx^2} \right|_0 = \frac{1}{(1-x^j)^3} \bigg|_0 = -1;
\]

\[
c_3 = \left. \frac{1}{3!} \frac{d^3w}{dx^3} \right|_0 = \frac{1}{(1-x^j)^4} \bigg|_0 = -j;
\]

However if we take derivatives of the series expansion it is much easier, and one can even figure out the term for \( c_n \):

\[
c_0 = 1; \quad c_1 = \left. \frac{d}{dx} \sum (jx)^n \right|_0 = j; \quad c_2 = \left. \frac{1}{2!} \frac{d^2}{dx^2} \sum (jx)^n \right|_0 = 2j^2;
\]

\[
c_3 = \left. \frac{1}{3!} \frac{d^3}{dx^3} \sum (jx)^n \right|_0 = (j)^3 = -j;
\]

\[ \cdots, \]

\[
c_n = \left. \frac{1}{n!} j^n n! \right|_0 = j^n.
\]

2. \( w(x) = e^{x^j} \) \[ \text{Sol: } c_n = \frac{1}{n!} j^n. \]

Determining the Region of convergence: Determining the RoC for a given analytic function is quite important, and may not always be obvious. In general the RoC is a circle having a radius, centered on the expansion point, out to the nearest pole. Thus when the expansion point is moved, the RoC changes, since the location of the pole is fixed.

Example: For the geometric series (Eq. 1.42), the expansion point is \( x_0 = 0 \), and the RoC is \( |x| < 1 \), since \( 1/(1-x) \) has a pole at \( x = 1 \). We may move the expansion point by a linear transformation. For example, by replacing \( x \) with \( z+3 \). Then the series becomes \( 1/((z+3)-1) = 1/(z+2) \), so the RoC becomes \( 2 \), because in the \( z \) plane, the pole has moved to \(-2\).

Example: A second important example is the function \( 1/(x^2 + 1) \), which has the same RoC as the geometric series, since it may be expressed in terms of its residue expansion (aka, partial fraction expansion)

\[
\frac{1}{x^2 + 1} = \frac{1}{(x + 1)(x - 1)} = \frac{1}{2j} \left( \frac{1}{x - 1} - \frac{1}{x + 1} \right).
\]

Each term has an RoC of \( |x| < |1/j| = 1 \). The amplitude of each pole is called the residue, defined in Section 1.5.5 Eq. 1.107, p. 121. The residue for the pole at \( 1/j \) is \( 1/2j \).

In summary, the function \( 1/(x^2 + 1) \) is the sum of two geometric series, with poles at \( \pm 1/j \) which are not as obvious because the roots are complex, and conjugate. Once factored, it becomes clear what is going on.

Exercise: Verify the above expression is correct, and show that the residues are \( \pm 1/2j \). \[ \text{Sol: Cross-multiply and cancel, } x \text{ cancels out and we are left with } 1, \text{ as required.} \]
Exercise: Find the residue of $\frac{d}{dz} z^n$. Sol: Taking the derivative gives $z^{n-1}$ which has a pole at $z = 0$. Applying the formula for the residue (Eq. 1.107, p. 121) we find
\[
c^{-1} = \lim_{z \to 0} zz^{n-1} = \lim z^n = 0.
\]
Thus the residue is zero.

The exponential series converges for every finite value of $x \in \mathbb{R}$ (the RoC is the entire real line), thus the exponential is called an *entire function*.

**Analytic functions:**

Any function that has a Taylor series expansion is called an *analytic function*. Within the RoC, the series expansion defines a single valued function. Polynomials, $1/(1 - x)$ and $e^x$ are examples of analytic functions that are real functions of their real argument $x$. This is not the entire story. Because analytic functions are easily manipulated term by term, they may be used to find solutions of differential equations, since the derivatives of a series are uniquely determined within the RoC, due to Eq. 1.41.

Every analytic function has a corresponding differential equation, that is determined by the coefficients $a_k$ of the analytic power series. An example is the exponential, which has the property that it is the eigen-function of the derivative operation
\[
d\frac{d}{dx} e^{ax} = ae^{ax},
\]
which may be verified using Eq. 1.43. This relationship is a common definition of the exponential function, which is a very special, because it is the eigen-function of the derivative.

The complex analytic power series (i.e., complex analytic functions) may also be integrated, term by term, since
\[
\int f(x) dx = \sum_0^\infty \frac{a_k}{k+1} x^{k+1}.
\]
(1.46)

Newton took full advantage of this property of the analytic function and used the analytic series (*Taylor series*) to solve analytic problems, especially for working out integrals, allowing him to solve differential equations. To fully understand the theory of differential equations, one must master single valued analytic functions and their analytic power series.

**Single- vs. multi-valued functions:** Polynomials, and their $\infty$-degree extensions (analytic functions) are single valued: for each $x$ there is a single value of $P_N(x)$. The roles of the domain and codomain may be swapped, to obtain an *inverse function*, which is typically quite different in its properties compared to the function. For example $y(x) = x^2 + 1$ has the inverse $x = \pm \sqrt{y - 1}$, which is double valued, and complex when $y < 1$. Periodic functions, such as $y(x) = \sin(x)$ are even more “exotic,” since $x(y) = \arcsin(x) = \sin^{-1}(x)$ has an $\infty$ number of $x(y)$ values for each $y$. This problem was first addressed in Riemann’s 1851 PhD thesis, while working with Gauss.

Exercise: Let $y(x) = \sin(x)$. Then $dy/dx = \cos(x)$. Show that $dx/dy = -1/\sqrt{1 - x^2}$. Hint: $x(y) = \cos^{-1}(y) = \arccos(y)$. Sol: See the *See implicit function theorem*. (D’Angelo, 2017, p. 104) Add solution.
Exercise: Let $y(x) = \sin(x)$. Then $dy/dx = \cos(xj)$. Show that $dx/dy = -j/\sqrt{1+x^2}$.

Sol: Add solution.

Exercise: Find the Taylor series coefficients of $y = \sin(x)$ and $x = \sin^{-1}(y)$. Sol: Add solution.

Complex analytic functions: When the argument of an analytic function $F(x)$ is complex, that is, $x \in \mathbb{R}$ is replaced by $s = \sigma + \omega j \in \mathbb{C}$ (recall that $\mathbb{R} \subset \mathbb{C}$)

$$F(s) = \sum_{n=0}^{\infty} c_n (s - s_0)^n, \quad (1.47)$$

with $c_n \in \mathbb{C}$, that function is said to be a complex analytic.

For example, when the argument of the exponential becomes complex, it is periodic on the $\omega$ axis, since

$$e^{st} = e^{(\sigma + \omega j)t} = e^{\sigma t} e^{\omega j t} = e^{\sigma t} [\cos(\omega t) + j\sin(\omega t)]. \quad (1.48)$$

Taking the real part gives

$$\Re\{e^{st}\} = e^{\sigma t} \frac{e^{\omega j t} + e^{-\omega j t}}{2} = e^{\sigma t} \cos(\omega t),$$

and $\Im\{e^{st}\} = e^{\sigma t} \sin(\omega t)$. Once the argument is allowed to be complex, it becomes obvious that the exponential and circular functions are fundamentally related. This exposes the family of entire circular functions [i.e., $e^s, \sin(s), \cos(s), \tan(s), \cosh(s), \sinh(s)$] and their inverses [$\ln(s), \arcsin(s), \arccos(s), \arctan(s), \cosh^{-1}(s), \sinh^{-1}(s)$], first fully elucidated by Euler (c1750) (Stillwell, 2010, p. 315). Note that because a function, such as $\sin(\omega t)$, is periodic, its inverse must be multi-valued. What is needed is some systematic way to account for this multi-valued property. That methodology was provided by Riemann 100 years later, in this 1851 PhD Thesis, supervised by Gauss, in the final years of Gauss’ life.

Given a complex analytic function of a complex variable, one must resort to the extended complex plane, Riemann sheets and branch cuts, as discussed in Section 1.4.13 (p. 95). The extended complex plane is a tool that extends the domain of complex analytic to include the point at infinity. This topic is critically important in engineering mathematics, and will be discussed in length in Sections 1.4.13-1.4.16 (pp. 95-101).

Definition of the Taylor series of a complex analytic function: However there is a fundamental problem, since we cannot formally define the Taylor series for the coefficients $c_k$, since we have not defined $dF(s)/ds$, the derivative with respect to the complex variable $s \in \mathbb{C}$. Thus simply substituting $s$ for $x$ in an analytic function is leaving a major hole in our understanding of the complex analytic function.

To gain a feeling of the nature of the problem, we make take derivatives of a function with respect to various variables. For example,

$$\frac{d}{dt} e^{st} = se^{st}.$$
and
\[ e^{\sigma t} \frac{d}{d\omega} e^{i\omega} = \omega ye^{st}. \]
are straightforward.

It was the work of Cauchy (1814) (Fig. 1.13), who uncovered much deeper relationships within complex analytic functions (Sect. 1.4.14, p. 98) by defining differentiation and integration in the complex plane, leading to several fundamental theorems of complex calculus, including the Fundamental theorem of complex integration, and Cauchy’s formula. We shall explore this in and several fundamental theorems in Sect. 1.5.1 (p. 108).

There seems to be some disagreement as to the status of multi-valued functions: Are they functions, or is a function strictly single valued? If so, then we are missing out on a host of interesting possibilities, including all the inverses of nearly every complex analytic function. For example, the inverse of a complex analytic function is a complex analytic function (e.g., \( e^s \) and \( \log(s) \)).

Impact on Physics: It seems likely, if not obvious, that the success of Newton was his ability to describe physics by the use of mathematics. He was inventing new mathematics at the same time as he was explaining new physics. The same might be said for Galileo. It seems likely that Newton was extending the successful techniques and results of Galileo (Galileo, 1638). Galileo died on Jan 8, 1642, and Newton was born Jan 4, 1643, just short of one year later. Certainly Newton was well aware of Galileo’s great success, and naturally would have been influenced by them.

The application of complex analytic functions to physics was dramatic, as may be seen in the six volumes on physics by Arnold Sommerfeld (1868-1951), and from the productivity of his many (36) students (e.g., Debye, Lenz, Ewald, Pauli, Guillemin, Bethe, Heisenberg\(^{51}\) and Seebach, to name a few), notable coworkers (i.e., Leon Brillouin) and others (i.e., John Bardeen), upon whom Sommerfeld had a strong influence. Sommerfeld is known for having many students who were awarded the Nobel Prize in Physics, yet he was not (the prize is not awarded in mathematics). Sommerfeld brought mathematical physics (the merging of physical and experimental principles with mathematics) to a new level with the use of complex integration of analytic functions to solve otherwise difficult problems, thus following the lead of Newton who used real integration of Taylor series to solve differential equations (Brillouin, 1960, Ch. 3 by Sommerfeld, A.).

1.4.6 Lec 12: Polynomial root classification by convolution

Following the exploration of algebraic relationships by Fermat and Descartes, the first theorem was being formulated by d’Alembert. The idea behind this theorem is that every polynomial of degree \( N \) (Eq. 1.27) has at least one root. This may be written as the product of the root and a second polynomial of degree of \( N - 1 \). By the recursive application of this concept, it is clear that every polynomial of degree \( N \) has \( N \) roots. Today this result is known as the fundamental theorem of algebra:

"Every polynomial equation \( P(z) = 0 \) has a solution in the complex numbers. As Descartes observed, a solution \( z = a \) implies that \( P(z) \) has a factor \( z - a \). The
quotient
\[ Q(z) = \frac{P(z)}{z - a} = \frac{P(z)}{a} \left[ 1 + \frac{z}{a} + \left( \frac{z}{a} \right)^2 + \left( \frac{z}{a} \right)^3 + \cdots \right] \] (1.49)

is then a polynomial of one lower degree. \ldots We can go on to factorize \( P(z) \) into \( n \) linear factors.


The ultimate expression of this theorem is given by Eq. 1.27 (p. 59), which indirectly states that an \( n^{th} \) degree polynomial has \( n \) roots. We shall use the term degree when speaking of polynomials and the term order when speaking of differential equations. A general rule is that order applies to the time domain and degree to the frequency domain, since the Laplace transform of a differential equation, having constant coefficients, of order \( N \), is a polynomial of degree \( N \) in Laplace frequency \( s \).

Exercise: Explore expressing Eq. 1.49 in terms of real 2x2 matrices, as described in Section 1.3.1, p. 32.

Today this theorem is so widely accepted we fail to appreciate it. Certainly about the time you learned the quadratic formula, you were prepared to understand the concept of polynomials having roots. The simple quadratic case may be extended to higher degree polynomials. The Matlab/Octave command \texttt{roots([1,a_2,a_1,a_0])} provides the roots \([s_1,s_2,s_3]\) of the cubic equation, defined by the coefficient vector \([1,a_2,a_1,a_0]\). The command \texttt{poly([s_1,s_2,s_3])} returns the coefficient vector. I don’t know the largest degree that can be accurately factored by Matlab/Octave, but I’m sure its well over \( N = 10^3 \). Today, finding the roots numerically is a solved problem.

Factorization versus convolution: The best way to gain insight into the polynomial factorization problem is through the inverse operation, multiplication of monomials. Given the roots \( x_k \), there is a simple algorithm for computing the coefficients \( a_k \) of \( P_N(x) \) for any \( n \), no matter how large. This method is called convolution. Convolution is said to be a trap-door since it is easy, while the inverse, factoring (deconvolution), is hard, and analytically intractable for degree \( N \geq 5 \) (Stillwell, 2010, p. 102).

Convolution of monomials

As outlined by Eq. 1.27, a polynomial has two descriptions, first as a series with coefficients \( a_n \), and second in terms of its roots \( x_r \). The question is “What is the relationship between the coefficients and the roots?” The simple answer is that they are related by convolution.

Let us start with the quadratic
\[(x + a)(x + b) = x^2 + (a + b)x + ab,\] (1.50)

where in vector notation \([-a, -b]\) are the roots and \([1, a + b, ab]\) are the coefficients.

To see how the result generalizes, we may work out the coefficients for the cubic \((N = 3)\). Multiplying the following three factors gives (p. 200)
\[(x - 1)(x - 2)(x - 3) = (x^2 - 3x + 2)(x - 3) = x(x^2 - 3x + 2) - 3(x^2 - 3x + 2) = x^3 - 6x^2 + 11x - 6.\]
When the roots are \([1, 2, 3]\) the coefficients of the polynomial are \([1, -6, 11, -6]\). To verify, substitute the roots into the polynomial, and show that they give zero. For example \(r_1 = 1\) is a root since \(P_3(1) = 1 - 6 + 11 - 6 = 0\).

As the degree increases, the algebra becomes more difficult. Imagine trying to work out the coefficients for \(N = 100\). What is needed is a simple way of finding the coefficients from the roots. Fortunately, convolution keeps track of the book-keeping, by formalizing the procedure.

**Convolution of two vectors:** To get the coefficients by convolution, write the roots as two vectors \([1, a]\) and \([1, b]\). To find the coefficients we must convolve the root vectors, indicated by \([1, a] \star [1, b]\), where \(\star\) denotes convolution. Convolution is a recursive operation. The convolution of \([1, a] \star [1, b]\) is done as follows: reverse one of the two monomials, padding unused elements with zeros. Next slide one monomial against the other, forming the local dot product (element-wise multiply and add):

\[
\begin{align*}
a & 1 0 0 \\
0 & 0 1 b & 0 1 b & 1 b 0 & 1 b 0 0 \\
= 0 & = x^2 & = (a + b)x & = abx^0 & = 0
\end{align*}
\]

resulting in coefficients \([\cdots, 0, 0, 1, a + b, ab, 0, 0, \cdots]\).

By reversing one of the polynomials, and then taking successive dot products, all the terms in the sum of the dot product correspond to the same power of \(x\). This explains why convolution of the coefficients gives the same answer as the product of the polynomials.

As seen by the above example, the position of the first monomial coefficients are reversed, and then slid across the second set of coefficients, the dot product is computed, and the result placed in the output vector. Outside the range shown, all the elements are zero. In summary,

\([1, -1] \star [1, -2] = [1, -1 - 2, 2] = [1, -3, 2]\).

In general

\([a, b] \star [c, d] = [ac, bc + ad, bd]\).

Convolving a third term \([1, -3]\) with \([1, -3, 2]\) gives (p. 200)

\([1, -3] \star [1, -3, 2] = [1, -3 - 3, 9 + 2, -6] = [1, -6, 11, -6]\),

which is identical to the cubic example, found by the algebraic method.

By convolving one monomial factor at a time, the overlap is always two elements, thus it is never necessary to compute more than two multiplies and an add for each output coefficient. This greatly simplifies the operations (i.e., they are easily done in your head). Thus the final result is more likely to be correct. Comparing this to the algebraic method, convolution has the clear advantage.

**Exercise:**

1. Given the coefficient vector \([\alpha, \beta, \gamma] = [1, a + b, ab]\), find the formula for the roots \([a, b]\).

**Sol:** To solve this one must complete the square of the polynomial (p. 59):

\[
x^2 + \beta x + \gamma = 0
\]

\[
(x + \beta/2)^2 = (\beta/2)^2 - \gamma
\]

\[
x + \beta/2 = \pm\sqrt{(\beta/2)^2 - \gamma}
\]

\[
x = -\beta/2 \pm \sqrt{(\beta/2)^2 - \gamma}.
\]
In terms of the coefficient vector (Eq. 1.50) \([1, a + b, ab]\), \(\beta = a + b, \gamma = ab\).

\[
x_\pm = -\frac{a + b}{2} \pm \sqrt{\left(\frac{a + b}{2}\right)^2 - ab}.
\] (1.51)

While it is not obvious that \(x_\pm = [a, b]\), it must be true. One may prove this several ways. The most obvious is working backwards from what we did above. However this gives no insight (try it), other than the fact that we did the algebra correctly. Let’s try something else:

\[
x_\pm + \frac{a + b}{2} = \pm \sqrt{\left(\frac{a + b}{2}\right)^2 - ab} = \sqrt{\frac{a^2 + b^2}{4} + ab - ab} = \frac{1}{2}\sqrt{(a + b)(a - b)}
\]

This does not seem to show the roots satisfy the starting equation.

Instead let’s try the product of the roots

\[
x_+ \cdot x_- = \left( -\frac{a + b}{2} + \sqrt{\left(\frac{a + b}{2}\right)^2 - ab} \right) \left( -\frac{a + b}{2} - \sqrt{\left(\frac{a + b}{2}\right)^2 - ab} \right) = \left(\frac{a + b}{2}\right)^2 - \left(\sqrt{\left(\frac{a + b}{2}\right)^2 - ab}\right)^2 = \left(\frac{a + b}{2}\right)^2 + ab.
\]

We have simultaneous solved two nonlinear equations that relate the roots \([a, b]\), and the coefficients of the polynomial \([1, \beta, \gamma]\):

\[
\beta = (a + b)/2
\]

\[
\gamma = ab
\]

where \(\beta\) mean and \(\gamma\) is the product of the roots.

2. What are the three nonlinear equations that one would need to solve to find the roots of a cubic? **Sol:** From our formula for the convolution of three monomials we may find the nonlinear “deconvolution” relations between the roots\(^{52}\) \([-a, -b, -c]\) and the cubic’s

\(^{52}\) By working with the negative roots we may avoid an unnecessary and messy alternating sign problem.
coefficients $[1, \alpha, \beta, \gamma]$

\[(x + a) \ast (x + b) \ast (x + c) = (x + c) \ast (x^2 + (a + b)x + ab)
\]
\[= x \cdot (x^2 + (a + b)x + ab) + c \cdot (x^2 + (a + b)x + ab)
\]
\[= x^3 + (a + b + c)x^2 + (ab + ac + cb)x + abc
\]
\[= [1, a + b + c, ab + ac + cb, abc].
\]

It follows that the nonlinear equations must be

\[
\begin{align*}
\alpha &= a + b + c \\
\beta &= ab + ac + bc \\
\gamma &= abc.
\end{align*}
\]

Clearly these are solve by the classic cubic solution which appears to be a deconvolution problem, also know as long division of polynomials.

The product of monomial $P_i(x)$ with a polynomial $P_N(x)$ gives $P_{N+1}(x)$: This statement is another way of stating the fundamental theorem of algebra. Each time we convolve a monomial with a polynomial of degree $N$, we obtain a polynomial of degree $N + 1$. The convolution of two monomials results in a quadratic (degree 2 polynomial). The convolution of three monomials give a cubic (degree 3). In general, the degree $k$, of the product of two polynomials, of degree $n, m$, is the sum of the degrees ($k = n + m$). For example, if the degrees are each 5 ($n = m = 5$), then the resulting degree is 10.

In summary, the product of two polynomials of degree $m, n$ having $m$ and $n$ roots, give a polynomial of degree $m+n$ having $m+n$ roots. This is an analysis process, of merging polynomials, by coefficient convolution. Multiplying polynomials is a merging process, into a single polynomial.

Note that the degree of a polynomial is one less than the length of the vector of coefficients. The coefficient on the lead term should always be set to 1 since it cannot be zero, resulting in an illogical result. Always normalize the lead term of $P_N(x)$ to 1 (i.e., $a_N = 1$). This has no effect on the roots.

While you already know this theorem from high school algebra class, it is important to explicitly state the fundamental theorem of algebra.

Composition of polynomials: Convolution is not the only important operation between two polynomials. Another is composition, which may be defined for two functions $f(z), g(z)$. Then the composition $c(z) = f(z) \circ g(z) = f(g(z))$. As a specific example, suppose $f(z) = 1 + z + z^2$ and $g(z) = e^{2z}$. With these definitions

\[f(z) \circ g(z) = 1 + e^{2z} + (e^{2z})^2 = 1 + e^{2z} + e^{4z}.
\]

Note that $f(z) \circ g(z) \neq g(z) \circ f(z)$.

Exercise: Find $g(z) \circ f(z)$. Sol: $e^{2f(z)} = e^{2(1+z+z^2)} = e^2 e^{(1+z+z^2)} = e^3 e^z e^{z^2}$. 

1.4.7 Lec 13 Residue expansions of rational functions (e.g., impedance)

As discussed in Section 1.4.1, p. 68, there are 5 important Matlab/Octave routines that are closely related: conv(), deconv(), poly(), polyder(), polyval(), residue(). Several of these are the complement of the other, or do a similar operation in a slightly different way. Routines conv, poly build polynomials from the roots while root solves for the roots given the polynomial coefficients. The operation residue() convert the ratio of two polynomials and expand it in a partial fraction expansion, with poles and residues.

When lines and planes are defined, the equations are said to be linear in the independent variables. In keeping with this definition of linear, we say that the equations are non-linear when the equations have degree greater than 1 in the independent variables. The term bilinear has a special meaning, in that both the domain and codomain are linearly related by lines (or planes). As an example, impedance is defined in frequency as the ratio of the voltage over the current, but it has a representation as the ratio of two polynomials \( N(s) \) and \( D(s) \)

\[
Z(s) = \frac{N(s)}{D(s)} = sL_0 + R_o + \sum_{k=0}^{K} \frac{K_k}{s - s_k}.
\]

(1.52)

Here \( Z(s) \) is the impedance and \( V \) and \( I \) are the voltage and current at radian frequency \( \omega \).

Such an impedance is typically specified as a rational or bilinear function, namely the ratio of two polynomials, \( P_N(s) = N(s) = [a_N, a_{n-1}, \ldots, a_o] \) and \( P_K(s) = D(s) = [b_K, b_{K-1}, \ldots, b_o] \) of degrees \( N, K \in \mathbb{N} \), as functions of complex Laplace frequency \( s = \sigma + j\omega \), having simple roots. Most impedances are rational functions since they may be written as \( D(s)V = N(s)I \). Since \( D(s) \) and \( N(s) \) are both polynomials in \( s \), rational functions are also called bilinear transformation or in the mathematical literature as Möbius transformation, which comes from a corresponding scalar differential equation, of the form

\[
\sum_{k=0}^{K} b_k \frac{d^k}{dt^k} i(t) = \sum_{n=0}^{N} a_n \frac{d^n}{dt^n} v(t) \leftrightarrow I(\omega) \sum_{k=0}^{K} b_k s^k = V(\omega) \sum_{n=0}^{N} a_n s^n.
\]

(1.53)

This equation, as well as 1.52, follow from the Laplace transform (See Section 1.4.16, p. 101) of the differential equation (on left), by forming the impedance \( Z(s) = V/I = A(s)/B(s) \). This form of the differential equation follows from Kirchhoff’s voltage and current laws (KCL, KVL) or from Newton’s laws (for the case of mechanics).

The physical properties of an impedance: Based on d’Alembert’s observation that the solution to the wave equation is the sum of forward and backward traveling waves, the impedance may be rewritten in terms of forward and backward traveling waves

\[
Z(s) = \frac{V}{I} = \frac{V^+ + V^-}{I^+ - I^-} = r_o \frac{1 + \Gamma(s)}{1 - \Gamma(s)},
\]

(1.54)

where \( r_o = P^+/I^+ \) is called the surge impedance of the transmission line (e.g., wire) connected to the load impedance \( Z(s) \), and \( \Gamma(s) = P^-/P^+ = I^-/I^+ \) is the reflection coefficient corresponding to \( Z(s) \). Like \( Z(s) \), \( \Gamma(s) \) is causal and complex analytic. Note that the impedance and

53Note that the relationship between the impedance and the residues \( K_k \) is a linear one, best solved by setting up a linear system of equations in the unknown residues. Methods for doing this will be discussed in Appendix F.1 (p. 279)
the reflectance function must both be complex analytic, since they are connected by the bilinear transformation, which assures the mutual complex analytic properties.

Due to the bilinear transformation, the physical properties of \( Z(s) \) and \( \Gamma(s) \) are very different. Specifically, the real part of the load impedance must be non-negative (\( \Re\{Z(\omega j)\} \geq 0 \)), if and only if \( |\Gamma(s)| \leq 1 \). In the time domain, the impedance \( z(t) \leftrightarrow Z(s) \) must have a value of \( r_o \) at \( t = 0 \). Correspondingly, the time domain reflectance \( \gamma(t) \leftrightarrow \Gamma(s) \) must be zero at \( t = 0 \).

This is the basis of conservation of energy, which may be traced back to the properties of the reflectance \( \Gamma(s) \).

**Exercise:** Show that if \( \Re\{Z(s)\} \geq 0 \) then \( |\Gamma(s)| \leq 1 \). 

**Sol:** Their two equivalent proofs, both of which start from the relation between \( Z(s) \) and \( \Gamma(s) \). Taking the real part of Eq. 1.54, which must be \( \geq 0 \), we find

\[
\Re\{Z(s)\} = \frac{r_o}{2} \left[ \frac{1 + \Gamma(s)}{1 - \Gamma(s)} + \frac{1 + \Gamma^*(s)}{1 - \Gamma^*(s)} \right] = r_o \frac{1 - |\Gamma(s)|^2}{|1 + \Gamma(s)|^2} \geq 0.
\]

Thus \( |\Gamma| \leq 1 \).

### 1.4.8 Lec 14: Introduction to Analytic Geometry

Analytic geometry is the natural consequence of Euclid’s Geometry (which deals with conical geometry (e.g., points, lines, triangles, circles, spheres, ellipses, cones, etc., in two and three physical dimensions), merged with algebra (which deals with simultaneous equations, roots of polynomials, analytic functions, and ultimately, solutions of differential equations). The combination of Euclid’s (323 BCE) geometry and al-Khwarizmi’s (830 CE) algebra resulted in a totally new powerful tool, analytic geometry, independently worked out by Descartes and Fermat (Stillwell, 2010). The utility and importance of this new tool cannot be understated.

There are many important relationships between Euclidean geometry and 16th century algebra. An attempt at a detailed comparison is summarized in Table 1.2. Important similarities include vectors, their Pythagorean lengths \([a, b, c]\)

\[
c = \sqrt{(x_2 - x_1)^2 + (y_2 - y_1)^2},
\]  

(1.55)

\(a = x_2 - x_1\) and \(b = y_2 - y_1\), and the angles. Euclid’s geometry had no concept of coordinates, thus of vectors. One of the main differences is the ability of analytic geometry is that one may compute with real, and soon after, complex numbers.

There are several new concepts that come with the development of analytic geometry:

1. Composition of functions: If \( y = f(x) \) and \( z = g(y) \) then the composition of functions \( f \) and \( g \) is denoted \( z(x) = g \circ f(x) = g(f(x)) \).

2. Elimination: Given two functions \( f(x, y) \) and \( g(x, y) \), elimination removes either \( x \) or \( y \). This procedure, known to the Chinese, called **Gaussian Elimination**.

3. Intersection: While one may speak of the intersection of two lines to give a point, or two planes to give a line, the term **intersection** is also an important but very different concept in set theory. This is a special case of elimination when the functions \( f(x, y), g(x, y) \) are linear in their arguments.
Table 1.2: An ad-hoc comparison between Euclidean geometry and analytic geometry. I am uncertain as to the classification of the items in the third column.

<table>
<thead>
<tr>
<th>Euclidean geometry: $\mathbb{R}^3$</th>
<th>Analytic geometry: $\mathbb{R}^n$</th>
<th>Uncertain</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Proof</td>
<td>1. Numbers</td>
<td>1. Cross product ($\mathbb{R}^3$)</td>
</tr>
<tr>
<td>2. Line length</td>
<td>2. Algebra</td>
<td>2. Recursion</td>
</tr>
<tr>
<td>3. Line intersection</td>
<td>3. Power series</td>
<td>3. Iteration $\in \mathbb{C}^2$ (e.g., Newton’s method)</td>
</tr>
<tr>
<td>4. Point</td>
<td>4. Analytic functions</td>
<td>4. Iteration $\in \mathbb{R}^n$</td>
</tr>
<tr>
<td>5. Projection (e.g. scalar product)</td>
<td>5. Complex analytic functions: e.g., $\sin \theta, \cos \theta, e^{i\theta}, \log z$</td>
<td></td>
</tr>
<tr>
<td>7. Vector (sort of)</td>
<td>7. Elimination</td>
<td></td>
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<tr>
<td>8. Conic Section</td>
<td>8. Integration</td>
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<td>9. Square Roots (e.g., Spiral of Theodorus)</td>
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<td>11. Polynomial $\in \mathbb{C}$</td>
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<td>13. Normed vector spaces</td>
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<td>14. ...</td>
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4. Vectors: Analytic geometry provides the new concept of a vector, as a line with length and orientation (i.e., direction). Analytic geometry defines vectors in any number of dimensions, as ordered sets of points.

5. Analytic geometry extends the ideas of Euclidean geometry with the introduction of the scalar (dot) product of two vectors $f \cdot g$, and the vector (cross) product $f \times g$ (see Fig. 1.16, p. 82).

What algebra also added to geometry was the ability to compute with complex numbers. For example, the length of a line (Eq. 1.55) was measured in Geometry with a compass: numbers played no role. Once algebra was available, the line’s Euclidean length could be computed numerically, directly from the coordinates of the two ends, defined by the 3-vector

$$e = x\hat{x} + y\hat{y} + z\hat{z} = [x, y, z]^T,$$

which represents a point at $(x, y, z) \in \mathbb{R}^3 \subset \mathbb{C}^3$ in three dimensions, having direction, from the origin $(0, 0, 0)$ to $(x, y, z)$. An alternative matrix notation is $e = [x, y, z]^T$, a column vector of three numbers. These two notations are different ways of representing exactly the same thing. I view them as equivalent notations.

By defining the vector, analytic geometry allows Euclidean geometry to become quantitative, beyond the physical drawing of an object (e.g., a sphere, triangle or line). With analytic geometry we have the Euclidean concept of a vector, a line having a magnitude (length) and direction, but...
Figure 1.15: Exploded time-line of the two centuries from 1665CE to 1855CE during which the full theory of differential equations were developed, in which Lagrange played a large role due to his overlap with Euler and Gauss. Mozart and Beethoven are helpful time-anchors, not part of the math timeline. Given the time line, it seems likely that Bombelli’s discovery of Diophantus’ book on “Arithmetic” in the Vatican library, triggered many of the ideas presented by Galileo, Descartes and Fermat. Bombelli’s discovery might be considered as a magic moment in mathematics. The vertical red lines connect mentor-student relationships.

Figure 1.16: Vectors $\mathbf{A}$, $\mathbf{B}$, $\mathbf{C}$ are used to define the scalar product $\mathbf{A} \cdot \mathbf{B}$, vector (cross) product $\mathbf{A} \times \mathbf{B}$ and triple product $\mathbf{C} \cdot (\mathbf{A} \times \mathbf{B})$. The two vector products complement each other, one proportional to the $\sin \theta$ of the angle $\theta$ between them and the other to the $\cos \theta$. It follows that $||\mathbf{A} \cdot \mathbf{B}||^2 + ||\mathbf{A} \times \mathbf{B}||^2 = ||\mathbf{A}||^2||\mathbf{B}||^2$

Scalar product of two vectors: When using algebra, many concepts, obvious with Euclid’s geometry, may be made precise. There are many examples of how algebra extends Euclidean
1.4. STREAM 2: ALGEBRAIC EQUATIONS (12 LECTURES)

geometry, the most basic being the scalar product (aka dot product) between vectors

\[ \mathbf{x} \cdot \mathbf{\kappa} = (x\hat{x} + y\hat{y} + z\hat{z}) \cdot (\alpha\hat{x} + \beta\hat{y} + \gamma\hat{z}), \quad \in \mathbb{C} \]

\[ = \alpha x + \beta y + \gamma z. \]

In matrix notation the scalar product is written as

\[ \mathbf{x} \cdot \mathbf{\kappa} = \begin{bmatrix} x \\ y \\ z \end{bmatrix}^T \begin{bmatrix} \alpha \\ \beta \\ \gamma \end{bmatrix} = [x, y, z] \begin{bmatrix} \alpha \\ \beta \\ \gamma \end{bmatrix} = \alpha x + \beta y + \gamma z. \quad (1.56) \]

The dot product takes the character of \( \kappa \). For example, if \( \kappa(s) \in \mathbb{C} \) is a function of complex frequency \( s \), then the dot product is complex. If \( \zeta \in \mathbb{R} \) is real, then the dot product is real.

**Norm (length) of a vector:** The norm of a vector

\[ ||\mathbf{e}|| \equiv +\sqrt{\mathbf{e} \cdot \mathbf{e}} \geq 0. \]

is defined as the positive square root of the scalar product of the vector with itself. This is a generalization of the length, in any number of dimensions, forcing the sign of the square-root to be non-negative. The length is a concept of Euclidean geometry, and it must always be positive and real. A complex (or negative) length is not physically meaningful. More generally, the Euclidean length of a line is given as the norm of the difference between two real vectors \( \mathbf{e}_1, \mathbf{e}_2 \in \mathbb{R} \)

\[ ||\mathbf{e}_1 - \mathbf{e}_2||^2 = (\mathbf{e}_1 - \mathbf{e}_2) \cdot (\mathbf{e}_1 - \mathbf{e}_2) \]

\[ = (x_1 - x_2)^2 + (y_1 - y_2)^2 + (z_1 - z_2)^2 \geq 0. \]

From this formula we see that the norm of the difference of two vectors is simply a compact expression for the Euclidean length. A zero-length vector, such as is a point, is the result of

\[ ||\mathbf{x} - \mathbf{x}||^2 = (\mathbf{x} - \mathbf{x}) \cdot (\mathbf{x} - \mathbf{x}), \]

will always be zero.

**Integral definition of a scalar product:** Up to this point, following Euclid, we have only considered a vector to be a set of elements \( \{x_n\} \in \mathbb{R} \), index over \( n \in \mathbb{N} \), as defining a linear vector space with scalar product \( x \cdot y \), with the scalar product defining the norm or length of the vector \( ||x|| = \sqrt{x \cdot x} \). Given the scalar product, the norm naturally follows.

At this point an obvious question presents itself: “Can we extend our definition of vectors to differentiable functions (i.e., \( f(t) \) and \( g(t) \)), indexed over \( t \in \mathbb{R} \), with coefficients labeled by \( t \in \mathbb{R} \), rather than by \( n \in \mathbb{N} \)?” Clearly, if the functions are analytic, there is no obvious reason why this should be a problem, since analytic functions may be represented by a convergent series having Taylor coefficients, thus integrable term by term.

Specifically, under certain conditions, the function \( f(t) \) may be thought of as a vector, defining a normed vector space. This intuitive and somewhat obvious idea is powerful. In this case the scalar product must be defined in terms of the integral

\[ f(t) \cdot g(t) = \int_t f(t)g(t)dt \quad (1.57) \]
summed over \( t \in \mathbb{R} \), rather than a sum over \( n \in \mathbb{N} \).

This definition of the vector scalar product allows for a significant, but straightforward generalization of our vector space, which will turn out to be both useful and an important extension of the concept of a normed vector space. In this space we can define the derivative of a norm with respect to \( t \), which is not possible for the case of the discrete case, indexed over \( n \). The distinction introduces the concept of continuity in the index \( t \), which does not exist for the discrete index \( n \in \mathbb{N} \).

**Pythagorean theorem and the Schwarz inequality:** Regarding Fig. 1.16, suppose we compute the difference between vector \( A \in \mathbb{R} \) and \( \alpha B \in \mathbb{R} \) as

\[
L = \|A - \alpha B\| \in \mathbb{R},
\]

where \( \alpha \in \mathbb{R} \) is a scalar that modifies the length of \( B \). We seek the value of \( \alpha \), which we denote as \( \alpha^* \), that minimizes the length of \( L \). From simple geometrical considerations, \( L(\alpha) \) will be minimum when the difference vector is perpendicular to \( B \), as shown in the figure by the dashed line from the tip of \( A \perp B \).

To show this algebraically we write out the expression for \( L(\alpha) \) and take the derivative with respect to \( \alpha \), and set it to zero, which gives the formula for \( \alpha^* \). The argument does not change, but the algebra greatly simplifies, if we normalize \( A, B \) to be unit vectors \( a = A/\|A\| \) and \( b = B/\|B\| \), which have norm = 1.

\[
L^2 = (a - \alpha b) \cdot (a - \alpha b) = 1 - 2\alpha a \cdot b + \alpha^2. \tag{1.58}
\]

Thus the length is shortest (\( L = L_* \), as shown in Fig. 1.16) when

\[
d\frac{d}{d\alpha} L^2_* = -2a \cdot b + 2\alpha^* = 0.
\]

Solving for \( \alpha^* \) we find \( \alpha^* = a \cdot b \). Since \( L_* > 0 \) (\( a \neq b \)), Eq. 1.58 becomes

\[
1 - 2|a \cdot b|^2 + |a \cdot b|^2 = 1 - |a \cdot b|^2 > 0.
\]

In conclusion \( \cos \theta \equiv |a \cdot b| < 1 \). In terms of \( A, B \) this is \(|A \cdot B| < \|A\| \|B\| \cos \theta \), as shown next to \( B \) in Fig. 1.16. Thus the scalar product between two vectors is their direction cosine. Furthermore since this forms a right triangle, the Pythagorean theorem must hold. The triangle inequality says that the lengths of the two sides must be greater than the hypotenuse. Note that \( \Theta \in \mathbb{R} \notin \mathbb{C} \).

This derivation is an abbreviated version of a related discussion in Section 3.2.1 (p. 87).

**Vector (×) product of two vectors:** As shown in Fig. 1.16, the vector product (aka, cross-product) \( a \times b \) is the second type of product between two vectors. The vector product defines a vector, perpendicular (\( \perp \)) to the plane of the two vectors being multiplied. The formula for computing the cross product is

\[
a \times b = \begin{vmatrix}
\hat{x} & \hat{y} & \hat{z} \\
a_1 & a_2 & a_3 \\
b_1 & b_2 & b_3
\end{vmatrix}.
\]

For example, if the two vectors are in \( \hat{x} \) and \( \hat{y} \), then the cross-product is \( \hat{z} \). It is strictly in \( \hat{z} \) if the two vectors are perpendicular to each other (i.e., \( \hat{z} = \hat{x} \times \hat{y} = -\hat{y} \times \hat{x} \)). The vector product of
a vector with itself (or the difference between two vectors) is zero. For example \( \hat{x} \times \hat{x} = \hat{y} \times \hat{y} = \hat{z} \times \hat{z} = 0 \). Typically \( a, b \in \mathbb{R} \). If they are complex, the definition must be modified to be consistent with the physics.

The scalar product of a third vector \( c \) with the vector product \( a \times b \), with \( a, b, c \in \mathbb{R} \),

\[
    c \cdot (a \times b) = \begin{vmatrix} c_1 & c_2 & c_3 \\ a_1 & a_2 & a_3 \\ b_1 & b_2 & b_3 \end{vmatrix} \quad (\in \mathbb{R}^3),
\]

is called the triple product, which represents the volume of a parallelepiped.

**Impact of Analytic Geometry:** The most obvious impact of analytic geometry, was a detailed analysis of the conic sections, using algebra, rather than drawings via a compass and ruler. An important example is the composition of the line and circle, a venerable construction, presumably going back to before Diophantus (250CE). Once algebra was invented, the composition could be done using formulas. With this analysis came complex numbers.

The first two mathematicians to appreciate this mixture of Euclid’s geometry and the new algebra were Fermat and Descartes (Stillwell, 2010, p. 111-115); soon Newton contributed to this effort, by the addition of physics (calculations in acoustics, orbits of the planets, and the theory of gravity and light Stillwell (2010, p. 115-117)), significant concepts for 1687.

Given these new methods, many new solutions to problems emerged. The complex roots of polynomials continued to appear, without any obvious physical meaning. Complex numbers seem to have been viewed as more of an inconvenience than a problem. Newton’s solution to this dilemma was to simply ignore the imaginary cases (Stillwell, 2010, p. 119).

**Development of Analytic Geometry**

**Intersection and Gaussian elimination:** The first “algebra” (al-jabr) is credited to al-Khwarizmi (830 CE). Its invention advanced the theory of polynomial equations in one variable, Taylor series, and composition versus intersections of curves. The solution of the quadratic equation had been worked out thousands of year earlier, but with algebra a general solution could be defined. The Chinese had found the way to solve several equations in several unknowns, for example, finding the values of the intersection of two circles. With the invention of algebra by al-Khwarizmi, a powerful tool became available to solve the difficult problems.

**Composition and Elimination** In algebra there are two contrasting operations on functions: composition and Elimination.

**Composition:** Composition is the merging of functions, by feeding one into the other. If the two functions are \( f, g \) then their composition is indicated by \( f \circ g \), meaning the function \( y = f(x) \) is substituted into the function \( z = g(y) \), giving \( z = g(f(x)) \).

Composition is not limited to linear equations, even though that is where it is most frequently applied. To compose two functions, one must substitute one equation into the other. That requires solving for that substitution variable, which is not always possible in the case of nonlinear equations. However many tricks are available that may work around this restrictions. For example if one equation is in \( x^2 \) and the other in \( x^3 \) or \( \sqrt{x} \), it may be possible to multiply the first by \( x \) or square the second. The point is that one of the variables must be isolated so that when it is substituted into the other equations, the variable is removed from the mix.
Examples: Let \( y = f(x) = x^2 - 2 \) and \( z = g(y) = y + 1 \). Then
\[
g \circ f = g(f(x)) = (x^2 - 2) + 1 = x^2 - 1.
\] (1.59)

In general composition does not commute (i.e., \( f \circ g \neq g \circ f \)), as is easily demonstrated. Swapping the order of composition for our example gives
\[
f \circ g = f(g(y)) = y^2 - 2 = (y + 1)^2 - 2 = y^2 + 2y - 1.
\] (1.60)

Intersection: Complimentary to composition is intersection (i.e., decomposition) (Stillwell, 2010, pp. 119,149). For example, the intersection of two lines is defined as the point where they meet. This is not to be confused with finding roots. A polynomial of degree \( N \) has \( N \) roots, but the points where two polynomials intersect has nothing to do with the roots of the polynomials. The intersection is a function (equation) of lower degree, implemented with Gaussian elimination.

Intersection of two lines Unless they are parallel, two lines meet at a point. In terms of linear algebra this may be written as 2 linear equations\(^\text{54}\) (on the left), along with the intersection point \([x_1, x_2]^T\), given by the inverse of the 2x2 set of equations (on the right)
\[
\begin{bmatrix} a & b \\ c & d \end{bmatrix} \begin{bmatrix} x_1 \\ x_2 \end{bmatrix} = \begin{bmatrix} y_1 \\ y_2 \end{bmatrix}
\]
\[
\begin{bmatrix} x_1 \\ x_2 \end{bmatrix} = \frac{1}{\Delta} \begin{bmatrix} d & -b \\ -c & a \end{bmatrix} \begin{bmatrix} y_1 \\ y_2 \end{bmatrix}.
\] (1.61)

By substituting the expression for the intersection point \([x_1, x_2]^T\) into the original equation, we see that it satisfies the equations. Thus the equation on the right is the solution to the equation on the left.

Note the structure of the inverse: 1) The diagonal values \((a, d)\) are swapped, 2) the off-diagonal values \((b, c)\) are negated and 3) the 2x2 matrix is divided by the determinant \(\Delta = ad - bc\). If \(\Delta = 0\), there is no solution. When the determinant is zero (\(\Delta = 0\)), the slopes of the two lines
\[
slope = \frac{dx_2}{dx_1} = \frac{b}{a} = \frac{d}{c},
\]
are equal, thus the lines are parallel. Only if the slopes differ can there be a unique solution.

Exercise: Show that the equation on the right is the solution of the equation on the left. Sol: By a direct substitution of the right equation into the left equation
\[
\begin{bmatrix} a & b \\ c & d \end{bmatrix} \cdot \frac{1}{\Delta} \begin{bmatrix} d & -b \\ -c & a \end{bmatrix} \begin{bmatrix} y_1 \\ y_2 \end{bmatrix} = \frac{1}{\Delta} \begin{bmatrix} ad - bc & -ab + ab \\ cd - cd & -cb + ad \end{bmatrix} = \frac{1}{\Delta} \begin{bmatrix} \Delta & 0 \\ 0 & \Delta \end{bmatrix},
\]
which gives the identity matrix.

Algebra will give the solution when geometry cannot. When the two curves fail to intersect on the real plane, the solution still exists, but is complex valued. In such cases, geometry, which only considers the real solutions, fails. For example, when the coefficients \([a, b, c, d]\) are complex,

\(^{54}\)When writing the equation \(Ax = y\) in matrix format, the two equations are \(ax_1 + bx_2 = y_1\) and \(dx_1 + cx_2 = y_2\) with unknowns \((x_1, x_2)\), whereas in the original equations \(ay + bx = c\) and \(dy + cx = f\), they were \(y, x\). Thus in matrix format, the names are changed. The first time you see this scrambling of variables, it can be confusing.
the solution exists, but the determinant can be complex. Thus algebra is much more general than geometric. Geometry fails when the solution has a complex intersection.

A system of linear equations $Ax = y$ has many interpretations, and one should not be biased by the notation. As engineers we are trained to view $x$ as the input and $y$ as the output, in which case then $y = Ax$ seems natural, much like the functional relation $y = f(x)$. But what does the linear relation $x = Ay$ mean, when $x$ is the input? The obvious answer is that $y = A^{-1}x$. But when working with systems of equations, there are many uses of equations, and we need to become more flexible in the our interpretation. For example $y = Ax^2$ has a useful meaning, and in fact we saw this type of relationship when working with Pell’s equation (p. 50) and the Fibonacci sequence (p. 53). As another example consider

$$\begin{bmatrix} z_1 \\ z_2 \end{bmatrix} = \begin{bmatrix} a_{1x} & a_{1y} \\ a_{2x} & a_{2y} \end{bmatrix} \begin{bmatrix} x \\ y \end{bmatrix}$$

which is reminiscent of a three-dimensional surface $z = f(x, y)$. We shall find that such generalizations are much more than a curiosity.

### 1.4.9 Scalar products and its applications

Another important example of algebraic expressions in mathematics is Hilbert’s generalization of Eq. 1.1, known as the Schwarz inequality, shown in Fig. 1.17. What is special about this generalization is that it proves that when the vertex is $90^\circ$, the length of the leg is minimum.

Vectors may be generalize to have $\infty$ dimensions: $\vec{U}, \vec{V} = [v_1, v_2, \ldots, v_\infty]$. The inner product (i.e., dot product) between two such vectors generalizes the finite dimensional case

$$\vec{U} \cdot \vec{V} = \sum_{k=1}^{\infty} u_k v_k.$$ 

As with the finite case, the norm $||\vec{U}|| = \sqrt{\vec{U} \cdot \vec{U}} = \sqrt{\sum u_k^2}$ is the dot product of the vector with itself, defining the length of the infinite component vector. Obviously there is an issue of convergence, if the norm for the vector to have a finite length.

It is a somewhat arbitrary requirement that $a, b, c \in \mathbb{R}$ for the Pythagorean theorem (Eq. 1.1). This seems natural enough since the sides are lengths. But, what if they are taken from the complex numbers, as for the lossy vector wave equation, or the lengths of vectors in $\mathbb{C}^n$? Then the equation generalizes to

$$\mathbf{c} \cdot \mathbf{c} = ||\mathbf{c}||^2 = \sum_{k=1}^{n} |c_k|^2,$$

where $||\mathbf{c}||^2 = (\mathbf{c}, \mathbf{c})$ is the inner (dot) product of a vector $\mathbf{c}$ with itself, where $|c_k|$ is the magnitude the complex $c_k$. As before, $||\mathbf{c}|| = \sqrt{||\mathbf{c}||^2}$ is the norm of vector $\mathbf{c}$, akin to a length.

#### Schwarz inequality

The Schwarz inequality\(^{55}\) says that the magnitude of the inner product of two vectors is less than or equal to the product of their lengths

$$|\mathbf{U} \cdot \mathbf{V}| \leq ||\mathbf{U}|| ||\mathbf{V}||.$$

\(^{55}\)A simplified derivation is provided in Sect. 1.4.8 (p. 80).
CHAPTER 1. INTRODUCTION

This may be simplified by normalizing the vectors to have unit length ($\hat{U} = U/||U||$, $\hat{V} = V/||V||$), in which case $-1 < \hat{U} \cdot \hat{V} \leq 1$. Another simplification is to define the scalar product in terms of the direction cosine

$$\cos \theta = |\hat{U} \cdot \hat{V}| \leq 1.$$  

A proof of the Schwarz inequality is as follows: From these definitions we may define the minimum difference between the two vectors as the perpendicular from the end of one to the intersection of the second. As shown in Fig. 1.17, $U \perp V$ may be found by minimizing the length of the vector difference:

$$\min_{\alpha} ||V - \alpha U||^2 = ||V||^2 + 2\alpha V \cdot U + \alpha^2 ||U||^2 > 0$$

$$0 = \partial_{\alpha} (V - \alpha U) \cdot (V - \alpha U)$$

$$= V \cdot U - \alpha^2 ||U||^2$$

$$\therefore \alpha^* = V \cdot U / ||U||^2.$$  

The Schwarz inequality follows:

$$I_{\text{min}} = ||V - \alpha^* U||^2 = ||V||^2 - \frac{|U \cdot V|^2}{||U||^2} > 0$$

$$0 \leq |U \cdot V| \leq ||U|| \ ||V||$$

An important example of such a vector space includes the definition of the Fourier Transform, where we may set

$$U(\omega) = e^{-j\omega_0 t} \quad V(\omega) = e^{j\omega t} \quad U \cdot V = \int_{\omega} e^{j\omega t} e^{-j\omega_0 t} \frac{d\omega}{2\pi} = \delta(\omega - \omega_0).$$

It seems that the Fourier transform is a result that follows from a minimization, unlike the Laplace transform that follows from causal system parameters. This explains the important differences between the two, in terms of their properties (Unlike the LT, the FT is not complex analytic). We also explored this topic in Lecture 1.4.15 (p. 100).

**Power vs. power series, linear vs. nonlinear**

Another place where equations of second degree appear in physical applications is in energy and power calculations. The electrical power is given by the product of the voltage $v(t)$ and current $i(t)$ (or in mechanics as the force times the velocity). For example if we define $P = v(t)i(t)$ to be the power $P$ [Watts], then the total energy $[\text{Joules}]$ at time $t$ is (Van Valkenburg, 1964a, Chapter 14)

$$E(t) = \int_0^t v(t)i(t)dt.$$
From this observe that the power is the rate of change of the total energy

\[ P(t) = \frac{d}{dt} E(t), \]

reminiscent of the Fundamental Theorem of calculus [Eq. 1.86, (p. 109)].

**Ohm’s law and impedance:** The ratio of voltage over the current is call the *impedance* which has units of [Ohms]. For example given a resistor of \( R = 10 \) [ohms],

\[ v(t) = R i(t). \]

Namely 1 amp flowing through the resistor would give 10 volts across it. Merging the linear relation due to Ohm’s law with the definition of power, shows that the instantaneous power in a resistor is quadratic in voltage and current

\[ P(t) = \frac{v(t)^2}{R} = \frac{i(t)^2}{R}. \] (1.62)

Note that Ohm’s law is linear in its relation between voltage and current whereas the power and energy are *nonlinear*.

Ohm’s law generalizes in a very important way, allowing the impedance (e.g., resistance) to be a linear complex analytic function of complex frequency \( s = \sigma + \omega j \) (Kennelly, 1893; Brune, 1931a). Impedance is a fundamental concept in many fields of engineering. For example: Newton’s second law \( F = ma \) obeys Ohm’s law, with mechanical impedance \( Z(s) = sm \). Hooke’s law \( F = kx \) for a spring is described by a mechanical impedance \( Z(s) = k/s \). In mechanics a “resistor” is called a *dashpot* and its impedance is a positive and real constant.\(^{57}\)

**Kirchhoff’s laws KCL, KVL:** The laws of electricity and mechanics may be written down using Kirchhoff’s laws current and voltage laws, (KCL, KVL), which lead to linear systems of equations in the currents and voltages (velocities and forces) of the system under study, with complex coefficients having positive real parts.

Points of major confusion are a number of terms that are misused, and overused, in the fields of mathematics, physics and engineering. Some of the most obviously abused terms are *linear/nonlinear*, *energy*, *power*, *power series*. These have multiple meanings, which can, and are, fundamentally in conflict.

**Transfer functions (Transfer matrix):** The only method that seems to work, to sort this out, is to cite the relevant physical application, in specific contexts. The most common touch point is a physical system that has an input \( x(t) \) and an output \( y(t) \). If the system is linear, then it may be represented by its *impulse response* \( h(t) \). In such cases the system equation is

\[ y(t) = h(t) \star x(t) \leftrightarrow Y(\omega) = H(s)|_{\sigma=0} X(\omega), \]

namely the convolution of the input with the impulse response gives the output. From Fourier analysis this method may be written in the real frequency domain as a product of the Laplace

\(^{56}\)In acoustics the pressure is a potential, like voltage. The force per unit area is given by \( f = -\nabla p \) thus \( F = -\int \nabla p \, dS \). Velocity is analogous to a current. In terms of the velocity potential, the velocity per unit area is \( v = -\nabla \phi \).

\(^{57}\)https://en.wikipedia.org/wiki/Impedance_analogy
transform of the impulse response, evaluated on the $\omega j$ axis and the Fourier transform of the input $X(\omega) \leftrightarrow x(t)$ and output $Y(\omega) \leftrightarrow y(t)$.

Mention ABCD Transfer matrix

If the system is nonlinear, then the output is not given by a convolution, and the Fourier and Laplace transforms have no obvious meaning.

The question that must be addressed is why is the power said to be nonlinear whereas a power series of $H(s)$ said to be linear. Both have powers of the underlying variables. This is massively confusing, and must be addressed. The question will be further addressed in Section 3.5.1 in terms of the system postulates of physical systems.

What's going on? The domain variables must be separated from the codomain variables. In our example, the voltage and current are multiplied together, resulting in a nonlinear output, the power. If the frequency is squared, this is describing the degree of a polynomial. This is not nonlinear because it does not impact the signal output, it characterizes the Laplace transform of the system response.

1.4.10 Lec 15 Gaussian Elimination

The method for finding the intersection of equations is based on the recursive elimination of all the variables but one. This method, known as Gaussian elimination (Appendix ??, p. ??), works across a broad range of cases, but may be defined as a systematic algorithm when the equations are linear in the variables.\(^{58}\) Rarely do we even attempt to solve problems in several variables of degree greater than 1. But Gaussian eliminations may still work in such cases (Stillwell, 2010, p. 90).

In Appendix B.3 (p. 262) the inverse of a 2x2 linear system of equations is derived. Even for a 2x2 case, the general solution requires a great deal of algebra. Working out a numeric example of Gaussian elimination is more instructive. For example, suppose we wish to find the intersection of the two equations

\[
\begin{align*}
x - y &= 3 \\
2x + y &= 2.
\end{align*}
\]

This 2x2 system of equations is so simple that you may immediately visualize the solution: By adding the two equations, $y$ is eliminated, leaving $3x = 5$. But doing it this way takes advantage of the specific example, and we need a method for larger systems of equations. We need a generalized (algorithmic) approach. This general approach is called Gaussian elimination.

Start by writing the equations in matrix format (note this is not in the standard form $Ax = y$)

\[
\begin{bmatrix}
1 & -1 \\
2 & 1
\end{bmatrix}
\begin{bmatrix}
x \\
y
\end{bmatrix} =
\begin{bmatrix}
3 \\
2
\end{bmatrix}.
\]

Next, eliminate the lower left term ($2x$) using a scaled version of the upper left term ($x$). Specifically, multiply the first equation by $-2$, add it to the second equation, replacing the second equation with the result. This gives

\[
\begin{bmatrix}
1 & -1 \\
0 & 3
\end{bmatrix}
\begin{bmatrix}
x \\
y
\end{bmatrix} =
\begin{bmatrix}
3 \\
2 - 3 \cdot 2
\end{bmatrix} =
\begin{bmatrix}
3 \\
-4
\end{bmatrix}.
\]

Note that the top equation did not change. Once the matrix is “upper triangular” (zero below the diagonal) you have the solution. Starting from the bottom equation, \( y = -\frac{4}{3} \). Then the upper equation then gives \( x - \left(-\frac{4}{3}\right) = 3 \), or \( x = 3 - 4/3 = 5/3 \).

In principle, Gaussian elimination is easy, but if you make a calculation mistake along the way, it is very difficult to find the error. The method requires a lot of mental labor, with a high probability of making a mistake. Thus you do not want to apply this method every time. For example suppose the elements are complex numbers, or polynomials in some other variable such as frequency. Once the coefficients become more complicated, the seeming trivial problem becomes highly error prone. There is a much better way, that is easily verified, which puts all the numerics at the end, in a single step.

The above operations may be automated by finding a carefully chosen upper-diagonalization matrix \( U \) that does the same operation. For example let

\[
U = \begin{bmatrix} 1 & 0 \\ -2 & 1 \end{bmatrix}.
\]

(1.65)

Multiplying Eq. 1.63 by \( U \) we find

\[
\begin{bmatrix} 1 & 0 \\ -2 & 1 \end{bmatrix} \begin{bmatrix} 1 & -1 \\ 2 & 1 \end{bmatrix} \begin{bmatrix} x \\ y \end{bmatrix} = \begin{bmatrix} 1 & -1 \\ 0 & 3 \end{bmatrix} \begin{bmatrix} x \\ y \end{bmatrix} = \begin{bmatrix} 3 \\ -4 \end{bmatrix}
\]

we obtain Eq. 1.64. At this point we can either back-substitute and obtain the solution, as we did above, or find a matrix \( L \) that finishes the job, by removing elements above the diagonal.

**Exercise:** Show that \( \det(U) = \det(UA) = 1 \). **Sol:** Since

\[
\det \begin{bmatrix} a & b \\ c & d \end{bmatrix} = ad - bc,
\]

and \( \det(UA) = \det U \cdot \det(A) \), and \( \det(U) = 1 - 0 = 1 \), the result follows naturally. Common notation is to let \( \det(A) = |A| \).

In Appendix B.3 the inverse of a general 2x2 matrix is summarized in terms of three steps: 1) swap the diagonal elements, 2) reverse the signs of the off-diagonal elements and 3) divide by the determinant \( \Delta = ab - cd \). Specifically

\[
\begin{bmatrix} a & b \\ c & d \end{bmatrix}^{-1} = \frac{1}{\Delta} \begin{bmatrix} d & -b \\ -c & a \end{bmatrix}.
\]

(1.67)

There are very few things that you must memorize, but the inverse of a 2x2 is one of them. It needs to be in your tool-bag of tricks, like the quadratic formula.

While it is difficult to compute the inverse matrix from scratch (Appendix B), it takes only a few seconds to verify it (steps 1 and 2)

\[
\begin{bmatrix} a & b \\ c & d \end{bmatrix} \begin{bmatrix} d & -b \\ -c & a \end{bmatrix} = \begin{bmatrix} ad - bc & -ab + ab \\ cd - cd & -bc + ad \end{bmatrix} = \begin{bmatrix} \Delta & 0 \\ 0 & \Delta \end{bmatrix}.
\]

(1.68)

Thus dividing by the determinant gives the 2x2 identity matrix. A good strategy, when you don’t trust your memory, is to write down the inverse as best you can, and then verify.
Using the 2x2 matrix inverse on our example (Eq. 1.63), we find
\[
\begin{bmatrix}
x
y
\end{bmatrix} = \frac{1}{1 + 2} \begin{bmatrix} 1 & 1 \\ -2 & 1 \end{bmatrix} \begin{bmatrix} 3 \\ 2 \end{bmatrix} = \frac{1}{3} \begin{bmatrix} 5 \\ -6 + 2 \end{bmatrix} = \begin{bmatrix} 5/3 \\ -4/3 \end{bmatrix}.
\] (1.69)

If you use this method, you will rarely (never) make a mistake, and the solution is easily verified. Either you can check the numbers in the inverse, as was done in Eq. 1.68, or you can substitute the solution back into the original equation.

1.4.11 Lec 16: Transmission (ABCD) matrix composition method

![Diagram of LC transmission line](image.png)

Figure 1.18: A single LC cell of the LC transmission line (see Fig. 2.2 (p. 189). Every cell of any transmission line may be modeled by the ABCD method, as the product of two matrices. For the example shown here, the inductance \( L \) of the coil and the capacitance \( C \) of capacitor are in units of [Henry/m] and [Farad/m], thus they depend on length \( \Delta_x [m] \) that the cell represents. Note the flows are always defined as into the + node.

Matrix composition: Matrix multiplication represents a composition of 2x2 matrices, because the input to the second matrix is the output of the first (this follows from the definition of composition: \( f(x) \circ g(x) = f(g(x)) \)). Thus the ABCD matrix is also known as the transmission matrix method, or occasionally the chain matrix. The general expression for an transmission matrix \( T(s) \) is
\[
\begin{bmatrix}
V_1 \\
I_1
\end{bmatrix} = \begin{bmatrix} A(s) & B(s) \\ C(s) & D(s) \end{bmatrix} \begin{bmatrix} V_2 \\
-I_2
\end{bmatrix}.
\] (1.70)

The four coefficients \( A(s), B(s), C(s), D(s) \) are all complex functions of the Laplace frequency \( s = \sigma + j\omega \) (p. 1). The derivation is repeated with more detail in Section 3.3.2 (p. 208).

It is a standard convention to always define the current into the node, but since the input current (on the left) is the same as the output current on the right (\( I_2 \)), hence the negative sign on \( I_2 \), to meet the convention of current into every node. When transmission matrices are cascaded, all the signs then match.

We have already used 2x2 matrix composition in representing complex numbers (p. 32), and for computing the \( \gcd(m,n) \) of \( m, n \in \mathbb{N} \) (p. 42), Pell’s equation (p. 50) and the Fibonacci sequence (p. 53).

Definitions of \( A, B, C, D \): The definitions of the four functions of Eq. 1.70 are easily read off of the equation, as
\[
\begin{align*}
\mathcal{A}(s) &= \left. \frac{V_1}{V_2} \right|_{I_2=0}, & \mathcal{B}(s) &= -\left. \frac{V_1}{I_2} \right|_{V_2=0}, & \mathcal{C}(s) &= \left. \frac{I_1}{V_2} \right|_{I_2=0}, & \mathcal{D}(s) &= -\left. \frac{I_1}{I_2} \right|_{V_2=0}.
\end{align*}
\] (1.71)

These definitions follow trivially from Eq. 1.70. These relations have general names. \( \mathcal{A}, \mathcal{D} \) are called voltage and current transfer functions, since they are the ratio of an output over an input,
whereas $B, C$ are known as the transfer impedance and transfer conductance. Each express an output (port 2) in terms of an input input at port 1.

**Exercise:** Explain why $C$ is given as above. **Sol:** Writing out the lower equation $I_1 = CV_2 - DI_2$. Setting $I_2 = 0$, we obtain the equation for $C$.

**Thévenin parameters of a source:** A very important concept in circuit theory is that of the Thévenin parameters (p. 92), the open-circuit voltage and the short-circuit current, the ratio of which define the Thévenin impedance. The open circuit voltage is defined as the voltage when the current is zero. In mechanics this is called the isometric force, the maximum force given zero velocity.

It trivially follows that the Thévenin voltage (open circuit voltage) is

$$V_{\text{Thév}} = \frac{1}{C} \equiv \frac{V_2}{I_1} \bigg|_{I_2=0}.$$

For example, the voltage measured across a battery having no load is its Thévenin voltage.

We may find the impedance $Z_1 = \frac{V_1}{I_1} \bigg|_{I_2=0}$ that would be seen looking into port 1, with nothing connected to port 2 ($I_2 = 0$) by taking the ratio of the upper and lower equations

$$Z_1 = \frac{AV_2 - BI_2}{CV_2 - DI_2} \bigg|_{I_2=0} = \frac{A}{C} = \frac{AV_{\text{Thév}}}{C}.$$

Using this relation we may find a second expression for the Thévenin voltage in terms of the a source voltage $V_1$ rather than in terms of the source current.

$$V_{\text{Thév}} = \frac{Z_1}{A} \equiv \frac{Z_1}{A} \frac{V_2}{V_1} \bigg|_{I_2=0}.$$

The choice of the appropriate expression depends on the physics of the actual source.

**Exercise:** Show that the Thévenin source impedance is

$$Z_{\text{Thév}} = \frac{b}{C}.$$

**Hint:** Use the fact that $V_1 = -BI_2 \big|_{V_2=0}$

**Properties of the transmission matrix:** The transmission matrix is always constructed as the product of elemental matrices of the form

$$\begin{bmatrix} 1 & Z(s) \\ 0 & 1 \end{bmatrix} \quad \text{or} \quad \begin{bmatrix} 1 & 0 \\ Y(s) & 1 \end{bmatrix}.$$

Since the determinant of each of these matrices is 1, the determinant, the product of many elemental matrices being 1, must be one. Thus for the case of reciprocal system (P6) (p. 104)

$$\det \begin{bmatrix} A(s) & B(s) \\ C(s) & D(s) \end{bmatrix} = 1.$$

Anti reciprocal system may be synthesized by the use of a gyrator, and in these cases $\Delta_T = -1$.  
1.4.12 The impedance matrix

With a bit of algebra, one may find the impedance matrix in terms of $A, B, C, D$ (Van Valkenburg, 1964a, p. 310)

$$
\begin{bmatrix}
V_1 \\
V_2
\end{bmatrix} = \begin{bmatrix}
z_{11} & z_{12} \\
z_{21} & z_{22}
\end{bmatrix}
\begin{bmatrix}
I_1 \\
I_2
\end{bmatrix}
= \frac{1}{C} \begin{bmatrix}
A & \Delta_T \\
1 & D
\end{bmatrix}
\begin{bmatrix}
I_1 \\
I_2
\end{bmatrix}.
$$

(1.72)

For reciprocal systems (P6, p. 104) $z_{12} = z_{21}$ since $\Delta_T = 1$. For anti-reciprocal systems, such as dynamic loudspeakers and microphones (Kim and Allen, 2013), $z_{21} = -z_{12} = 1/C$, since $\Delta_T = -1$. Not the primary definition

Impedance is a very general concept, closely tied to the definition of power $P(t)$ (and energy). Power is defined as the product of the effort (force) and the flow (current). As described in Fig. 1.19, these concepts are very general, applying to mechanics, electrical circuits, acoustics, thermal circuits, or any other case where conservation of energy applies. Two basic variables are defined, generalized force and generalized flow, also called conjugate variables. The product of the conjugate variables is the power, and the ratio is the impedance. For example for the case of voltage and current,

$$
P(t) \equiv \int v(t)i(t)dt, \quad Z(s) \equiv \frac{V(\omega)}{I(\omega)}.
$$

**Ohm’s law** In general, impedance is defined as the ratio of a force over a flow. For electrical circuits, the voltage is the ‘force’ and the current is the ‘flow.’ Ohm’s law states that the voltage across and the current through a circuit element are related by the impedance of that element (which is typically a function of the Laplace frequency $s = \sigma + \omega j$). For resistors, the voltage over the current is called the resistance, and is a constant (e.g. the simplest case, $V/I = R$). For inductors and capacitors, the impedance depends on the Laplace frequency $s$ [e.g. $V/I = Z(s)$]).

<table>
<thead>
<tr>
<th>Case</th>
<th>Force</th>
<th>Flow</th>
<th>Impedance</th>
<th>units</th>
</tr>
</thead>
<tbody>
<tr>
<td>Electrical</td>
<td>voltage (V)</td>
<td>current (I)</td>
<td>$Z = V/I$</td>
<td>Ohms [$\Omega$]</td>
</tr>
<tr>
<td>Mechanics</td>
<td>force (F)</td>
<td>velocity (U)</td>
<td>$Z = F/U$</td>
<td>Mechanical Ohms [$\Omega$]</td>
</tr>
<tr>
<td>Acoustics</td>
<td>pressure (P)</td>
<td>particle velocity (V)</td>
<td>$Z = P/V$</td>
<td>Acoustic Ohms [$\Omega$]</td>
</tr>
<tr>
<td>Thermal</td>
<td>temperature (T)</td>
<td>heat-flux (J)</td>
<td>$Z = T/J$</td>
<td>Thermal Ohms [$\Omega$]</td>
</tr>
</tbody>
</table>

Figure 1.19: Impedance is defined as the ratio of a force over a flow, a concept also holds in mechanics and acoustics. In mechanics, the ‘force’ is equal to the mechanical force on an element (e.g. a mass, dash-pot, or spring), and the ‘flow’ is the velocity. In acoustics, the ‘force’ is pressure, and the ‘flow’ is the volume velocity or particle velocity of air molecules.

As discussed in Fig. 1.19, the impedance concept also holds for mechanics and acoustics. In mechanics, the ‘force’ is equal to the mechanical force on an element (e.g. a mass, dash-pot, or spring), and the ‘flow’ is the velocity. In acoustics, the ‘force’ is pressure, and the ‘flow’ is the volume velocity or particle velocity of air molecules.

In this section we shall derive the method of linear composition of systems, known by several names as the $ABCD$ Transmission matrix method, or in the mathematical literature as the Möbius (bilinear) transformation. Using the method of matrix composition, a linear system of 2x2 matrices can represent a large and important family of networks. By the application of Ohm’s law to the circuit shown in Fig. 1.18, we can model a cascade of such cells.
Example of the use of the ABCD matrix composition: In Fig. 1.18 we see the network is composed of a series inductor (mass) having an impedance $Z_l = sL$, and a shunt capacitor (compliance) having an impedance $Z_c = 1/sC$, where $s \in \mathbb{C}$. As determined by Ohm’s law, each impedance is describe by a linear relation between the current and the voltage. For the inductive impedance, applying Ohm’s law, we find

$$V_1 - V_2 = Z_l I_1$$

where $Z_l$ is the complex impedance of the inductor. For the capacitive impedance, applying Ohm’s law we find

$$V_2 = (I_1 + I_2)Z_c,$$

where $Z_c$ is the complex impedance of the capacitor.

Each of these linear impedance relations may be written in matrix form. The series inductor equation gives (note $I_1 = -I_2$)

$$\begin{bmatrix} V_1 \\ I_1 \end{bmatrix} = \begin{bmatrix} 1 & Z_l \\ 0 & 1 \end{bmatrix} \begin{bmatrix} V_2 \\ -I_2 \end{bmatrix},$$

while the shunt capacitor equation yields (note $V_1 = V_2$)

$$\begin{bmatrix} V_2 \\ I_1 \end{bmatrix} = \begin{bmatrix} 1 & 0 \\ Y_c & 1 \end{bmatrix} \begin{bmatrix} V_2 \\ -I_2 \end{bmatrix},$$

where $Y_c = 1/Z_c$ is called the admittance.

When the second matrix equation for the shunt admittance (Eq. 1.74) is substituted into the series impedance equation (Eq. 1.73), we find the ABCD matrix ($T_1 \circ T_2$), for the cell is simply the product of two matrices

$$\begin{bmatrix} V_1 \\ I_1 \end{bmatrix} = \begin{bmatrix} 1 & Z_l \\ 0 & 1 \end{bmatrix} \begin{bmatrix} 1 + Z_l Y_c & Z_l \\ Y_c & 1 \end{bmatrix} \begin{bmatrix} V_2 \\ -I_2 \end{bmatrix}. $$

(1.75)

Note that the determinant of the matrix $\Delta = AD - BC = 1$.

Thus $A(s) = 1 + Z_L u_c = 1 + s^2 L C$, $B(s) = Z_l$, $C(s) = Y_c$ and $D = 1$. This equation characterizes every possible relation between the input and output voltage and current of the cell.

For example, the ratio of the output to input voltage with the output unloaded ($I_2 = 0$), known as the voltage divider relation may be found from the upper equation with $I_2 = 0$. Writing this out gives

$$\left. \frac{V_2}{V_1} \right|_{I_2=0} = \frac{1}{A(s)} = \frac{1}{1 + Z_l Y_c} = \frac{Z_c}{Z_c + Z_l}.$$

To derive the formula for the current divider equation, use the lower equation and set $V_2 = 0$. This gives

$$\left. \frac{-I_2}{I_1} \right|_{V_2=0} = \frac{1}{D} = 1.$$
key development, heavily used by both Newton and Euler. Also the investigations of Cauchy made important headway with his work on complex variables. Of special note was integration and differentiation in the complex plane of complex analytic functions, which is the topic of stream 3.

It was Riemann, working with Gauss in the final years of Gauss’ life, who made the breakthrough, with the concept of the extended complex plane.\textsuperscript{59} This concept was based on the composition of a line with the sphere, similar to the derivation of Euclid’s formula for Pythagorean triplets (Fig. 2.3, p. 191). While the importance of the extended complex plane was unforeseen, it changed analytic mathematics forever, along with the physics it supported. It unified and thus simplified many important integrals, to the extreme. This idea is captured by the fundamental theorem of complex integral calculus (Table 1.5 p. 168) and 1.5, p. 106.

\textbf{Figure 1.20: }The left panel shows how the real line may be composed with the circle. Each real \( x \) value maps to a corresponding point \( x' \) on the unit circle. The point \( x \to \infty \) maps to the north pole \( N \). This simple idea may be extended with the composition of the complex plane with the unit sphere, thus mapping the plane onto the sphere. As with the circle, the point on the complex plane \( z \to \infty \) maps onto the north pole \( N \). This construction is important because while the plane is open (does not include \( z \to \infty \)), the sphere is analytic at the north pole. Thus the sphere defines the closed extended plane. Figure from Stillwell (2010, pp. 299-300).

The idea is outlined in Fig. 1.20. On the left is a circle and a line. The difference between this case and the derivation of the Pythagorean Triplets is, that the line starts at the north pole, and ends on the real \( x \in \mathbb{R} \) axis at point \( x \). At point \( x' \), the line cuts through the circle. Thus the mapping from \( x \) to \( x' \) takes every point on \( \mathbb{R} \) to a point on the circle. For example, the point \( x = 0 \) maps to the south pole (not indicated). To express \( x' \) in terms of \( x \) one must compose the line and the circle, similar to the composition used in Fig. 2.3 (p. 191). The points on the circle, indicated here by \( x' \), require a traditional polar coordinate system, having a unit radius and an angle defined between the radius and a vertical line passing through the north pole. When \( x \to \infty \) the point \( x' \to N \), known as the point at infinity. But this idea goes much further, as shown on the right half of Fig. 1.20.

Here the real tangent line is replaced by the a tangent complex plane \( z \in \mathbb{C} \), and the complex puncture point \( z' \in \mathbb{C} \), in this case on the complex sphere, called the extended complex plane. This is a natural extension of the chord/tangent method on the left, but with significant consequences. The main difference between the complex plane \( z \) and the extended complex plane, other than the coordinate system, is what happens at the north pole. The point at \( |z| = \infty \) is not defined on the plane, whereas on the sphere, the point at the north pole is simply another point, like every other point on the sphere.

\textsuperscript{59}“Gauss did lecture to Riemann but he was only giving elementary courses and there is no evidence that at this time he recognized Riemann’s genius.” http://www-groups.dcs.st-and.ac.uk/~history/Biographies/Riemann.html “In 1849 he [Riemann] returned to Göttingen and his Ph.D. thesis, supervised by Gauss, was submitted in 1851.” http://www-groups.dcs.st-and.ac.uk/~history/Biographies/Riemann.html
Open vs. closed sets: Mathematically the plane is said to be an open set, since the limit \( z \to \infty \) is not defined, whereas on the sphere, the point \( z' \) is a member of a closed set, since the north pole is defined. The distinction between an open and closed set is important, because the closed set allows the function to be complex-analytic at the north pole, which it cannot be on the plane (since the point at infinity is not defined).

The \( z \) plane may be replaced with another plane, say the \( w = F(z) \in \mathbb{C} \) plane, where \( w \) is some function \( F \) of \( z \in \mathbb{C} \). For the moment we shall limit ourselves to complex analytic functions of \( z \), namely \( w = F(z) = u(x, y) + v(x, y)j = \sum_{n=0}^{\infty} c_n z^n \).

In summary, given a point \( z = x + yj \) on the open complex plane, we map it to \( w = F(z) \in \mathbb{C} \), the complex \( w = u + vj \) plane, and from there to the closed extended complex plane \( w'(z) \). The point of doing this is that it allows us to allow the function \( w'(z) \) to be analytic at the north pole, meaning it can have a convergent Taylor series at the point at infinity \( z \to \infty \). Since we have not yet defined \( \frac{dw(z)}{dz} \), the concept of a complex Taylor series remains undefined.

**Bilinear transformation**

In mathematics the bilinear transformation has special importance because it is linear in its action on both the input and output variables. Since we are engineers we shall stick with the engineering terminology. But if you wish to read about this on the internet, be sure to also search for the mathematical term, Möbius transformation.

When a point on the complex plane \( z = x + yj \) is composed with the bilinear transform \((a, b, c, d \in \mathbb{C})\), the result is \( w(z) = u(x, y) + v(x, y)j \)

\[
w = \frac{az + b}{cz + d} \quad \text{(1.76)}
\]

the transformation from \( z \to w \) is a cascade of four independent compositions

1. translation \((w = z + b; \ a = 1, b \in \mathbb{C}, c = 0, d = 1)\),
2. scaling \((w = |a|z; \ a \in \mathbb{R}, b = 0, c = 0, d = 1)\)
3. rotation \((w = \frac{a}{|a|}z; \ a \in \mathbb{C}, b = 0, c = 0, d = |a|)\) and
4. inversion \((w = \frac{1}{z}; \ a = 0, b = 1, c = 1, d = 0)\).

Each of these transformations are a special case of Eq. 1.76, with the inversion the most complicated. A wonderful video showing the effect of the bilinear (Möbius) transformation on the plane is available that I highly recommend: [https://www.youtube.com/watch?v=0z1fIsUNhO4](https://www.youtube.com/watch?v=0z1fIsUNhO4)

The bilinear transformation is the most general way to move the expansion point in a complex analytic expansion. For example starting from the harmonic series, the bilinear transform gives

\[
\frac{1}{1-w} = \frac{1}{1-\frac{az+b}{cz+d}} = \frac{cz+d}{(c-a)z+(d-b)}.
\]

The RoC is transformed from \(|w| < 1\) to \(|(az-b)/(cz-d)| < 1\). An interesting application might be in moving the expansion point until it is on top of the nearest pole, so that the RoC goes to zero. This might be a useful way of finding a pole, for example.
CHAPTER 1. INTRODUCTION

When the extended plane (Riemann sphere) is analytic at \( z = \infty \), one may take the derivatives there, defining the Taylor series with the expansion point at \( \infty \). When the bilinear transformation rotates the Riemann sphere, the point at infinity is translated to a finite point on the complex plane, revealing the analytic nature at infinity. A second way to transform the point at infinity by the bilinear transformation \( \zeta = 1/z \), mapping a zero (or pole) at \( z = \infty \) to a pole (or zero) at \( \zeta = 0 \). Thus this construction of the Riemann sphere and the Möbius (bilinear) transformation allow us to understand the point at infinity, and treat it like any other point. If you felt that you never understood the meaning of the point at \( \infty \) (likely), this should help.

1.4.14 Lec 18: Complex analytic mappings (Domain-coloring)

One of the most difficult aspects of complex functions of a complex variable is understanding the mapping from \( z = x + yj \) to \( w(z) = u + vj \). For example, \( w(z) = \sin(x) \) is trivial when \( z = x + yj \) is real (i.e., \( y = 0 \)), because \( \sin(x) \) is real. Likewise for the case where \( x = 0 \), where

\[
\sin(yj) = \frac{e^{-y} - e^{y}}{2j} = j\sinh(y)
\]

is purely imaginary. But the general case, \( w(z) = \sin(z) \in \mathbb{C} \)

\[
\sin(zj) = \sin(xj - y) = j\sinh(z).
\]

is not easily visualized. Thus when \( u(x, y) \) and \( v(x, y) \) are not well known functions, \( w(z) \) are can be much more difficult to visualize.

Fortunately with computer software today, this problem can be solved by adding color to the chart. A Matlab/Octave script \texttt{zviz.m} has been used to make the charts shown here.\textsuperscript{60} This tool is also known as Domain-coloring.\textsuperscript{61} Rather than plotting \( u(x, y) \) and \( v(x, y) \) separately, domain-coloring allows us to display the entire function on one chart. Note that for this visualization we see the complex polar form of \( w(s) = |w|e^{j\phi} \), rather than as the four dimensional Cartesian graph \( w(x + yj) = u(x, y) + v(x, y)j \).

**Visualizing complex functions:** The mapping from \( z = x + iy \) to \( w(z) = u(x, y) + iv(x, y) \) is a \( 2 \cdot 2 = 4 \) dimensional graph. This is difficult to visualize, because for each point in the domain \( z \), we would like to represent both the magnitude and phase (or real and imaginary parts) of \( w(z) \). A good way to visualize these mappings is to use color (hue) to represent the phase and intensity (dark to light) to represent the magnitude. The Matlab

\[\text{Figure 1.21: LEFT: Domain–colorized map showing the complex mapping from the } s = \sigma + \omega \text{ plane to the } w(s) = u(\sigma, \omega) + v(\sigma, \omega)j \text{ plane. This mapping may be visualized by the use of intensity (light/dark) to indicate magnitude, and color (hue) to indicate angle (phase) of the mapping. RIGHT: This shows the } w(z) = z - \sqrt{3} \text{ plane, shift to the right and up by } \sqrt{2}/2 = 0.707. \text{ The white and black lines are the iso-real and iso-imaginary contours of the mapping.}
\]

\[\text{Figure 1.22: Plot of } \sin(0.5\pi(z - i)).\]
Figure 1.23: This domain–color map allows one to visualize complex mappings by the use of intensity (light/dark) to indicate magnitude, and color (hue) to indicate angle (phase). The white and black lines are the iso-real and iso-imaginary contours of the mapping. LEFT: This figure shows the domain–color map for the complex mapping from the $z = x + iy$ plane to the $w(z) = u + iv = e^{z}$ plane, which goes to zero as $x \to -\infty$, causing the domain–color map to become dark for $x < -2$. The white and black lines are always perpendicular because $e^{z}$ is complex analytic everywhere. RIGHT: This shows the principal value of the inverse function $w(x, y) + v(x, y)j = \log(x + yj)$, which has a zero (dark) at $z = 1$, since there $\log(1) = 0$ (the imaginary part is zero). Note the branch cut from $x = 0$ to $x = -\infty$. On branches other than the one shown, there is are no zeros, since the phase ($\angle z = 2\pi n$) is not zero.

program $zviz$.m has been provided to do this (see Lecture 17 on the class website).

To use the program in Matlab/Octave, use the syntax $zviz <function of z>$ (for example, type $zviz z$^2). Note the period between $z$ and ^2. This will render a ‘domain coloring’ (aka colorized) version of the function. Examples you can render are given in the comments at the top of the $zviz$.m program. A good example for testing is $zviz z - \sqrt{j}$, which should show a dark spot (a zero) at $(1+j)/\sqrt{2} = 0.707(1+j)$.

Example: Figure 1.22 shows a colorized plot of $w(z) = \sin(\pi(z-i)/2)$ resulting from the Matlab/Octave command $zviz \sin(pi*(z-i)/2)$. The abscissa (horizontal axis) is the real $x$ axis and the ordinate (vertical axis) is the complex $iy$ axis. The graph is offset along the ordinate axis by $1i$, since the argument $z-i$ causes a shift of the sine function by $1$ in the positive imaginary direction. The visible zeros of $w(z)$ appear as dark regions at $(-2, 1), (0, 1), (2, 1)$. As a function of $x$, $w(x+1j)$ oscillates between red (phase is zero degrees), meaning the function is positive and real, and sea-green (phase is $180^\circ$), meaning the function is negative and real.

Along the vertical axis, the function is either a $\cosh(y)$ or $\sinh(y)$, depending on $x$. The intensity becomes lighter as $|w|$ increases, and white as $w \to \infty$. The intensity becomes darker as $|w|$ decreases, and black as $w \to 0$.

Mathematicians typically use the more abstract (i.e., non–physical) notation $w(z)$, where $w = u + iv$ and $z = x + iy$. Engineers think in terms of a physical complex impedance $Z(s) = R(s) + X(s)j$, having resistance $R(s)$ and reactance $X(s)$ [Ohms], as function of the complex Laplace radian frequency $s = \sigma + \omega j$ [rad], as used, for example, with the Laplace transform (Sect. 1.4.16, p. 101). In Fig. 1.21 we use both notations, with $Z(s) = s$ on the left and $w(z) = z - \sqrt{j}$ on the right, where we show this color code as a 2x2 dimensional domain-coloring graph. Intensity (dark to light) represents the magnitude of the function, while hue (color) represents the phase, where (see Fig. 1.21) red is $0^\circ$, sea-green is $90^\circ$, blue-green is $135^\circ$, blue is $180^\circ$, and violet is $-90^\circ$ (or $270^\circ$). The function $w = s$ has a dark spot (a zero) at $s = 0$, and becomes brighter away from the
origin. On the right is \( w(z) = z - \sqrt{j} \), which shifts the zero to \( z = \sqrt{j} \). Thus domain–coloring gives the full picture of the complex analytic function mappings \( w(x, y) = u(x, y) + v(x, y)j \) in colorized polar coordinates.

Two additional examples are given in Fig. 1.23 to help you interpret the two complex mappings \( w = e^s \) (left) and its inverse \( s = \ln(w) \). The exponential is relatively easy to understand because \( w(s) = |e^s e^{\omega t}| = e^\sigma \). The red region is where \( \omega \approx 0 \) in which case \( w \approx e^\sigma \). As \( \sigma \) becomes large and negative, \( w \to 0 \), thus the entire field becomes dark on the left. The field is becoming light on the right where \( w = e^\sigma \to \infty \). If we let \( \sigma = 0 \) and look along the \( \omega \) axis, we see that the function is changing phase, sea-green (90°) at the top and violet (-90°) at the bottom.

In the right panel note the zero for \( \ln(w) = \ln|w| + \omega \sigma \) at \( w = 1 \). The root of the log function is \( \log(w) = 0 \) is \( w_r = 1, \phi = 0 \), since \( \log(1) = 0 \). More generally, the log of \( w = |w|e^{\phi j} \) is \( s = \ln|w| + \phi \sigma \). Thus \( s(w) \) can be zero only when the angle of \( w \) is zero.

The \( \ln(w) \) function has a branch cut along the \( \phi = 180^\circ \) axis. As one crosses over the cut, the phase goes above 180°, and the plane changes to the next sheet of the log function. The only sheet with a zero is the principle value, as shown. All others, the log function is either increasing or decreasing monotonically, and there is no zero, as seen on sheet 0 (the one showing in Fig. 1.23).

1.4.15 Lec 19: Signals: Fourier transforms

Two basic transformations in engineering mathematics are the Fourier and the Laplace transforms, which deal with time–frequency analysis.

**Notation:** The Fourier transform takes a time domain signal \( f(t) \in \mathbb{R} \) and transforms it to the frequency domain \( \omega \in \mathbb{R} \), where it is complex \((F(\omega) \in \mathbb{C})\). For the Fourier transform, both the time \(-\infty < t < \infty\) and frequency \(\infty < \omega < \infty\) are strictly real. The relationship, between \( f(t) \) and its transform \( F(\omega) \), is indicated by the double arrow symbol

\[
f(t) \leftrightarrow F(\omega).
\]

Since the FT obeys superposition, it is possible to define a FT of a complex time function \( f(t) \in \mathbb{C}, t \in \mathbb{R} \). This is useful in certain engineering applications (i.e., Hilbert envelope, Hilbert transforms). It is accepted in the engineering and Physics literature to use the case of the variable to indicate the type of argument. A time function is \( f(t) \), where \( t \) has units of seconds [s], is in lower case, whereas its Fourier transform, a function of frequency, having units of either Hertz [Hz] or radians [rad] is written using upper case \( F(\omega) \). This helps the reader parse the type of variable under consideration. This is a helpful piece of notation, but not entirely in agreement with notation used in Mathematics.

**Definition of the Fourier transform:** The forward transform takes \( f(t) \) to \( F(\omega) \) while the inverse transform takes \( F(\omega) \) to \( f(t) \). The tilde symbol indicates that in general recovered inverse transform signal can be slightly different from \( f(t) \). We give examples of this in Table F.2.

\[
F(\omega) = \int_{-\infty}^{\infty} f(t) e^{-j\omega t} dt = \frac{1}{2\pi} \int_{-\infty}^{\infty} F(\omega) e^{j\omega t} d\omega
\]

\[
F(\omega) \leftrightarrow f(t) \quad \tilde{f}(t) \leftrightarrow F(\omega).
\]
Periodic signals: Besides these two basic types of time–frequency transforms, there are several variants that depend on the symmetry in time and frequency. For example, when the time signal is sampled (discrete in time), the frequency response becomes periodic, leading to the Discrete-time Fourier transform (DTFT). When a time response is periodic, the frequency response is sampled (discrete in frequency), leading to the Fourier Series. These two symmetries may be simply characterized as periodic in time ⇒ discrete in frequency, and periodic in frequency ⇒ discrete in time. When a function is both discrete in time and frequency, it is necessarily periodic in time and frequency, leading to the Discrete Fourier Transform (DFT). The DFT is typically computed with an algorithm called the Fast Fourier Transform (FFT), which can dramatically speed up the calculation when the data is a power of 2 in length.

A very important symmetry is when functions that are causal (in time) and periodic (in frequency). The best known example is the class of signals that have z-transforms, which are causal (one-sided in time) discrete-time signals. The harmonic series (Eq. 1.42, p. 70) is the z-transform of the discrete-time step function, thus, by symmetry, analytic within the RoC in the complex frequency (z) domain.

The double brackets on \( f((t))_{T_o} \) indicates that \( f(t) \) is periodic in \( t \) with period \( T_o \), i.e., \( f(t) = f(t + kT_o) \) for all \( k \in \mathbb{N} \). Averaging over one period and dividing by the \( T_o \) gives the average value of \( f(t) \).

Exercise: Consider the \( \mathcal{F} \) as a scalar (dot) product (Eq. 1.56, p. 83) between “vectors” \( f(t) \) and \( e^{-j\omega t} \)

\[
f(t) \cdot e^{-j\omega k t} = \frac{1}{T_o} \int_0^{T_o} f(t)e^{-j\omega k t} dt,
\]

where \( \omega_0 = 2\pi/T_o \) and \( \omega_k = k\omega_o \). If \( f(t) = e^{j\omega n t} \) what is the value of the scalar product? Sol: Evaluating the scalar product we find

\[
e^{j\omega n t} \cdot e^{-j\omega k t} = \frac{1}{T_o} \int_0^{T_o} e^{j\omega n t} e^{-j\omega k t} dt = \frac{1}{T_o} \int_0^{T_o} e^{2\pi j(n-k)t/T_o} dt = \begin{cases} 1 & n = k \\ 0 & n \neq k \end{cases}.
\]

The two signals (vectors) are orthogonal.

1.4.16 Lec 20: Systems: Laplace transforms

The Laplace transform takes a real signals \( f(t) \in \mathbb{R} \), as a function of real time \( t \in \mathbb{R} \), that are causal, i.e., strictly zero for negative time \( f(t) = 0 \) for \( t < 0 \), and transforms them into complex functions \( F(s) \in \mathbb{C} \) of complex frequency \( s = \sigma + \omega j \). As for the Fourier transform, there is the notation \( f(t) \leftrightarrow F(s) \).

When a signal is zero for negative time \( f(t < 0) = 0 \), it is said to be causal, and the resulting transform \( F(s) \) is then complex analytic over significant regions of the \( s \) plane. For a function of time to be causal, time must be real \( (t \in \mathbb{R}) \), since if it were complex, it would lose the order property (thus it could not be causal). It is helpful to emphasize the causal nature of \( f(t)u(t) \) by indicating the causal nature, using the Heaviside step function \( u(t) \).

Restrictions on a function (e.g., real, causal, periodic, positive real part, etc.) are called a symmetry property. There are many forms of symmetry (Section 1.4.17, p. 103). The concept of symmetry is very general and widely used in both mathematics and physics, where it is more generally known as Group theory. We shall restrict ourselves to only a few very basic symmetries (Section 3.5.1, p. 224).
**Definition of the Laplace transform:** The forward and inverse Laplace transforms are

\[
F(s) = \int_0^\infty f(t)e^{-st}\,dt \quad \quad f(t) = \frac{1}{2\pi j} \int_{\sigma_0-j\infty}^{\sigma_0+j\infty} F(s)e^{st}\,ds
\]

(1.79) \hspace{1cm} (1.80)

When dealing with engineering problems it is convenient to separate the *signals* we use from the *systems* that process them. We do this by treating signals, such as a musical signal, differently from a system, such as a filter. In general signals may start and end at any time. The concept of causality has no mathematical meaning in signal space. Systems, on the other hand, obey very rigid rules (to assure that they remain physical). These physical restrictions are described in terms of the *Network Postulates*, which are first discussed in Sect. 1.4.17, and then in greater detail in Sect. 3.5.1.

**Example:** Here \(s = \sigma + j\omega\) is the complex Laplace frequency in [radians] and \(t\) is time in [seconds].

As discussed in Section 1.5.8 (p. 124), we must use the Cauchy Residue Theorem (CRT), requiring closure of the contour \(C\) at \(\omega_j \rightarrow \pm j\infty\)

\[
\oint_C = \int_{\sigma_0-j\infty}^{\sigma_0+j\infty} + \int_{\infty}
\]

where the path represented by \(\subset_{\infty}\) is a semicircle of infinite radius. For a causal, ‘stable’ (e.g. doesn’t “blow up” in time) signal, all of the poles of \(F(s)\) *must* be inside of the Laplace contour, in the left-half \(s\)-plane.

![Diagram of a mechanical resonant circuit](image)

**Figure 1.24:** *This three element mechanical resonant circuit consisting of a spring, mass and dash-pot (e.g., viscous fluid).*

Hooke’s law for a spring states that the force \(f(t)\) is proportional to the displacement \(x(t)\), i.e., \(f(t) = Kx(t)\). The formula for a dash-pot is \(f(t) = Rv(t)\), and Newton’s famous formula for mass is \(f(t) = d[Mv(t)]/dt\), which for constant \(M\) is \(f(t) = Mdv/dt\).

The equation of motion for the mechanical oscillator in Fig. 1.24 is given by Newton’s second law; the sum of the forces must balance to zero

\[
M \frac{d^2}{dt^2}x(t) + R \frac{d}{dt}x(t) + Kx(t) = f(t).
\]

(1.81)

These three constants, the mass \(M\), resistance \(R\) and stiffness \(K\) are all real and positive. The dynamical variables are the driving force \(f(t) \leftrightarrow F(s)\), the position of the mass \(x(t) \leftrightarrow X(s)\) and its velocity \(v(t) \leftrightarrow V(s)\), with \(v(t) = dx(t)/dt \leftrightarrow V(s) = sX(s)\).

Newton’s second law (c1650) is the mechanical equivalent of Kirchhoff’s (c1850) voltage law (KCL), which states that the sum of the voltages around a loop must be zero. The gradient of the voltage results in a force on a charge (i.e., \(F = qE\)).
Equation 1.81 may be re-expressed in terms of impedances, the ratio of the force to velocity, once it is transformed into the Laplace frequency domain.

**Example:** The divergent series

\[ e^t u(t) = \sum_{n=0}^{\infty} t^n \leftrightarrow \frac{1}{s-1} \]

is a valid description of \( e^t u(t) \), with an unstable pole at \( s = 1 \). For values of \( |x - x_o| < 1 \), the analytic function \( P(x) \) has a region of convergence (RoC). For cases where the argument is complex \( (x \in \mathbb{C}) \), this is called the radius of convergence (RoC). We call the region \( |x - x_o| > 1 \) the region of divergence (RoD), and \( |x - x_o| = 0 \), the singular circle. Typically the underlying function \( P(s) \), defined by the series, has a pole on the singular circle for \( s \in \mathbb{C} \). Once may isolate such poles by moving the expansion point \( s_o \) until the RoC approaches zero.

This is in conflict with the example of \( e^{at} u(t) \), which has a divergent series, that agrees with the unstable function. I’m not sure how to justify this conflict, other than to point out that \( t \in \mathbb{R} \), thus the series expansion of the diverging exponential is real-analytic, not complex analytic. It does have a Laplace transform, with a pole at \( s = 1 \), in agreement with its unstable nature. **Second**, the analytic function must be single valued. This follows from the fact that each term in Eq. 1.40 is single valued. **Third**, analytic functions are very “smooth,” since the may be differentiated an \( \infty \) number of times, and the series still converges. There can be no jumps or kinks in such functions.

The key idea that every impedance must be complex analytic and \( \geq 0 \) for \( \sigma > 0 \) was first proposed by Otto Brune in his PhD at MIT, supervised by a student of Arnold Sommerfeld, Ernst Guilliman, an MIT ECE professor, who played a major role in the development of circuit theory. Brune’s primary (non-MIT) advisor was Cauer, who was also well trained in 19th century German mathematics.\(^{62}\)

**Summary:** While the definitions of the FT (\( \mathcal{F}T \)) and LT (\( \mathcal{L}T \)) transforms appear superficially similar, they are not. The key difference is that the time response of the Laplace transform is causal, leading to a complex analytic frequency responses. The frequency response of the Fourier transform is real, thus typically not analytic. These are not superficial differences. The concept of symmetry is helpful in understanding the many different types of time-frequency transforms. Two fundamental types of symmetry are causality and periodicity.

The Fourier transform \( \mathcal{F}T \) characterizes the steady-state response while the Laplace transform \( \mathcal{L}T \) characterizes both the transient and steady-state response. Given a system response \( H(s) \leftrightarrow h(t) \) with input \( x(t) \), the output is

\[ y(t) = h(t) \ast x(t) \leftrightarrow Y(\omega) = H(s) \bigg|_{s=j\omega} X(\omega). \]

### 1.4.17 Lec 21: Ten network postulates

Solutions of differential equations, such as the wave equation, are conveniently described in terms of mathematical properties, which we present here in terms of 10 **network postulates**:

(P1) **causality** (non-causal/acausal): Causal systems respond when acted upon. Virtually all physical systems obey causality. An example of a causal system is an integrator, which has a

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\(^{62}\)It must be noted that Prof. ‘Mac’ Van Valkenburg from the University of IL., was arguably more influential in circuit theory, during the same period. Mac’s book are certainly more accessible, but perhaps less widely cited.
response of a step function. Filters are also examples of causal systems. Signals represent acausal responses. They do not have a clear beginning or end, such as the sound of the wind or traffic noise.

(P2) linearity (nonlinear): Linear systems obey superposition. If two signals $x(t)$ and $y(t)$ are the inputs to a linear system, producing outputs $x'(t)$ and $y'(t)$, then if the inputs are presented together as $ax(t) + by(t)$, with weights $a, b \in \mathbb{R}$, then the output will be $ax'(t) + by'(t)$. If either $a, b$ are zero, that signal is removed from the output. Nonlinear system mix two inputs, thereby producing other signals not present in the input. For example, if the inputs to a nonlinear system are two sine waves, the output will contain distortion components, having frequencies not present at the input. An example of a nonlinear system is one that multiplies the two inputs. A second is a diode, which rectifies a signal, letting current flow only in one direction. Most physical systems have some degree of nonlinear response, but this is not always desired. Other systems are designed to be nonlinear, such as the diode example.

(P3) passive (active): An active system has a power source, such as a battery while a passive system has no power source. While you may consider a transistor amplifier to be active, it is only so when connected to a power source.

(P4) real (complex) time response: Typically systems are “real in, real out.” They do not naturally have complex responses (real and imaginary parts). While a Fourier transform takes real inputs and produces complex outputs, this is not an example of a complex time response. P4 is a characterization of the input signal, not its Fourier transform.

(P5) time-invariant (time varying): For a system to be a time varying system its properties must depend on when the signal starts and stops. If the output, relative to the input, is independent of the starting time, then the system is time-invariant (static).

(P6) reciprocal (non- or anti-reciprocal): In many ways this is the most difficult propriety to understand. It is characterized by the ABCD matrix. If $B = C$ the system is said to be reciprocal. If $B = -C$ it is said to be anti-reciprocal. The impedance matrix is reciprocal while a loudspeaker is anti-reciprocal, thus modeled by the gyrator rather than a transformer. All non-reciprocal systems are modeled by such gyrator, which swap the force and flow variables.

(P7) reversibility (non-reversible): If the system can be flipped, between input and output, the it is said to be reversible. In terms of the ABCD parameters, if $A = D$ is is reversible.

(P8) space-invariant (space-variant): If a system operates independently as a function of where it physically is in space, then it is space-invariant. When the parameters that characterize the system depend on position, it is space-variant.

(P9) Deterministic (random): Predictability depends on the light cone. Given the wave equation, along with the boundary conditions, the solution may be deterministic, or not, depending on the extent of the “light cone.” Consider the wave with a wave propagating out into uncharted territory. When the wave hits an object, the reflection will return a wave that might considered not be predicted. This is an example of the boundary condition being the radiation impedance, which is not known in advance. One must do the experiment to do solve the
problem, and the radiation impedance is not known in advance. To determine it, one must do the experiment. In this example, that determination could take light years. Thus is, I believe, an example of a “random” system. But the system is deterministic only if the radiation impedance is known accurately, in advance. To define the impedance requires that it be time invariant (P5). One cannot look beyond the light-cone, which depends on the system to be time invariant.

(P10) quasi-static (aka, uni-modal assumption): Quasi-statics follows from systems that have dimensions that are small compared to the local wavelength. This is a very general assumption, that must be false when the frequency is raised and the wavelength becomes short. Thus this is also known as the long-wavelength approximation. It is a very powerful tool, frequently used in modeling 1 dimensional systems, such as transmission lines, horns and one-dimensional systems. Quasi-statics is normally stated as \( \lambda < ka \), where \( k = \frac{c_o}{f} \) and \( a \) is the minimum dimension of the system.

In acoustics this is restrictive because the speed of sound is only \( c_o = 345 \) [m/s], as compared to the speed of light which is \( c_o = 0.3 \times 10^8 \) [m/s]. Given the wave speed the wavelength is \( \lambda = \frac{c_o}{f} \), we see that the wavelength is directly proportional to the wave speed, and inversely proportional to frequency.

Postulate P10 is closely related to the Feynman (1970c, Ch. 12-7) titled The “underlying unity” of nature, well worth a read once you have understood the meaning of quasi-statics. He asks the question “Why do we need to treat the fields as smooth?” My answer is so that we may model them as analytic functions. In some ways this is a mathematical approximation which critically depends on quasi-statics, which in turn depends on the frequency being low enough that the wavelength is much greater than the distance between molecules, air particles, electrons, etc.

This raises the question “Are Maxwell’s equations a band-limited approximation to reality? Personally, I have no idea what the answer is.

Summary discussion of the 10 network postulates: Each postulate has two (or more) categories. For example for (P2) a system is either linear or non-linear and for (P1) is either causal, non-causal or acausal. P6 and P9 only apply to 2-port algebraic networks (those having an input and an output). The others can apply to both a 2– or 1–port networks (e.g., an impedance is a 1-port). An excellent example is the transmission matrix of a loudspeaker shown in Fig. 3.7 (p. 224).

Related forms of these postulates had been circulating in the literature for many years, widely accepted in the network theory literature (Van Valkenburg, 1964a,b; Ramo et al., 1965). The first six of these were formally introduced Carlin and Giordano (1964). Three more (P7-P9) were added by Kim et al. (2016). As noted by Carlin and Giordano (1964, p. 5), while linearity (P2), passivity (P3), realness (P4) and time-invariance (P5) are independent, causality (P1) is a consequence of linearity (P2) and passivity (P3), are not.

1.4.18 Lec 22: Exam II (Evening Exam)
1.5 Stream 3a: Scalar (i.e., Ordinary) Differential Equations

Stream 3 is $\infty$, a concept which typically means unbounded (immeasurably large), but in the case of calculus, $\infty$ means *infinitesimal* (immeasurably small), since taking a limit requires small numbers. Taking a limit means you may never reaching the target, a concept that the Greeks called *Zeno’s paradox* (Stillwell, 2010, p. 76).

When speaking of the class of *ordinary* (versus *vector*) differential equations, the term *scalar* is preferable, since the term “ordinary” is vague, if not a meaningless label. There are a special subset of fundamental theorems for scalar calculus, all of which are about integration, as summarized in Table 1.5 (p. 168), starting with Leibniz’s theorem. These will be discussed below, and more extensively in Lec. 1.5.1 (p. 108) and Chapters 4 (p. 231) and 5 (p. 231).

Following the integral theorems on scalar calculus, are those on vector calculus, without which there could be no understanding of Maxwell’s equations. Of these, the *fundamental theorem of complex calculus* (aka, Helmholtz decomposition), Gauss’s law and Stokes theorem, form the cornerstone of modern vector field analysis. These theorems allow one to connect the differential (point) and macroscopic (integral) relationships. For example, Maxwell’s equations may be written as either vector differential equations, as shown by Heaviside (along with Gibbs and Hertz), or in integral form. It is helpful to place these two forms side-by-side, to fully appreciate their significance. To understand the differential (microscopic) view, one must understand the integral (macroscopic) view). These are presented in Sections 1.6.11 (p. 163) and Fig. 1.37 (p. 167).

The beginning of modern mathematics

Mathematics as we know it today, began in the 16th to 18th centuries, arguably starting with Galileo, Descartes, Fermat, Newton, the Bernoulli family, and Euler, outlined in Fig. 1.25. Galileo was formidable, due to his fame, fortune, and his “successful” stance against the powerful Catholic establishment. His creativity in scientific circles was certainly well known due to his many skills and accomplishments. Descartes and Fermat were at the forefront of merging algebra and geometry. While Fermat kept meticulous notebooks, he did not publish, and tended to be secretive. Thus Descartes’ contributions were more widely acknowledged, but not necessarily deeper.

Regarding the development of calculus, much was yet to be done by Newton and Leibniz, using term by term integration of functions based on Taylor series representation. This was a powerful technique, but as stated earlier, incomplete because the Taylor series can only represent single-valued functions, within the RoC. But more importantly, Newton (and others) failed to recognize the powerful generalization to complex analytic functions. The first major breakthrough was Newton’s publication of Principia (1687), and a second was Riemann (1851), advised by Gauss.

Following Newton’s lead, the secretive and introverted behavior of the typical mathematician dramatically changed with the Bernoulli family (Fig. 1.25). The oldest brother, Jacob, taught his much younger brother Johann, who then taught his son Daniel. But Johann’s star pupil was Euler. Euler first mastered all the tools and then published, with a prolific voracity previously unknown.

**Euler and the circular functions:** The first major task was to understand the family of analytic circular functions, $e^x$, $\sin(x)$, $\cos(x)$, and $\log(x)$, a task begun by the Bernoulli family, but mastered by Euler. Euler sought relations between these many functions, some of which may not be thought

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1.5. STREAM 3A: SCALAR CALCULUS (11 LECTURES)

Chronological history post 16\textsuperscript{th} century

\textit{16\textsuperscript{th}} Bombelli 1526-1572

\textit{17\textsuperscript{th}} Galileo 1564-1642, Kepler 1571-1630, Newton 1642-1727 Principia 1687; Mersenne; Huygen; Pascal; Fermat, Descartes (analytic geometry); Bernoullis Jakob, Johann & son Daniel

\textit{18\textsuperscript{th}} Euler 1707-1783 Student of Johann Bernoulli; d’Alembert 1717-1783; Kirchhoff; Lagrange; Laplace; Gauss 1777-1855

\textit{19\textsuperscript{th}} Möbius, Riemann 1826-1866, Cauchy 1789-1857, Helmoltz 1821-1894, Maxwell 1831-1879, Heaviside 1850-1925, Rayleigh 1842-1919

\textit{20\textsuperscript{th}} Hilbert; Einstein; . . .

\textbf{Time-Line}

\begin{center}
<table>
<thead>
<tr>
<th>1525</th>
<th>1600</th>
<th>1700</th>
<th>1800</th>
<th>1875</th>
<th>1925</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bombelli</td>
<td>Descartes</td>
<td>Newton</td>
<td>Mersenne</td>
<td>Newton</td>
<td>Gauss</td>
</tr>
<tr>
<td>Huygen</td>
<td>Johann Bernoulli</td>
<td>Daniel Bernoulli</td>
<td>Euler</td>
<td>Gauss</td>
<td>Maxwell</td>
</tr>
<tr>
<td>Galileo</td>
<td>Fermat</td>
<td>d’Alembert</td>
<td>Helmholtz</td>
<td>Rayleigh</td>
<td>Hilbert</td>
</tr>
<tr>
<td>Mozart</td>
<td>US Civil War</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
\end{center}

Figure 1.25: Time-line of the four centuries from the 16\textsuperscript{th} and 20\textsuperscript{th} CE. Given the time line, it is likely that Bombelli’s discovery of Diophantus’ book on “Arithmetic” in the Vatican library, triggered many of the ideas presented by Descartes and Fermat, followed by others (i.e., Newton). Bombelli’s discovery might be considered as a magic moment in mathematics. The vertical red lines connect mentor-student relationships. Figure 1.15 (p. 82) for a closer look at the 15-18c, and Fig. 1.13 (p.58) for a closer look at the 18-20c. \[\text{fig:TimeLineCE}\]

of as being related, such as the log and sin functions. The connection that may “easily” be made is through their complex Taylor series representation (Eq. 1.41, p. 69). By the manipulation of the analytic series representations, the relationship between $e^x$, and the sin($x$) and cos($x$) was precisely captured with the relation

$$e^{j\omega} = \cos(\omega) + j\sin(\omega),$$

and its analytic inverse (Greenberg, 1988, p. 1135)

$$\tan^{-1}(z) = \frac{1}{2j} \ln \left(\frac{1-jz}{1+jz}\right).$$

Exercise: Find the inverse of Eq. 1.82. Solution: From the ratio of the real and imaginary parts of Eq. 1.82, and call it $z(\omega)$

$$z(\omega) = \tan(\omega) = -j \frac{e^{j\omega} - e^{-j\omega}}{e^{j\omega} + e^{-j\omega}} = -j \frac{e^{2j\omega} - 1}{e^{2j\omega} + 1}.$$  \hspace{1cm} (1.84)

Solving for $e^{2j\omega}$

$$e^{2j\omega} = \frac{1 + zj}{1 - zj}.$$
1.5.1 Lec 23: Fundamental theorems of calculus

History of the fundamental theorem of calculus:  It some sense, the story of calculus begins with the fundamental theorem of calculus (FTC), also known generically as Leibniz’s formula. The simplest integral is the length of a line \( L = \int_0^L dx \). If we label a point on a line as \( x = 0 \) and wish to measure the distance to any other point \( x \), we form the line integral between the two points. If the line is straight, this integral is simply the Euclidean length given by the difference between the two ends (Eq. 1.4.8, p. 83).

If \( F(x) \) describes a height above the line, then the area under \( F(x) \) is

\[
 f(x) - f(0) = \int_{x=0}^{x} F(\chi) d\chi, \tag{1.85}
\]

where \( \chi \) is a dummy variable of integration. Thus the area under \( F(x) \) \( (F(x), x \in \mathbb{R}) \) only depends on the difference of the area evaluated at the end points. This makes intuitive sense, by viewing \( f(x) \) as the area of a graph of \( F(x) \).
1.5. STREAM 3A: SCALAR CALCULUS (11 LECTURES)

This property, of the area as an integral over an interval, only depending on the end points, has important consequences in physics in terms of conservation of energy, making generalizations important.

Fundamental theorems of real calculus:

If \( F(x) \) is analytic (Eq. 1.40, p. 69), then

\[
F(x) = \frac{d}{dx} f(x),
\]

which is known as the Fundamental theorem of (real) calculus (FTC). Thus Eq. 1.86 may be viewed as an anti-derivative, or exact differential. This is easily shown by evaluating

\[
\frac{d}{dx} f(x) = \lim_{\delta \to 0} \frac{f(x + \delta) - f(x)}{\delta} = F(x),
\]

starting from Eq. 1.85. If \( F(x) \) is not analytic then the limit may not exist, so this seems like a necessary condition. There are many important variations on this very basic theorem (see Sect. 1.5). For example, the limits could depend on time. Also when taking Fourier and Laplace transforms, the integrand depends on both time and frequency via an exponential “kernel” function \( e^{\pm st} \).

Case of a complex path

The Fundamental theorem of complex calculus (FTCC) states that for any complex analytic function \( F(s) \in \mathbb{C} \) with \( s = \sigma + \omega j \in \mathbb{C} \) (Greenberg, 1988, p. 1197)

\[
f(s) - f(s_o) = \int_{s_o}^{s} F(\zeta) d\zeta.
\]

If we compare this to Eq. 1.86, they differ in that the path of the integral is complex, meaning that the integral is over \( s \in \mathbb{C} \), rather than a real integral over \( x \in \mathbb{R} \). The Fundamental theorems of complex calculus (FTCC) states that the integral only depends on the end points, since

\[
F(s) = \frac{d}{ds} f(s).
\]

Comparing Eq. 1.86 (FTC) with Eq. 1.88 (FTCC), it would appear that this can only be true if \( F(s) \in \mathbb{C} \) is complex analytic, meaning it must have a Taylor series in powers of \( s \in \mathbb{C} \).

Complex analytic functions:

The definition of a complex analytic function \( F(s) \) of \( s \in \mathbb{C} \) is that the function may be expanded in a Taylor series (Eq. 1.47, p. 73) about an expansion point \( s_o \in \mathbb{C} \). This definition follows the same logic as the FTC. Thus we need a definition for the coefficients \( c_n \in \mathbb{C} \), which most naturally follow from Taylor’s formula

\[
c_n = \frac{1}{n!} \frac{d^n}{ds^n} F(s) \bigg|_{s=s_o}.
\]

The requirement that \( F(s) \) have a Taylor series naturally follows by taking derivatives with respect to \( s \) at \( s_o \). The problem is that both integration and differentiation of functions of complex Laplace frequency \( s = \sigma + \omega j \) have not yet been defined.

Thus the question is: “What does it mean to take the derivative of a function \( F(s) \in \mathbb{C}, s = \sigma + \omega j \in \mathbb{C} \), with respect to \( s \), where \( s \) defines a plane rather than a real line?” We learned
how to form the derivative on the real line. Can the same derivative concept be extended to the complex plane?

The answer is affirmative. The question may be resolved by applying the rules of the real derivative when defining the derivative in the complex plane. However for the complex case, there is an issue, regarding direction. Given any analytic function \( F(s) \), is the partial derivative with respect to \( \sigma \) different from the partial derivative with respect to \( \omega \)? For complex analytic functions, the FTCC states that the integral is independent of the path in the \( s \) plane. Based on the chain rule, the derivative must also be independent of direction at \( s_0 \). This directly follows from the FTCC. If the integral of a function of a complex variable is to be independent of the path, the derivative of a function with respect to a complex variable must be independent of the direction. This follows from Taylor’s formula, Eq. 1.89 for the coefficients of the complex analytic formula. The FTC defines the area as an integral over a real differential \( (dx \in \mathbb{R}) \), while the FTCC relates an integral over a complex function \( F(s) \in \mathbb{C} \), along a complex interval (i.e., path) \( (ds \in \mathbb{C}) \). For the FTC the area under the curve only depends on the end points of the anti-derivative \( f(x) \). But what is the meaning of an “area” along a complex path? The Cauchy-Riemann conditions provide the answer.

1.5.2 Lec 24: Cauchy-Riemann conditions

For the integral of \( Z(s) = R(\sigma, \omega) + X(\sigma, \omega)j \) to be independent of the path, the derivative of \( Z(s) \) must be independent of the direction of the derivative. As we show next, this leads to a pair of equations known as the Cauchy-Riemann conditions. This is an important generalization of Eq. 1.1, p. 18 which goes from real integration \( (x \in \mathbb{R}) \) to complex integration \( (s \in \mathbb{C}) \), based on lengths, thus on area.

To define

\[
\frac{d}{ds} Z(s) = \frac{d}{ds} [R(\sigma, \omega) + jX(\sigma, \omega)]
\]

take partial derivatives of \( Z(s) \) with respect to \( \sigma \) and \( j\omega \), and equate them:

\[
\frac{\partial Z}{\partial \sigma} = \frac{\partial R}{\partial \sigma} + j \frac{\partial X}{\partial \sigma} \quad \equiv \quad \frac{\partial Z}{j\partial \omega} = \frac{\partial R}{\partial j\omega} + j \frac{\partial X}{\partial j\omega}.
\]

This says that a horizontal derivative, with respect to \( \sigma \), is equivalent to a vertical derivative, with respect to \( j\omega \). Taking the real and imaginary parts gives the Cauchy Riemann conditions

\[
\text{CR-1: } \frac{\partial R(\sigma, \omega)}{\partial \sigma} = j \frac{\partial X(\sigma, \omega)}{\partial \omega} \quad \text{and} \quad \text{CR-2: } \frac{\partial R(\sigma, \omega)}{\partial \omega} = -j \frac{\partial X(\sigma, \omega)}{\partial \sigma}.
\]

(1.90)

The \( j \) cancels in CR-1, but introduces a \( j^2 = -1 \) in CR-2. These equations are known as the Cauchy-Riemann (CR) conditions. They may also be written in polar coordinates \( (s = re^{\theta}) \) as

\[
\frac{\partial R}{\partial r} = \frac{1}{r} \frac{\partial X}{\partial \theta} \quad \text{and} \quad \frac{\partial X}{\partial r} = -\frac{1}{r} \frac{\partial R}{\partial \theta}.
\]

If you are wondering what would happen if we took a derivative at 45 degrees, then we only need to multiply the function by \( e^{\pi/4} \). But that will not change the derivative. Thus we may take the derivative in any direction by multiplying by \( e^{\theta} \), and the CR conditions will not change.

The CR conditions are necessary conditions that the integral of \( Z(s) \), and thus its derivative, be independent of the path, expressed in terms of conditions on the real and imaginary parts of \( Z \).
This is a very strong condition on \( Z(s) \), which follows assuming that \( Z(s) \) may be approximated by a Taylor series in \( s \)

\[
Z(s) = Z_0 + Z_1 s + \frac{1}{2} Z_2 s^2 + \cdots ,
\]  

(1.91)

where \( Z_n \in \mathbb{C} \) are complex constants given by the Taylor series formula (Eq. 1.89, p. 109).

**Complex analytic functions:** Every function that may be expressed as a Taylor series about a point \( s \) is said to be complex analytic at that point. This series, which is single valued, is said to converge within a radius of convergence (RoC). This highly restrictive condition has significant physical consequences. As an important example, every impedance function \( Z(s) \) obeys the CR conditions over large regions of the \( s \) plane, including the entire right half plane (RHP) \( (\sigma > 0) \). This condition is summarized by the Brune condition \( \Re\{Z(\sigma > 0)\} \geq 0 \) (Section 1.5.3, Eq. 1.99, p. 114). When the CR condition is generalized to volume integrals, it is called Green’s theorem, used heavily in the solution of boundary value problems in engineering and physics. (Kusse and Westwig, 2010) The last chapter of this course shall depend heavily on these concepts.

We may merge these equations into a pair of second order equations by taking a second round of partials. Specifically, eliminating the real part \( R(\sigma,\omega) \) of Eq. 1.90 gives

\[
\frac{\partial^2 R(\sigma,\omega)}{\partial\sigma\partial\omega} = \frac{\partial^2 X(\sigma,\omega)}{\partial^2 \omega} = -\frac{\partial^2 X(\sigma,\omega)}{\partial^2 \sigma} ,
\]  

(1.92)

which may be compactly written as \( \nabla^2 X(\sigma,\omega) = 0 \). Eliminating the imaginary part gives

\[
\frac{\partial^2 X(\sigma,\omega)}{\partial\omega\partial\sigma} = \frac{\partial^2 R(\sigma,\omega)}{\partial^2 \sigma} = -\frac{\partial^2 R(\sigma,\omega)}{\partial^2 \omega} ,
\]  

(1.93)

which may be written as \( \nabla^2 R(\sigma,\omega) = 0 \).

In summary, for a function \( Z(s) \) to be complex analytic, the derivative \( dZ/ds \) must be independent of direction (path), which requires that the real and imaginary parts of the function obey Laplace’s equation, i.e.,

\[
\text{CR-3: } \nabla^2 R(\sigma,\omega) = 0 \quad \text{and} \quad \text{CR-4: } \nabla^2 X(\sigma,\omega) = 0 .
\]  

(1.94)

The CR equations are easier to work with because they are first order, but the physical intuition is best understood by noting fact 1) the derivative of a complex analytic function is independent of its direction, and fact 2) the real and imaginary parts of the function each obey Laplace’s equation. Such relationships are known as harmonic functions.\(^{64}\)

As we shall see in the next few lectures, complex analytic functions must be smooth since every analytic function may be differentiated an infinite number of times, within the RoC. The magnitude must attain its maximum and minimum on the boundary. For example, when you stretch a rubber sheet over a jagged frame, the height of the rubber sheet obeys Laplace’s equation. Nowhere can the height of the sheet rise above or below its value at the boundary.

Harmonic functions define Conservative fields, which means that energy (like a volume or area) is conserved. The work done in moving a mass from \( a \) to \( b \) in such a field is conserved. If you return the mass from \( b \) back to \( a \), the energy is retrieved, and zero net work has been done.

\(^{64}\)When the function is the ratio of two polynomials, as in the cases of the Brune impedance, they are also related to Möbius transformations, also know as bi-harmonic operators.
Lec 25: Complex Analytic functions and Brune Impedance

It is rarely stated that the variable that we are integrating over, either $x$ (space) or $t$ (time), is real ($x, t \in \mathbb{R}$), since that fact is implicit, due to the physical nature of the formulation of the integral. But this intuition must be refined once complex numbers are included with $s \in \mathbb{C}$, where $s = \sigma + \omega j$.

The fact that time and space are real variables is more than an assumption, rather it is a requirement, due to the order property. Real numbers have order. For example, if $t = 0$ is now (the present), then $t < 0$ is the past and $t > 0$ is the future. The order property of time and space allows one to order these along a real axis. If the axis were complex, as in frequency $s$, the order property is invalid. It follows that if we require order, time and space must be real ($t, x \in \mathbb{R}$). Interestingly, it was shown by d’Alembert (1747) that time and space are related by the pure delay due to the wave speed $c_0$. To obtain a solution to the governing wave equation, that d’Alembert first proposed for sound waves, $x, t \in \mathbb{R}^3$ may be combined as functions of

$$\zeta_{\pm} = t \pm x/c_0,$$

where $c [\text{m/s}]$ is the phase velocity of the waves. The d’Alembert solution to the wave equation, describing waves on a string under tension, is

$$u(x, t) = F(t - x/c_0) + G(t + x/c_0),$$

(1.95)

which describes the transverse velocity (or displacement) of two independent waves $F(\zeta_-), G(\zeta_+)$ on the string, which represent forward and backward traveling waves. For example, starting with a string at rest, if one displaces the left end, at $x = 0$, by a step function $u(t)$, then that step displacement will propagate to the right as $u(t - x/c_0)$, arriving at location $x_o [\text{m}]$, at time $x_o/c_0 [\text{s}]$. Before this time, the string will not move to the right of the wave-front, at $x_o [\text{m}]$, and after $t_o [\text{s}]$ it will have displacement 1. Since the wave equation obeys superposition (postulate P1, p. 103), it follows that the “plane-wave” eigen-functions of the wave equation for $x, \mathbf{k} \in \mathbb{R}^3$ are given by

$$\psi_{\pm}(\mathbf{x}, t) = \delta(t \mp \mathbf{k} \cdot \mathbf{x}) \leftrightarrow e^{\pm \mathbf{k} \cdot \mathbf{x}}$$

(1.96)

where $|\mathbf{k}| = 2\pi/|\lambda| = \omega/c_0$ is the wave number, $|\lambda|$ is the wavelength, and $s = \sigma + \omega j$.

When propagation losses are considered, we must replace $\mathbf{j}\mathbf{k}$ with a complex analytic wave number $\kappa(s) = k_r(s) + jk_i(s)$, which denoted as either the complex propagation function, propagation function, or the dispersion relation. Forms of loss include viscosity and radiation. Important examples include acoustic plane waves, electromagnetic wave propagation, antenna theory, and the more complex case of 3D electron wave propagating in crystals (i.e., silicon), where electrons and EM waves are in a state of equilibrium. Perhaps electrons and photons are simply different EM states, where $\kappa(x, s)$ describes the crystal’s dispersion relations as functions of both frequency and directions, famously known as Brillouin zones. Dispersion is a property of the medium such that the wave velocity is a function of frequency and possibly direction, as in silicon. Several highly informative discussions on the history of this topic may be found in Brillouin (1953).

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65 D’Alembert’s solution is valid beyond analytic functions, such as $F(\zeta) = \delta(\zeta)$.

66 In case you missed it, I’m suggesting is that photons and electrons are the same thing, in different propagation “states.” The difference is the medium, which determines the dispersion relation.
Complex impedance functions

Conservation of energy (or power) is a cornerstone of modern physics. It may have first been under consideration by Galileo Galilei (1564-1642). Today the question is not if it is true, the questions is, why. Specifically, what is the physics behind conservation of energy? Surprisingly, the answer is straightforward, based on its definition, and the properties of impedance. Recall that the power is the product of the force and flow, and impedance is their ratio.

The power is given by the product of two variables, sometimes called conjugate variables, the force and the flow. In electrical terms, these are voltage (force) \( v(t) \leftrightarrow V(\omega) \) and current (flow) \( i(t) \leftrightarrow I(\omega) \), thus the electrical power at any instant of time is

\[
P(t) = v(t)i(t). \tag{1.97}
\]

The total energy \( w(t) \) is the integral of the power, since \( P(t) = dw/dt \). Thus if we start with all the elements at rest (no currents and no voltages, then the energy as a function of time is always positive

\[
w(t) = \int_0^t P(t)dt \geq 0, \tag{1.98}
\]

and is simply the total energy applied to the network (Van Valkenburg, 1964a, p. 376). Since the voltage and current are related by either an impedance or an admittance, conservation of energy depends on the property of impedance. From Ohm’s law and P1 (every impedance is causal)

\[
v(t) = z(t) \star i(t) = \int_{\tau=0}^t z(\tau)i(t - \tau)d\tau \leftrightarrow Z(s)I(s)
\]

From the expression for the energy, in terms of Ohm’s law

\[
w(t) = u(t) \star (v(t)i(t))
\]

\[
= u(t) \star i(t) \star z(t) \star i(t)
\]

\[
= \int_{t=0}^t i(t)\left(\int_{\tau=0}^t z(\tau)i(t - \tau)d\tau\right)dt
\]

\[
= \int_{\tau=0}^t z(\tau)\left(\int_{t=0}^t i(t)i(t - \tau)dt\right)d\tau
\]

\[
\leq \int_{\tau=0}^t z(\tau)|w|^2(\tau)]d\tau
\]

\[
\leftrightarrow \frac{1}{s}Z(s)|I(\omega)|^2 \geq 0.
\]

The step from time to frequency follows from the fact that

\[
|w|^2(\tau) = i(t) \star i(t) = \int_{\tau=0}^t i(\tau)i(t - \tau) \rightarrow |I(\tau, \omega)|^2 > 0
\]

always has a positive Fourier transform for every possible \( i(t) \)

**Example:** Let \( i(t) = \delta(t) \). Then \( |w|^2(\tau) = i(t) \star i(t) = \delta(\tau) \). Thus

\[
w(t) = \int_{\tau=0}^t z(\tau)|w|^2(\tau)d\tau = \int_{\tau=0}^t z(\tau)\delta(\tau)d\tau = \int_{t}^t z(\tau)d\tau.
\]

The Brune impedance always has the form \( z(t) = r_o \delta(t) + \zeta(t) \). The surge impedance is defined as

\[
r_o = \int_0^\infty z(t)dt.
\]
The integral of the reactive part \( \zeta(t) \) is always zero, since the reactive part cannot store energy.

Perhaps easier to visualize is when working in the frequency domain where the total energy, equal to the integral of the real part of the power, is

\[
\frac{1}{s} \Re V I = \frac{1}{2s} (V^* I + V I^*) = \frac{1}{2s} (Z^* I^* I + Z I I^*) = \frac{1}{s} \Re Z(s)|I|^2 \geq 0.
\]

Formally this is related to a positive definite operator where the positive resistance forces the definiteness, which is sandwiched between the current.

In conclusion conservation of energy is totally dependent on the properties of the impedance. Thus of the most important and obvious applications of complex functions of a complex variable is the impedance function. This seems to be the ultimate example of the FTC, applied to \( z(t) \), in the name of conservation of energy.

**Poles and zeros of PR functions must be first degree:** We conjecture that this proof also requires that the poles and the zeros of the impedance function be simple (only first degree). Second degree poles would have a reactive “secular” responses of the form \( h(t) = t \sin(\omega_k t + \phi)u(t) \), and these terms would not average to zero, depending on the phase, as is required of an impedance. As a result, only single degree poles would be possible.\(^{67}\) Furthermore, when the impedance is the ratio of two polynomials, where the lower degree polynomial is the derivative of the higher degree one, then the poles and zeros must alternate. This is a well known property of the Brune impedance, that has never been adequately explained except for very special cases (Van Valkenburg, 1964b, p. 104-107). I suspect that no one has ever reported an impedance having second degree poles and zeros as that would be rare impedance. Network analysis books never report 2d degree poles and zeros in their impedance functions. Nor has there ever been any guidance as to where the poles and zeros might lie in the LHP. This would resolve this long-standing mystery.

**Every impedance must obey conservation of energy (P3):** According to Postulate P3 Section 1.4.17 (p. 103), a system is passive if it does not contain a power source. Drawing power from an impedance violates conservation of energy. This propriety is also called *positive real*, which is defined as (Brune, 1931b,a)

\[
\Re \{Z(s \geq 0)\} \geq 0,
\]

namely the real part of every impedance must be non-negative for \( \sigma \geq 0 \). When this condition holds, one cannot draw more power than stored in the impedance. A second condition requires that the impedance has simple poles. If there were a pole in the region \( \sigma > 0 \), then the first condition would not be met. Therefore there can only be simple poles (degree of 1) in the region \( \sigma \leq 0 \). The region \( \sigma \leq 0 \) is called the left half s plane (LHP), and the complementary region, \( \sigma > 0 \), is called the right half s plane (RHP). The condition on the simple poles is sufficient, but not necessary, as \( Z(s) = 1/\sqrt{s} \) is a physical impedance, but is not a first degree pole. The impedance function \( Z(s) = R(\sigma, \omega) + jX(\sigma, \omega) \) has resistance \( R \) and reactance \( X \) as a function of complex frequency \( s = \sigma + j\omega \). The function \( z(t) \leftrightarrow Z(s) \) are defined by a Laplace transform pair. From the causality postulate (P1) of Sections 1.4.17 and 3.5.1 (p. 224), \( z(t < 0) = 0 \).

As an example, a resistor \( R_o \) in series with an capacitor \( C_o \) has an impedance given by (Table F.3, p. 278)

\[
Z(s) = R_o + 1/sC_o \leftrightarrow R_o \delta(t) + \frac{1}{C_o} u(t) = z(t),
\]

\(^{67}\)Secular terms result from second degree poles since \( u(t) \star u(t) = tu(t) \).
with constants $R_o, C_o \in \mathbb{R} > 0$. In mechanics an impedance composed of a dash-pot (damper) and a spring have the same form. A resonant system has an inductor, resistor and a capacitor, with an impedance given by (Table F.4, p. 279)

$$Z(s) = \frac{sC_o}{1 + sC_oR_o + s^2C_oM_o} \leftrightarrow C_o \frac{d}{dt} \left( c_+ e^{s+t} + c_- e^{s-t} \right) = z(t) \quad (1.101)$$

is a second degree polynomial with two complex resonant frequencies $s_{\pm}$. When $R_o > 0$ these roots are in the left half $s$ plane, with $z(t) \leftrightarrow Z(s)$.

Systems (networks) containing many elements, and transmission lines can be much more complicated, yet still have a simple frequency domain representation. This is the key to understanding how these physical systems work, as will be described below.

**Complex analytic functions:** To solve a differential equation, or integrate a function, Newton used the Taylor series to integrate one term at a time. However he only used real functions of a real variable, due to the fundamental lack of appreciation of the complex analytic series. This same method is how one finds solutions to scalar differential equations today, but using an approach that makes the solution method less obvious. Rather than working directly with the Taylor series, today we use the complex exponential, since the complex exponential is an eigen-function of the derivative

$$\frac{d}{dt} e^{st} = se^{st}.$$ 

Since $e^{st}$ may be expressed as a Taylor series, having coefficients $c_n = 1/n!$, in some real sense the modern approach is a compact way of doing what Newton did. Thus every linear constant coefficient differential equation in time may be simply transformed into a polynomial in complex Laplace frequency $s$, by looking for solutions of the form $A(s)e^{st}$, transforming the differential equation into a polynomial $A(s)$ in complex frequency. For example

$$\frac{d}{dt} f(t) + af(t) \leftrightarrow (s + a)F(s).$$

The root of $A(s_r) = s_r + a = 0$ is the eigen-value of the differential equation. A powerful tool for understanding the solutions of differential equations, both scalar and vector, is to work in the Laplace frequency domain. The Taylor series has been replaced by $e^{st}$, transforming Newton’s real Taylor series into the complex exponential eigen-function. In some sense, these are the same method, since

$$e^{st} = \sum_{n=0}^{\infty} \frac{(st)^n}{n!}. \quad (1.102)$$

Taking the derivative with respect to time gives

$$\frac{d}{dt} e^{st} = se^{st} = s \sum_{n=0}^{\infty} \frac{(st)^n}{n!}, \quad (1.103)$$

which is also complex analytic. Thus if the series for $F(s)$ is valid (i.e., it converges), then its derivative is also valid. This was a very powerful concept, exploited by Newton for real functions of a real variable, and later by Cauchy and Riemann for complex functions of a complex variable. The key here is “Where does the series fail to converge?” in which case, the entire representation fails. This is the main message behind the FTCC (Eq. 1.88).
The FTCC (Eq. 1.88) is formally the same as the FTC (Eq. 1.86) (Leibniz formula), the key (and significant) difference being that the argument of the integrand \( s \in \mathbb{C} \). Thus this integration is a line integral in the complex plane. One would naturally assume that the value of the integral depends on the path of integration.

But, according to FTCC, it does not. In fact they are clearly distinguishable from the FTC. And the reasoning is the same. If \( F(s) = df(s)/ds \) is complex analytic (i.e., has a power series \( f(s) = \sum_k c_k s^k \), with \( f(s), c_k, s \in \mathbb{C} \)), then it may be integrated, and the integral does not depend on the path. At first blush, this is sort of amazing. The key is that \( F(s) \) and \( f(s) \) must be complex analytic, which means they are differentiable. This all follows from the Taylor series formula Eq. 1.89 (p. 109) for the coefficients of the complex analytic series. For Eq. 1.88 to hold, the derivatives must be independent of the direction, as discussed in Section 1.5.2. The concept of a complex analytic function therefore has eminent consequences, in the form of several key theorems on complex integration discovered by Cauchy (c1820).

The use of the complex Taylor series formula in the complex plane, generalizes the functions they describe, with unpredictable consequences, as nicely shown by the domain coloring diagrams presented in Section 1.4.14 (p. 98). The tool of complex integration was first exploited in physics by Sommerfeld (1952), to explain the onset transients in waves, as explained in detail in Brillouin (1960, Chap. 3). Up to 1910, when Sommerfeld first published his results using complex analytic signals and saddle point integration in the complex plane, there was a poor understanding of the implications of the causal wave-front. It would be reasonable to say that his insights changed our understanding of wave propagation, for both light and sound. Sadly this insight has not been fully appreciated, even to this day. If you question this summary, please read Brillouin (1960, Chap. 1).

The full power of the complex analytic function was first appreciated by Bernard Riemann (1826-1866), in his University of Göttingen PhD Thesis of 1851, under the tutelage of Carl Friedrich Gauss (1777-1855), and drawing heavily on the work of Cauchy.

The key definition of a complex analytic function is that it has a Taylor series representation over a region of the complex frequency plane \( s = \sigma + j \omega \), that converges in a region of convergence (RoC) about the expansion point, with a radius determined by the nearest pole of the function. A further surprising feature of all analytic functions is that within the RoC, the inverse of that function also has a complex analytic expansion. Thus given \( w(s) \), one may also determine \( s(w) \) to any desired accuracy, critically depending on the RoC.

1.5.4 Lec 26: Multi-valued functions

In the field of mathematics there seems to have been a tug-of-war regarding the basic definition of the concept of function. The accepted definition today seems to be a single-valued mapping from the domain to the codomain (or range). This makes the discussion of multi-valued functions somewhat tedious. In 1851 Riemann (working with Gauss) seems to have resolved this problem for the natural set of multi-valued functions, by introducing the concept of the branch-cut and sheets.

Two simple examples multi-valued functions are the circle \( z^2 = x^2 + y^2 \) and \( w = \log(z) \). For example, assuming \( z \) is the radius of the circle, when solving for \( y(x) \) gives the multi-valued function

\[
y(x) = \pm \sqrt{z^2 - x^2}.
\]

If we accept the modern definition of a function, then \( y(x) \) is not a function, nor even two functions. For example what if \( x > z \)? Or worse, what if \( z = 2j \) with \( |x| < 1 \). Riemann’s solution resolves all these difficulties (as best I know).
1.5. STREAM 3A: SCALAR CALCULUS (11 LECTURES)

To proceed, we need definitions and classifications of the various types of complex singularities:

1. Poles of degree 1 are called simple poles. Their amplitude called the residue (e.g. \( \alpha/s \) has residue \( \alpha \)). Simple poles are special (Eq. 1.106, p. 121) and play a key role in mathematical physics. Consider the function \( y(s) = \sqrt[\alpha]{1-s^\alpha} \) with \( \alpha \in \mathbb{Z}, \mathbb{F}, \mathbb{R} \) and \( \mathbb{C} \).

2. When the numerator and denominator of a rational function have a common root (i.e., factor), that root is said to be removable.

3. A singularity that is not 1) removable, a 2) pole or 3) branch point, is called essential.

4. When the first derivative of a function \( Z(s) \) has a simple pole at \( s_0 \), then \( s_0 \) is said to be a branch point of \( Z(s) \) (e.g., \( d \ln(s^\alpha)/ds = \alpha/s \)). However the converse does not necessarily hold.

5. A complex function which is analytic, except for isolated poles, is called metamorphic.\(^{69}\)

Metamorphic functions can have any number of poles, even an infinite number.

More complex typologies are being researched today, and progress that is expected to accelerate due to modern computing technology.\(^{70}\) It is helpful to identify the physical meaning of these more complex surfaces, to guide us in their interpretation and possible applications.\(^{71}\)

Branch cuts: Up to this point we have only considered poles of degree \( k \), of the form \( 1/s^k \), with \( k \in \mathbb{Z} \). The concept of a branch cut allows one to manipulate (and visualize) multi-valued functions, for which \( k \in \mathbb{F} \). This is done by breaking each region into a single valued sheets. The concepts of the branch cut, the sheets, and the extended plane, were first devised by Riemann, working with Gauss (1777-1855), and first described in his thesis of 1851. Of course it was these three mathematical and geometrical constructions that provide the deep insight to complex analytic functions, supplementing the important earlier work of Cauchy (1789-1857) on the calculus of complex analytic functions. For an intuitive discussion of Riemann sheets and branch cuts, see Boas (1987, Section 29, pp. 221-225).

Square root function: The branch cut is a line that separates the various single valued parts of a multi-valued function. For example, in Fig. 1.27 we see the single valued function \( w(z) = s^2 \) (left), and on the right, its inverse, the double-valued mapping of \( s(w) = \pm \sqrt{w} \).

The multi-valued nature of the square root is best understood by working with the function in polar coordinates. Let

\[
s_k = re^{i\theta}e^{2\pi jk},
\]

where \( k \) is the sheet-index, and

\[
w = \rho e^{i\psi} = \sqrt{r}e^{i\theta/2}e^{\pi jk},
\]

where \( r, \rho \in \mathbb{R} \) are the magnitudes and \( \theta, \psi \in \mathbb{R} \) are the angles. The domain-coloring program \texttt{zviz.m} assumes that the angles go from \(-\pi < \theta < \pi\), with \( \theta = 0 \) being a light red and \( \pm\pi \) a blue color. This angle to color map is shown in the left panel of Fig. 1.29. The first Riemann sheet \((k = 0)\) of \( s_k \) is defined for \(-\pi < \theta < \pi\).

\(^{68}\)https://en.wikipedia.org/wiki/Pole_(complex_analysis)
\(^{69}\)https://en.wikipedia.org/wiki/Meromorphic_function
\(^{70}\)https://www.maths.ox.ac.uk/about-us/departmental-art/theory
\(^{71}\)https://www.quantamagazine.org/secret-link-uncovered-between-pure-math-and-physics-20171201
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Figure 1.27: Here we see the mapping for the square root function \( w(z) = \pm \sqrt{z} \) which has two single-valued sheets, corresponding to the two signs of the square root. The plot at the bottom represents the \( z \) plane while the two interleaved plots on top, the two sheets of \( \pm \sqrt{z} \). The branch cut runs from \( z = 0 \) along the real axis out to \( \infty \). The location of the branch cut may be moved by rotating the \( z \) coordinate system. For example, \( w(z) = \pm j \sqrt{z} \) and \( w(z) = \pm \sqrt{-z} \) have a different branch cuts, as may be easily verified using the Matlab/Octave commands \( j*zviz(z) \) and \( zviz(-z) \), as shown in Fig. 1.29. Every function is analytic on the branch cut, since the cut may be moved. If a Taylor series is formed on the branch cut, it will describe the function on the two different sheets. Thus the complex analytic series (i.e., the Taylor formula, Eq. 1.89) does not depend on the location of a branch cut, as it only describes the function uniquely (as a single valued function), valid in its local region of convergence. This figure has been taken from Stillwell (2010, p. 303). A more comprehensive view presented in the text. The branch cut lies in the domain \( x \in \mathbb{C} \), not in the codomain \( w(x) \in \mathbb{C} \) (Kusse and Westwig, 2010). This becomes clear by studying how \( zviz.m \) works, and understanding its output. A example that shows this is Fig. 1.29, where the axes are \( s \) and \( \sigma \omega \), with the branch cut along the negative \( \sigma \) axis (\( \theta = \pi \)).

The second sheet \((k = 1)\) picks up at \( \theta = \pi \) and continues on to \( \pi + 2\pi = 3\pi \). The first sheet maps the angle of \( w \) (i.e., \( \phi = \angle w = \theta/2 \)) from \(-\pi/2 < \phi < \pi/2 \) (\( w = \sqrt{r}e^{i\theta/2} \)). This corresponds to \( u = \Re\{w(s)\} > 0 \). The second sheet maps \( \pi/2 < \psi < 3\pi/2 \) (i.e., \( 90^\circ \) to \( 270^\circ \)), which is \( \Re\{w\} = u < 0 \). In summary, twice around the \( s \) plane is once around the \( w(s) \) plane, because the angle is half due to the \( \sqrt{s} \). This then describes the multi-valued nature of the square root function.

This concept of analytic inverses becomes important only when the function is multi-valued. For example, if \( w(s) = s^2 \) then \( s(w) = \pm \sqrt{w} \) is multivalued. Riemann dealt with such extensions with the concept of a branch-cut with multiple sheets, labeled by a sheet number. Each sheet describes an analytic function (Taylor series), that converges within some RoC, having a radius out to the nearest pole. This Riemann’s branch cut and sheets explicitly deal with the need to define unique single valued inverses of multi-valued functions. Since the square root function has two overlapping regions, corresponding to the \( \pm \) due to the radical, there must be two connected regions, sort of like mathematical Siamese-twins, distinct, yet the same.

Branch cuts emanate from branch points, singularities that can have fractional degree, for example \( 1/\sqrt{s} \), and terminate at either another singularity having the same fractional degree, which
Figure 1.29: Colorized plots of two sheets of \( w(s) = \pm \sqrt{s} \) with the branch cut at \( \theta = \pi \). **Left:** Color map reference plane \((\zeta = s)\). **Center:** \( e^{\pi \sqrt{s}} = -\sqrt{s} \). This sheet was generated by rotating \( w \) by 180°. This should be compared to the right panel of Fig. 1.28 which shows \(-\sqrt{s}\), and to the upper sheet of Fig. 1.27. Note how the panel on the right of Fig. 1.28 matches the right half of \( s \) (purple = -90°, yellow/green = +90°) while the middle panel above comes from the left side of \( s \) (green to purple). The center sheet is green at -180°, and purple at +180°, which matches the right sheet at \( \pm 180° \) respectively (i.e., \( e^{\pi \sqrt{s}} \)). **Right:** the lower sheet of Fig. 1.27.

may be at \( \infty \). For example, suppose that in the neighborhood of the pole, at \( s_0 \) the function is

\[
f(s) = \frac{w(s)}{(s - s_0)^K},
\]

where \( w, s, s_0 \in \mathbb{C} \) and \( k \in \mathbb{Q} \). When \( k = 1 \), \( s_0 = \sigma_o + \omega_o j \) is a first degree “simple pole”, having degree 1 in the \( s \) plane, with residue \( w(s_o) \). Typically the order (and degree) are positive integers, but fractional degrees and orders are not uncommon in modern Engineering applications (Lighthill, 1978). Here shall allow both the degree and order to be fractional (\( \in \mathbb{F} \)).

When \( k \in \mathbb{F} \subset \mathbb{R} \) (a real fraction), it defines the degree of a fractional pole. In such cases there must be a branch cut, of degree \( k \). For example, if \( k = 1/2 \), the singularity (branch cut) is of degree 1/2, and there are two Riemann sheets, as shown in Figs. 1.27 and 1.28.

An important example is the Bessel function

\[
\delta(t) + \frac{1}{t} J_1(t)u(t) \leftrightarrow \sqrt{s^2 + 1},
\]

which is related to the solution to the wave equation in two-dimensional cylindrical coordinates (Table F.3, p. 278). Bessel functions are the solutions to guided acoustic waves in round pipes, or surface waves on the earth (seismic waves), or waves on the surface of a pond.

It is important to understand that the function is analytic on the branch cut, but not at the branch point. One is free to move the branch cut (at will). It does not need to be on a line, it could be cut in almost any connected manner, such as a spiral. The only rule is that it must start and stop at the matching branch points, or at \( \infty \), which must have the same degree.

There are a limited number of possibilities for the degree, \( k \in \mathbb{Z} \) or \( \in \mathbb{F} \). If the degree is drawn from \( \mathbb{R} \notin \mathbb{F} \), the pole cannot not have a residue. According to the definition of the residue, \( k \in \mathbb{F} \) will not give a residue. But there remains open the possibility of generalizing the concept of the Riemann Integral Theorem, to include \( k \in \mathbb{F} \). This seems unlikely however.

If the singularity had an irrational degree (\( k \in \mathbb{I} \)), the branch cut has the same “irrational degree.” Accordingly there would be an infinite number of Riemann sheets, as in the case of the

---

72 We shall refer to the order of a derivative, or differential equation, and the degree of a polynomial, as commonly used in Engineering applications (https://en.wikipedia.org/wiki/Differential_equation#Equation_order).
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log function. An example is $k = \pi$, for which

$$F(s) = \frac{1}{s^\pi} = e^{-\log(s^\pi)} = e^{-\pi \log(s)} = e^{-\pi \log(\rho) e^{-\pi \theta} f_j},$$

where the domain is expressed in polar coordinates $s = \rho e^{\theta j}$. When $k \in \mathbb{F}$ may be arbitrarily close to 1 (e.g., two very close primes, such as 881 and 883, primes 152 and 153), the branch cut could be very subtle (it could even be unnoticed), but it would have a significant impact on the nature of the function, and of course, on the inverse Laplace transform.

Multivalued functions: The two basic functions we review, to answer the questions about multivalued functions and branch cuts, are $w(s) = s^2$ and $w(s) = e^s$, along with their inverse functions $w(s) = \sqrt{s}$ and $w(s) = \log(s)$. For uniformity we shall refer to the complex abscissa ($s = \sigma + \omega j$) and the complex ordinate $w(s) = u + v j$. When the complex abscissa and domain are swapped, by taking the inverse of a function, multi-valued functions are a common consequence. For example, $f(t) = \sin(t)$ is single valued, and analytic in $t$, thus has a Taylor series. The inverse function $t(f)$ is not so fortunate as it is multivalued.

The modern terminology is the domain and range, or alternatively the co-domain.

Log function: Next we discuss the multi-valued nature of the log function. In this case there are an infinite number of Riemann sheets, not well captured by Fig. 1.23 (p. 99), which only displays the principal sheet. However if we look at the formula for the log function, the nature is easily discerned. The abscissa $s$ may be defined as multi-valued since

$$s_k = r e^{2\pi k j} e^{\theta j}.$$

Here we have extended the angle of $s$ by $2\pi k$, where $k$ is the sheet index $\in \mathbb{Z}$. Taking the log

$$\log(s) = \log(r) + (\theta + 2\pi k) j.$$

---

73 While there are no even primes, the minimum difference is 2. Out of $10^6$ primes there are a spacing of 80, and the distribution is linear on a log scale.)

74 The best way to create confusion is to rename things. The confusion grows geometrically with each renaming. I suspect that everyone who cares knows the terms abscissa and ordinate, and some fraction know the equivalent terms domain and range.
When \( k = 0 \) we have the principal value sheet, which is zero when \( s = 1 \). For any other value of \( k w(s) \neq 0 \), even when \( r = 1 \), since the angle is not zero, except for the \( k = 0 \) sheet.

### 1.5.5 Lec 27: Three Cauchy Integral Theorems

**Cauchy’s theorems for integration in the complex plane**

There are three basic definitions related to Cauchy’s integral formula. They are closely related, and can greatly simplify integration in the complex plane.

1. **Cauchy’s (Integral) Theorem:**

   \[
   \oint_C F(s) ds = 0, \quad (1.104)
   \]

   if and only if \( F(s) \) is complex-analytic inside of a simple closed curve \( C \) (Boas, 1987, p. 45), (Stillwell, 2010, 319). The FTCC (Eq. 1.88) says that the integral only depends on the end points if \( F(s) \) is complex-analytic. By closing the path (contour \( C \)) the end points are the same, thus the integral must be zero, as long as \( F(s) \) is complex analytic.

2. **Cauchy’s Integral Formula:**

   \[
   \frac{1}{2\pi j} \oint_{C(s_o)} \frac{F(s)}{s - s_o} ds = \begin{cases} 
   F(s_o), & s_o \in C \text{ (inside)} \\
   0, & s_o \notin C \text{ (outside)}. 
   \end{cases} \quad (1.105)
   \]

   Here \( F(s) \) is required to be analytic everywhere within (and on) the contour \( C \) (Greenberg, 1988, p. 1200), (Boas, 1987, p. 51), (Stillwell, 2010, p. 220). The value \( F(s_o) \in \mathbb{C} \) is the residue of the pole \( s_o \) of \( F(s)/(s - s_o) \).

3. **The (Cauchy) Residue Theorem** (Greenberg, 1988, p. 1241), (Boas, 1987, p. 73)

   \[
   \oint_C F(s) ds = 2\pi j \sum_{k=1}^{K} c_k = \sum_{k=1}^{K} \oint_{C(s_k)} \frac{F(s)}{s - s_k} ds, \quad (1.106)
   \]

   where the residues \( c_k \in \mathbb{C} \), corresponding to the \( k \)th poles of \( f(s) \), enclosed by the contour \( C \). By the use of Cauchy’s integral formula, the last form of the residue theorem is equivalent to the residue form.\(^{75}\)

**How to calculate the residue:** The case of first degree poles, while special, is important, since the Brune impedance only allows simple poles and zeros, increasing the utility of this special case. The residues for simple poles are \( F(s_k) \), which is complex analytic in the neighborhood of the pole, but not at the pole.

Consider the function \( f(s) = F(s)/(s - s_k) \), where we have factored \( f(s) \) to isolate the first-order pole at \( s = s_k \), with \( F(s) \) analytic at \( s_k \). Then the residue of the poles at \( c_k = F(s_k) \). This coefficient is computed by removing the singularity, by placing a zero at the pole frequency, and taking the limit as \( s \to s_k \), namely

\[
   c_k = \lim_{s \to s_k} [(s - s_k)F(s)] \quad (1.107)
   \]


\(^{75}\)This theorem is the same as a 2D version of Stokes thm (citations).
When the pole is an $N^{th}$ degree, the procedure is much more complicated, and requires taking $N - 1$ order derivatives of $f(s)$, followed by the limit process (Greenberg, 1988, p. 1242). Higher degree poles are rarely encountered, thus it is good to know that this formula exists, but perhaps it is not worth the effort to memorize it.

**Summary and examples:** These three theorems, all attributed to Cauchy, collectively are related to the fundamental theorems of calculus. Because the names of the three theorems are so similar, they are easily confused.

1. In general it makes no sense to integrate through a pole, thus the poles (or other singularities) must not lie on $C$.

2. The Cauchy integral theorem (Eq. 1.104), follows trivially from the fundamental theorem of complex calculus (Eq. 1.88, p. 109), since if the integral is independent of the path, and the path returns to the starting point, the closed integral must be zero. Thus Eq. 1.104 holds when $F(s)$ is complex analytic within $C$.

3. Since the real and imaginary parts of every complex analytic function obey Laplace’s equation (Eq. 1.94, p. 111), it follows that every closed integral over a Laplace field, i.e., one defined by Laplace’s equation, must be zero. *In fact this is the property of a conservative system*, corresponding to many physical systems. If a closed box has fixed potentials on the walls, with any distribution whatsoever, and a point charge (i.e., an electron) is placed in the box, then a force equal to $F = qE$ is required to move that charge, thus work is done. However if the point is returned to its starting location, the net work done is zero.

4. Work is done in charging a capacitor, and energy is stored. However when the capacitor is discharged, all of the energy is returned to the load.

5. Soap bubbles and rubber sheets on a wire frame, obey Laplace’s equation.

6. These are all cases where the fields are Laplacian, thus closed line integrals must be zero. Laplacian fields are commonly observed, because they are so basic.

7. We have presented the impedance as the primary example of a complex analytic function. Physically, every impedance has an associated stored energy, and every system having stored energy has an associated impedance. This impedance is usually defined in the frequency $s$ domain, as a force over a flow (i.e., voltage over current). The power $P(t)$ is defined as the force times the flow and the energy $E(t)$ as the time integral of the power

$$E(t) = \int_{-\infty}^{t} P(t)dt,$$

(1.108)

which is similar to Eq. 1.85 (p. 108) [see Section 3.2.1, Eq. 1.62 (p. 89)]. In summary, impedance and power and energy are all fundamentally related.

1.5.6 **Lec 28: Cauchy Integral Formula & Residue Theorem**

The Cauchy integral formula (Eq. 1.105) is an important extension of the Cauchy integral theorem (Eq. 1.104) in that a pole has been explicitly injected into the integrand at $s = s_0$. If the pole location is outside of the curve $C$, the result of the integral is zero, in keeping with Eq. 1.104.
However, when the pole is inside of $C$, the integrand is no longer complex analytic, and a new result follows. By a manipulation of the contour in Eq. 1.105, the pole can be isolated with a circle around the pole, and then taking the limit, the radius goes to zero.

For the related Cauchy residue theorem (Eq. 1.106) the same result holds, except it is assumed that there are $K$ simple poles in the function $F(s)$. This requires the repeated application of Eq. 1.105, $K$ times, so it represents a minor extension of Eq. 1.105. The function $F(s)$ may be written as $f(s)/P_K(s)$, where $f(s)$ is analytic in $C$ and $P_K(s)$ is a polynomial of degree $K$, with all of its roots $s_k \in C$.

**Non-integral degree singularities:** The key point is that this theorem applies when $n \in \mathbb{I}$, including fractionals $n \in \mathbb{F}$, but for these cases the residue is always zero, since by definition the residue is the amplitude of the $1/s$ term (Boas, 1987, p. 73).

**Examples:**
1. The function $1/\sqrt{s}$ has a zero residue (Hint: apply the definition of the residue Eq. 1.107).
2. When $n \in \mathbb{F}$, the residue is, by definition, zero.
3. When $n \in \mathbb{I}$, the residue is, by definition, zero.
4. When $n = 1$, the residue is non-zero.

This point is equally important when defining the inverse Laplace transform. When integrating over $\omega \in \mathbb{R}$, the value passes through all possible exponents, including all rational and irrational numbers. The only value of $\omega$ that has a residue, are those at the poles of the integrand.

### 1.5.7 Lec 29: Inverse Laplace transform & Cauchy residue theorem

The inverse Laplace transform Eq. 1.80 transforms a function of complex frequency $F(s)$ and returns a causal function of time $f(t)$

$$f(t) \leftrightarrow F(s),$$

where $f(t) = 0$ for $t < 0$. Examples are provided in Table F.3 (p. 278). We next discuss the details of finding the inverse transform by use of the Cauchy residue theorem, and how the causal requirement $f(t < 0) = 0$ comes about.

The integrand of the inverse transform is $F(s)e^{st}$ and the limits of integration are $-\sigma_o \mp \omega j$. To find the inverse we must close the curve, at infinity, and show that the integral at $\omega j \to \infty$. There are two ways to close these limits, to the right $\sigma > 0$ (RHP), and to the left $\sigma < 0$ (LHP), but there needs to be some logical reason for this choice. That logic is the sign of $t$. For the integral to converge the term $e^{st}$ must go to zero as $\omega \to \infty$. In terms of the real and imaginary parts of $s = \sigma + \omega j$, the exponential may be rewritten as $e^{\sigma t}e^{j\omega t}$. Note that both $t$ and $\omega$ go to $\infty$. So it is the interaction between these two limits that determines how we pick the closure, RHP vs. LHP.

**Case for causality ($t < 0$):** Let us first consider negative time, including $t \to -\infty$. If we were to close $C$ in the left half plane ($\sigma < 0$), then the product $\sigma t$ is positive ($\sigma < 0, t < 0$, thus $\sigma t > 0$). In this case as $\omega \to \infty$, the closure integral $|s| \to \infty$ will diverge. Thus we may not close in the LHP for negative time. If we close in the RHP $\sigma > 0$ then the product $\sigma t < 0$ and $e^{st}$ will go to zero as $\omega \to \infty$. This then justifies closing the contour, allowing for the use the Cauchy theorems.
If $F(s)$ is analytic in the RHP, the FTCC applies, and the resulting $f(t)$ must be zero, and the inverse Laplace transform must be causal. This argument holds for any $F(s)$ that is analytic in the RHP ($\sigma > 0$).

**Case of unstable poles:** An important but subtle point arises: If $F(s)$ has a pole in the RHP, then the above argument still applies if we pick $\sigma_o$ to be to the right of the RHP pole. This means that the inverse transform may still be applied to unstable poles (those in the RHP). This explains the need for the $\sigma_o$ in the limits. If $F(s)$ has no RHP poles, then $\sigma_o = 0$ is adequate, and this factor may be ignored.

**Case for zero time ($t = 0$):** When time is zero, the integral does not, in general, converge, leaving $f(t)$ undefined. This is most clear in the case of the step function $u(t) \leftrightarrow 1/s$, where the integral may not be closed, because the convergence factor $e^{st} = 1$ is lost for $t = 0$.

The fact that $u(t)$ does not exist at $t = 0$ explains the Gibbs phenomenon in the inverse Fourier transform. At times where a jump occurs, the derivative of the function does not exist, thus the time response function is not analytic. The Fourier expansion cannot converge at places where the function is not analytic. A low pass filter may be used to smooth the function, but with the cost of temporal resolution. Forcing the function to be analytic at the discontinuity, by smoothing the jumps, is an important computational method.

**1.5.8 Lec 30: Inverse Laplace transform ($t > 0$)**

**Case of $t > 0$:** Next we investigate the convergence of the integral for positive time $t > 0$. In this case we must close the integral in the LHP ($\sigma < 0$) for convergence, so that $st < 0$ ($\sigma \leq 0$ and $t > 0$). When there are poles on the $\omega = 0$ axis, $\sigma_o > 0$ assures convergence by keeping the on-axis poles inside the contour. At this point the Cauchy residue theorem (Eq. 1.106) is relevant. If we restrict ourselves to simple poles (as required for a Brune impedance), the residue theorem may be directly applied.

The most simple example is the step function, for which $F(s) = 1/s$ and thus

$$u(t) = \int_{LHP} e^{st} \frac{ds}{s} 2\pi j \leftrightarrow \frac{1}{s},$$

which is a direct application of the Cauchy Residue theorem, Eq. 1.106 (p. 121). The forward transform of $u(t)$ is straightforward, as discussed in Section 1.4.16 (p. 101). This is true of most if not all of the elementary forward Laplace transforms. In these cases, causality is built into the integral by the limits, so is not a result, as it must be in the inverse transform. An interesting problem is proving that $u(t)$ is not defined at $t = 0$.

The inverse Laplace transform of $F(s) = 1/(s + 1)$ has a residue of 1 at $s = -1$, thus that is the only contribution to the integral. More demanding cases are Laplace transform pairs

$$\frac{1}{\sqrt{\pi t}} u(t) \leftrightarrow \sqrt{\frac{\pi}{s}} \quad \text{and} \quad J_0(t)u(t) \leftrightarrow \frac{1}{s^2 + 1} = \frac{1}{(s + j)(s - j)}.$$

Many of these are easily proved in the forward direction, but are much more difficult in the inverse direction, due to the properties at $t = 0$, unless of course, the residue theorem (Eq. 1.106, p. 121) is invoked. The last $\mathcal{LT}$-pair helps us understand the basic nature of the Bessel functions $J_0(z)$, and $H_0^{(1)}(z^2)$, with a branch cut along the negative axis (see Fig. 3.2, p. 215).
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\[ u+jv = \text{besselj}(0, \pi(x+jy)) \]

\[ x \]

\[ jy \]

\[ 2 \]

\[ 1 \]

\[ 0 \]

\[ -1 \]

\[ -2 \]

\[ 2 \]

\[ 1 \]

\[ 0 \]

\[ -1 \]

\[ -2 \]

\[ u+jv = \cos(\pi(x+jy)) \]

\[ x \]

\[ jy \]

\[ 2 \]

\[ 1 \]

\[ 0 \]

\[ -1 \]

\[ -2 \]

\[ 2 \]

\[ 1 \]

\[ 0 \]

\[ -1 \]

\[ -2 \]

Figure 1.31: Left: Colorized plot of \( w(z) = J_0(\pi z) \). The first zero is at 2.405, thus appears at 0.7655 = 2.405/\( \pi \), somewhat larger than the root of \( \cos(\pi/2) \). Right: Note the similarity to \( w(z) = \sin(z) \). The LT’s have similar characteristics, as documented in Table F.3 (p. 278).

**Some open questions:** Without the use of the CRT (Eq. 1.106) it is difficult to see how to evaluate the inverse Laplace transform of \( 1/s \) directly. For example, how does one show that the above integral is zero for negative time (or that it is 1 for positive time)? The CRT neatly solves this difficult problem, by the convergence of the integral for negative and positive time. Clearly the continuity of the integral at \( \omega \to \infty \) plays an important role. Perhaps the Riemann sphere plays a role in this, that has not yet been explored.

1.5.9 **Lec 31:** Properties of the LT (e.g., Linearity, convolution, time-shift, modulation, etc.)

As shown in the table of Laplace transforms, there are integral (i.e., integration, not integer) relationships, or properties, that are helpful to identify. The first of these is a definition not a property.\(^{76}\)

\[ f(t) \leftrightarrow F(s). \]

When taking the LT, the time response is given in lower case (e.g., \( f(t) \)) and the frequency domain transform is denoted in upper case (e.g., \( F(s) \)). It is required, but not always explicitly specified, that \( f(t < 0) = 0 \), that is, the time function must be *causal* (P1: Section 1.4.17).

**Linearity:** A key property so basic that it almost is forgotten, is the linearity property of the LT. These properties are summarized as P2 of Section 1.4.17, 104).

**Convolution property:** One of the most basic and useful properties is that the product of two LTs in frequency, results in convolution in time

\[ f(t) \star g(t) = \int_0^t f(\tau)g(t-\tau)d\tau \leftrightarrow F(s)G(s), \]

where we use the \( \star \) operator as a shorthand for the convolution of two time functions.

\(^{76}\)Put this notional property in Appendix A.
A key application of convolution is filtering, which takes on many forms. The most basic filter is the moving average, the moving sum of data samples, normalized by the number of samples. Such a filter has very poor performance. It also introduces a delay of half the length of the average, which may, or may not constitute a problem, depending on the application. Another important example is a low-pass filter, that removes high frequency noise, or a notch filter that removes line-noise (i.e., 60 [Hz] in the US, and its 2d and 3d harmonics, 120 and 180 [Hz]). Such noise is typically a result of poor grounding and ground loops. It is better to solve the problem at its root, than to remove it with a notch filter. Still, filters are very important in engineering.

By taking the LT of the convolution we can derive this relationship

\[
\int_{0}^{\infty} [f(t) * g(t)]e^{-st}dt = \int_{t=0}^{\infty} \left[ \int_{0}^{t} f(\tau)g(t - \tau)d\tau \right] e^{-st}dt \\
= \int_{0}^{t} f(\tau) \left( \int_{0}^{\infty} g(t - \tau)e^{-st}d\tau \right) d\tau \\
= \int_{0}^{t} f(\tau) e^{-s\tau} \int_{0}^{\infty} g(t')e^{-st'}dt'd\tau \\
= G(s) \int_{0}^{t} f(\tau)e^{-s\tau}d\tau \\
= G(s)F(s)
\]

We first encountered this relationship in Section 1.4.7 (p. 79)) in the context of multiplying polynomials, which was the same as convolving their coefficients. Hopefully the parallel is obvious. In the case of polynomials, the convolution was discrete in the coefficients, and here it is continuous in time. But the relationships are the same.

**Time-shift propriety:** When a function is time-shifted by time \(T_o\), the LT is modified by \(e^{sT_o}\), leading to to the propriety

\[f(t - T_o) \leftrightarrow e^{-sT_o}F(s)\]

This is easily shown by applying the definition of the LT to a delayed time function.

**Time derivative:** The key to the eigen-function analysis provided by the LT, is the transformation of a time derivative on a time function, that is

\[\frac{d}{dt}f(t) \leftrightarrow sF(s)\]

Here \(s\) is the eigen-value corresponding to the time derivative of \(e^{st}\). Given the definition of the derivative of \(e^{st}\) with respect to time, this definition seems trivial. Yet that definition was not obvious to Euler. It needed to be extended to the space of complex analytic function \(e^{st}\), which did not happen until at least Riemann (1851).

**Given a differential equation of order \(K\), the LT results in a polynomial in \(s\), of degree \(K\).**

It follows that this LT property is the cornerstone of why the LT is so important to scalar differential equations, as it was to the early analysis of Pell’s equation and the Fibonacci sequence, as presented in earlier chapters. This property was first uncovered by Euler. It is not clear if he fully appreciated its significance, but by the time of his death, it certainly would have been clear to him. Who first coined the term *eigen-value* and *eigen-function*? The word *eigen* is a German word meaning *of one*. It seem likely that this term became popular somewhere between the 19th and 20th century.
Initial and final value theorems: There are much more subtle relations between $f(t)$ and $F(s)$ that characterize $f(0^+)$ and $f(t \to \infty)$. While these properties can be very important in certain application, they are are beyond the scope of the present treatment. These relate to so-called initial value theorems. If the system under investigation has potential energy at $t = 0$, then the voltage (velocity) need not be zero for negative time. An example is a charged capacitor or a moving mass. These are important situations, but better explored in a more in depth treatment.

1.5.10 Solving differential equations: Method of Frobenius

Many differential equations may be solved by assuming a power series (i.e., Taylor series) solution of the form

$$y(x) = x^r \sum_{n=0}^{\infty} c_n x^n \quad (1.109)$$

with coefficients $c_n \in \mathbb{C}$. This method, known as the method of Frobenius is quite general (Greenberg, 1988, p. 193).

Example: When a solution of this form is substituted into the differential equation, a recursion relation in the coefficients results. For example, if the equation is

$$y''(x) = \lambda^2 y(x)$$

the recursion is $c_n = c_{n-1}/n$. The resulting equation is

$$y(x) = e^{\lambda x} = x^0 \sum_{n}^{\infty} \frac{1}{n!} x^n,$$

namely $c_n = 1/n!$, thus $nc_n = 1/(n - 1)! = c_{n-1}$.

Exercise: Find the recursion relation for Bessel’s equation of order $\nu$

$$x^2 y''(x) + xy'(x) + (x^2 - \nu^2)y(x) = 0$$

Solution: If we assume a complex analytic solution of the form Eq. 1.109, we find the Bessel recursion relation

$$c_k = -\frac{1}{k(k + 2\nu)} a_{k-2}$$

(Greenberg, 1988, p. 231)

1.5.11 Lec 32: ’Lecture:' Review for Exam III (Evening Exam)
1.6 Vector Calculus (Stream 3b)

1.6.1 Lec 33 Properties of Fields and potentials

Before we can define the vector operations $\nabla()$, $\nabla\cdot()$, $\nabla\times()$, $\nabla^2()$, we must define the objects they operate on, scalar$^{77}$ and vector$^{78}$ fields.$^{79}$ The word field has two very different meanings, a mathematical definition, which defines an algebraic structure,$^{80}$ and a physical one, discussed next.

Ultimately we wish to integrate in $\in \mathbb{R}^3, \mathbb{R}^n$ and $\in \mathbb{C}^n$. Integration is quantified by several fundamental theorems of calculus, each about integration.$^{81}$

Scalar fields: We use the term scalar field interchangeably with analytic in a connected region of the spatial vector $\mathbf{x} = [x, y, z]^T \in \mathbb{R}^3$. In mathematics, functions that are piece-wise differentiable are called smooth, which is quite different from analytic. Since an analytic function may be written as a power series, it is both single valued and infinitely differentiable. A smooth function has at least one or more derivatives, but need not be single valued.

Example: The function $tu(t)$ is smooth, and has one derivative $d/dt (tu(t)) = u(t)$, but does not have a second derivative at $t = 0$. Thus $tu(t)$ is not analytic at $t = 0$. The function $\phi(r) = \pm \sqrt{x^2 + y^2 + z^2}$ is analytic everywhere except at $r = 0$, yet is double-valued, thus requiring a branch cut to fully describe it.

Initially we simplify the field by limiting the definition to an analytic surface $S(x)$, as shown in Fig. 1.32, having height $z(x, y) \in \mathbb{R}$, as a function of $x, y \in \mathbb{R}^2$ (a plane)

$$z(x, y, t) = \phi(x, y, t),$$

where $z(x, y, t)$ describes a surface that is analytic in $\mathbf{x}$. We must allow the field to optionally be a single valued function of time $t \in \mathbb{R}$, since that is the nature of the solutions of the equations we wish to solve. Examples will be given below.

For example, picture of a smooth single-valued mountain (Fig. 1.32), having isoclines (lines on a surface with constant slope).$^{82}$

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77https://en.wikipedia.org/wiki/Scalar_field
78https://en.wikipedia.org/wiki/Vector_field
79https://www.grc.nasa.gov/www/k-12/airplane/vectors.html
80https://en.wikipedia.org/wiki/Field_(mathematics)
81https://en.wikipedia.org/wiki/Line_integral
82https://en.wikipedia.org/wiki/Euler_method
Vector fields: A vector field is composed of three scalar fields. For example, the electric field used in Maxwell’s equations \( \mathbf{E}(x, t) = [E_x, E_y, E_z]^T \) [V/m] has three components, each of which is a scalar field. When the magnetic flux vector \( \mathbf{B}(x) \) is static (P5, p. 104), the potential \( \phi(x) \) [V] uniquely defines \( \mathbf{E}(x, t) \), via the gradient.

\[
\mathbf{E}(x, t) = -\nabla \phi(x, t). \quad [V/m]
\]

The electric force on a charge \( q \) is \( \mathbf{F} = q \mathbf{E} \).

Scalar potentials: The above discussion shows the importance of the potential in defining a vector field (Eq. 1.110). The difference between a potential and a scalar field is that potentials have units, thus have a physical meaning. Scalar potentials (i.e., voltage \( \phi(x, t) \) [V], temperature \( T(x, t) \) [°C] and pressure \( p(x, t) \) [Pascals]) are examples of physical scalar fields. All potentials are composed of scalar fields, but not all scalar fields are potentials. For example \( E_y(x, t) = \hat{y} \cdot \mathbf{E}(x, t) \) [V/m], the \( \hat{y} \) component of \( \mathbf{E} \), is not a potential. While \( \nabla E_y \) is mathematically reasonable, as the gradient of one component of a vector field, it has no physical meaning (as best I know).

Example: The step function \( u(t) \leftrightarrow 1/s \) is not analytic at \( t = 0 \) in time. In the Laplace frequency domain, \( 1/s \) is not complex analytic at \( s = 0 \), due to the pole.

Vector potentials: Vector potentials, like scalar potentials, are vector fields with a physical meaning (have physical units). Thus they are more complicated than scalar potentials because they are composed of three scalar fields, rather than one. But in every other way they are just as important. This follows from the Fundamental theorem of vector calculus or simply Helmholtz’ decomposition theorem.

If you found it useful set up (define) your problem using a potential, such as voltage, and then take the gradient to find \( \mathbf{E}(x, t) \), the same will hold when using the vector potential to determine the magnetic field \( \mathbf{B}(x, t) \). When operating on a scalar potential we use a gradient, whereas for the vector potential, we must operate with the curl. In Eq. 1.110 we assumed that the magnetic flux vector \( \mathbf{B}(x) \) was static, thus \( \mathbf{E}(x, t) \) is the gradient of the time dependent voltage \( \phi(x, t) \). However when the magnetic field is dynamic (not static), Eq. 1.110 is not valid due to magnetic induction: A voltage induced into a loop of wire, proportional to the time varying flux cutting across that loop of wire. This is known as Ampere’s law. In the static case the induced voltage is zero. Let’s explore just how useful the potentials can be in terms of quantifying Maxwell’s equations. When the magnetic field is time varying, Eq. 1.110 must be extended to include both the scalar \( \phi(x, t) \) and vector potentials \( \mathbf{A}(x, t) \)

\[
\mathbf{E}(x, t) = -\nabla \phi(x, t) - \frac{\partial \mathbf{A}}{\partial t}, \quad [Volts/m]
\]

(Sommerfeld, 1952, p. 146), (Feynman, 1970b, p. 18-10).

The magnetic flux \( \mathbf{B}(x, t) \) may also be written in terms of potential

\[
\mathbf{B}(x, t) = \nabla \times \mathbf{A}(x, t) + \epsilon_o \frac{\partial}{\partial t} \nabla \phi \quad [Wb/m]
\]

(1.112)

where the term in red is a wild guess on an additional electrical current term, that needs justification as discussed by (Jackson, 1967, p. 179-181) and (Feynman, 1970b, p. 18-9). Equation 1.112 is

\[\text{To be consistent with Helmholtz theorem, I wildly speculate that the expression for } \mathbf{B} \text{ is missing a term, that depends on the electrical potential } \phi(x, t), \text{ such as } \nabla \partial_t \Phi. \text{ Obviously the units must to agree, and } \nabla \cdot B = 0 \text{ must hold. The units of } \mathbf{B} \text{ are webers/m and } \mathbf{A} \text{ are [Wb/m²]} \] The units of permittivity } \epsilon_o \text{ [Farads/m]}.\]
equivalent to
\[ H(x, t) = \frac{1}{\mu_o} \nabla \times A(x, t) + \epsilon_o \frac{\partial}{\partial t} \nabla \phi \quad \text{[Wb/m]} \]

Thus the electric field strength includes both the scalar potential \( \phi(x, t) \) and magnetic flux vector potential \( A(x, t) \) components, while the magnetic field strength only depends on the magnetic potential.\(^{84}\)

To better understand the difference between scalar and vector potentials, I recommend the first chapters of Feynman (1970b, Chapter 1.1). The two discussions have a similar presentation style.

### 1.6.2 Lec 34: Gradient \( \nabla \), divergence \( \nabla \cdot \), Curl \( \nabla \times \), and Laplacian \( \nabla^2 \)

There are three key vector differential operators that are required for understanding linear partial differential equations, such as the wave and diffusion equations. All of these begin with the \( \nabla \) operator

\[ \nabla = \hat{x} \frac{\partial}{\partial x} + \hat{y} \frac{\partial}{\partial y} + \hat{z} \frac{\partial}{\partial z}. \]

The official name of this operator is *nabla*. It has three basic uses: 1) the *gradient* of a scalar field, the 2) *divergence* of a vector field, and 3) the *curl* of a vector field. If properly noted, the shorthand notation \( \nabla \phi(x, t) = (\hat{x} \partial_x + \hat{y} \partial_y + \hat{z} \partial_z) \phi(x, t) \) is convenient.

<table>
<thead>
<tr>
<th>Name</th>
<th>Input</th>
<th>Output</th>
<th>Operator</th>
<th>Mnemonic</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gradient</td>
<td>Scalar</td>
<td>Vector</td>
<td>( \nabla() )</td>
<td>Grad</td>
</tr>
<tr>
<td>Divergence</td>
<td>Vector</td>
<td>Scalar</td>
<td>( \nabla \cdot () )</td>
<td>Div</td>
</tr>
<tr>
<td>Curl</td>
<td>Vector</td>
<td>Vector</td>
<td>( \nabla \times () )</td>
<td>Curl</td>
</tr>
<tr>
<td>Laplacian</td>
<td>Scalar</td>
<td>Scalar</td>
<td>( \nabla \cdot \nabla = \nabla^2 () )</td>
<td>DoG</td>
</tr>
<tr>
<td>Vector Laplacian</td>
<td>Vector</td>
<td>Vector</td>
<td>( \nabla \nabla = \nabla^2 () )</td>
<td>GoD</td>
</tr>
</tbody>
</table>

Table 1.3: The three vector operators manipulate scalar and vector fields, as indicated here. The gradient converts scalar fields into vector fields. The divergence maps vector fields to scalar fields. Finally the curl maps vector fields to vector fields. It is helpful to have a name for second order operators (e.g., DoG, GoD: mnemonics defined in Sect. 1.6.11, p. 168).

**Basic differential vector operator definitions:** The basic definitions of each of the vector operators are summarized in Table 1.3.

**Gradient:** The *gradient* transforms a scalar field into vector field. In \( \mathbb{R}^3 \) the gradient of a scalar field \( \nabla \phi(x) \) is defined as

\[ \nabla \phi(x) = \left( \frac{\partial}{\partial x} \hat{x} + \frac{\partial}{\partial y} \hat{y} + \frac{\partial}{\partial z} \hat{z} \right) \phi(x) = \frac{\partial \phi}{\partial x} \hat{x} + \frac{\partial \phi}{\partial y} \hat{y} + \frac{\partial \phi}{\partial z} \hat{z}. \]  

The gradient may be factored into a unit vector \( \hat{n} \), as defined in Fig. 1.32, defining the direction of the gradient, and the gradient’s length \( ||\nabla()|| \), defined in terms of the norm. Thus the gradient of

\(^{84}\)As will be discussed in Section 1.6.12, based on a symmetry argument, it seems likely that Eq. 1.112 should include a current, such as a capacitive \( \nabla \phi(x, t) \).
\( \phi(x) \) may be written in “polar coordinates” as \( \nabla \Phi(x) = \left\| \nabla(\Phi) \right\| \hat{n} \), useful for defining \( \hat{n} \) as

\[
\hat{n} = \frac{\nabla(\Phi(x))}{\left\| \nabla(\Phi) \right\|}.
\]

Important examples of the use of the gradient include the electric field vector \( E(x) = -\nabla \phi(x) \) [Volts/m], which is the gradient of a voltage [V], and the force density \( f(x) = -\nabla \rho(x) \) [N/m], which is the gradient of a pressure [Pa].

**Divergence:** The divergence of a vector field results in a scalar field. For example, the divergence of the electric field flux vector \( D(x) \) [Col/m²] equals the scalar field charge density \( \rho(x) \) [Col/m³]

\[
\nabla \cdot D(x) \equiv \left( \frac{\partial}{\partial x} \hat{x} + \frac{\partial}{\partial y} \hat{y} + \frac{\partial}{\partial z} \hat{z} \right) \cdot D(x) = \frac{\partial D_x}{\partial x} + \frac{\partial D_y}{\partial y} + \frac{\partial D_z}{\partial z} = \rho(x). \tag{1.114}
\]

This it is analogous to the scalar (dot) product between two vectors.

When working with guided waves (narrow tubes of flux), when the diameter is small compared with the wavelength (P10, p. 105), the divergence is

\[
\nabla \cdot D(x) = \nabla_r D_r = \frac{1}{A(r)} \frac{\partial}{\partial r} A(r) D_r(r), \tag{1.115}
\]

where \( r \) is the distance down the horn (range variable), \( A(r) \) is the area of the iso-response surface, as a function of range \( r \), and, \( D_r(r) \) is the radial component of vector \( D \), as a function of range \( r \). For example, in spherical coordinates the area \( A(r) = A_o r^2 \) is proportional to the square of the range. This expression reduces to the radial component of the divergence of \( D(x) \) in spherical coordinates. In cylindrical coordinates \( A(r) = A_o r \), and in rectangular coordinates the area \( A = \pi r_o^2 \) is independent of the range \( r \).

A general and detailed derivation of these cases is derived in Section ??, p. ??.

**Curl:** The curl transforms a vector field into a vector field. For example the curl of the magnetic intensity \( H(x) \) [Amp/m²] vector is equal to the vector current density \( C(x) \) [Amps/m²]:

\[
\nabla \times H(x) \equiv \begin{vmatrix}
\hat{x} & \hat{y} & \hat{z} \\
\frac{\partial}{\partial x} & \frac{\partial}{\partial y} & \frac{\partial}{\partial z} \\
H_x & H_y & H_z
\end{vmatrix} = C(x). \tag{1.116}
\]

The notation \( \left| \cdot \right| \) indicates the determinant (Appendix A, p. 251), \( \partial_x \) is shorthand for \( \partial/\partial x \) and \( H = [H_x, H_y, H_z]^T \).

**Laplacian:** The Laplacian \((\nabla^2 \equiv \nabla \cdot \nabla)\) is

\[
\frac{\partial^2}{\partial x^2} + \frac{\partial^2}{\partial y^2} + \frac{\partial^2}{\partial z^2}, \tag{1.117}
\]

takes a scalar field onto a scalar field.

Example: Starting from a scalar field, the gradient produces a vector, which is then operated on by the divergence to take the output of the gradient back to a scalar field. One of the classic
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cases is the Laplacian of a voltage field. The gradient of the scalar voltage \( \Phi(x) \) [V] results in the electric field vector \( \mathbf{E} \)

\[
\mathbf{E}(x) = [E_x(x), E_y(x), E_z(x)]^T = -\nabla \Phi(x), \quad \text{[V/m]}
\]

which in free space is proportional to the electric flux \( \mathbf{D} = \varepsilon_o \mathbf{E} \) [C/m\(^2\)], the divergence of which gives the charge density \( \rho(x) \) [C/m\(^3\)]. Here \( \varepsilon_o \) [F/m] is the *vacuum permittivity*, which is \( \approx 8.8542 \times 10^{-12} \) [F/m].

**Example:** The simplest example of a scalar field is the voltage between two very large (think \( \infty \)) conducting parallel planes, or plates (large so that we may ignore the edge effects). In this case the voltage varies linearly (the voltage is complex analytic) between the two plates. For example

\[
V(x, y, z) = V_o (1 - x)
\]

is a scalar field. At \( x = 0 \) the voltage is \( V_o \) and at \( x = 1 \) the voltage is zero. Between 0 and 1 the voltage varies linearly. Thus \( V(x, y, z) \) defines a *scalar field*.

If the same setup were used but the two plates were \( 1 \times 1 \) [cm\(^2\)], with a 1 [mm] air gap, there will be a small “fringe” effect at the edges that would (slightly) modify the ideal fields. This effect can be made small by changing the air gap to area ratio, so that the ends do not significantly impact the capacitor’s value.

**Example:** A second classic example is an acoustic pressure field \( \varrho(x, t) \) [Pa], which defines a vector force density \( \mathbf{f}(x, t) = -\nabla \varrho(x, t) \) [N/m\(^2\)] (Eq. 1.128, p. 142). When this force density [N/m\(^2\)] is integrated over an area, the net radial force [N] is

\[
F_r = - \int_S \nabla \varrho(x)||dS||. \quad \text{[N]}
\]

An inflated balloon with a static internal pressure of 3 [atm], in an ambient pressure of 1 [atm], forms a sphere due to the elastic nature of the rubber, which acts as a stretched spring under tension. The net force on the surface of the balloon is its area times the pressure drop of 2 atm across the surface. Thus the static pressure is

\[
\varrho(x) = 3u(r_o - r) + 1, \quad \text{[Pa]}
\]

where \( u(r) \) is a step function of the radius \( r = ||x|| > 0 \), centered at the center of the balloon, having radius \( r_o \).

Taking the gradient gives the negative\(^{85} \) of the radial force density (i.e., perpendicular to the surface of the balloon)

\[
-f_r(r) = \nabla \varrho(x) = \frac{\partial}{\partial r} 3u(r_o - r) + 1 = -2\delta(r_o - r). \quad \text{[Pa]}
\]

This describes a static pressure that is 3 atmospheres inside the balloon, and 1 atmosphere [At] (1 [At] = 10\(^5\) [Pa]) outside. Note that the net positive force density is the negative of the gradient of the static pressure.

Taking the divergence of this radial force gives the Laplacian of the scalar pressure field

\[
\nabla^2 \varrho(x) = \nabla \cdot \nabla \varrho(x) = -\nabla \cdot \mathbf{f}(x).
\]

\(^{85}\)The force is pointing out, stretching the balloon.
From Appendix 5.2.1 (Eq. 5.11 p. 237) the Laplacian in spherical coordinates simplifies (Eq. 5.13, p. 237)
\[
\nabla^2 \rho(x) = \frac{1}{r^2} \frac{\partial}{\partial r} r^2 \frac{\partial}{\partial r} \rho(x) = \frac{1}{r} \frac{\partial^2}{\partial r^2} r \rho(x).
\]

Thus the net force \( f(x) = [F_r, 0, 0]^T \), (Eq. 1.119), in spherical coordinates, has a radial component \( F_r \), and angular components of zero. This force is well approximated by a delta function across the thin sheet of stretched rubber.

Example: The previous example may be extended in an interesting way to the case of a rigid hose, a rigid tube, terminated on the right in an elastic medium (the above example of a ballon, for example an automobile tire. On the far left lets assume there is a pump, injecting the fluid into the hose. Lets consider two different types of fluid air and water. In the case of air it may be treated as a compressable fluid, whereas water is incompressible. This is a relative term, determined by the compliance of the ballon (i.e., tire) at the end of the rigid tube (pump hose).

This may be modeled as either an electrical or mechanical system. If we take the electrical analog, the pump is a current source, injecting charge \( Q \), into the hose, which being rigid cannot expand (has a fixed volume). The hose may be modeled as a resistor, and the tire as a capacitor \( C \), which fills with charge as it is delivered via the resistor, from the pump. A capacitor obeys the same law as a spring \( F = KV \), or in electrical terms, \( Q = CV \). Here \( V \) is the voltage, which acts as a force \( F \), \( Q \) is the charge, which acts like the mass of the fluid. The charge is conserved, just as the mass of the fluid is conserved, meaning they cannot be created or destroyed. The flow of the fluid is called the flux, which is the general term for the mass or charge current. The two equations may be rewritten directly in terms of the force \( F \), \( V \) and flow, as an electrical current \( I = dQ/dt \) of mass flux \( J = dM/dt \), giving two impedance relations

\[
I = \frac{d}{dt} CV \quad \text{[amps]} \quad (1.120)
\]

for the electrical analogue, and

\[
J = \frac{d}{dt} CF \quad \text{[kgm/m²]} \quad (1.121)
\]

It is common to treat the stiffness of the ballon, which acts as a spring with compliance \( C \) (stiffness \( K = 1/C \)), in which case, the equations reduce to the same equation, in terms of an impedance \( Z \), typically defined in the frequency domain as the ratio of the generalized force over the generalized flow

\[
Z(s) = \frac{1}{sC}. \quad \text{[Ohms]}
\]

In the case of the mechanical system \( Z_m(s) \equiv F/J \), and for the electrical system \( Z_e(s) \equiv V/I \). It is convential to use the term Ohms when working with any impedance. It is convienent to use a uniform terminology for different physical situations, greatly simplifying the notation.

While the two systems are very different in their physical realization, the are mathematically identical, and form a perfect analogue. The formula for the impedance is typically expressed in the Laplace frequency domain, which of course is the Laplace transform of the time variables. In the frequency domain is then Ohm’s law becomes Eq. 1.121 for the case of a spring and Eq. 1.120 for the capacitor.

The final solution of this system is solved in the frequency domain. The impedance seen by the source is the sum of the resistance \( R \) added to the impedance of the load, giving

\[
Z = R + \frac{1}{sC}.
\]
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The solution is simply the relation between the force and the flow, as determined by the action of the source on the load $Z(s)$. The final answer is given in terms of the voltage across the compliance in terms of the voltage $V_s$ (or current $I_s$) due to the source. Once the algebra is done, in the frequency domain, the voltage across the compliance $V_c$ divided by the voltage of the source is given as

$$\frac{V_c}{V_{source}} = \frac{R}{R + 1/sC}. \quad (1.122)$$

Thus the problem reduces to some algebra in the frequency domain. The time domain response is found by taking the inverse Laplace transform, which in this case has a simple pole at $s_p = 1/RC$. Cauchy’s residue formula gives the final answer, which describes how the voltage across the compliance builds exponentially with time, from zero to the final value. Given the voltage, the current may also be computed, as a function of time. This then represents the entire process of either blowing up a balloon, or charging a capacitor, the difference only being the physical notation, the math being identical.

Note that the differential equation is first order in time, which in frequency means the impedance has a single pole. This means that the equations describing the charging of a capacitor, or pumping up a balloon is a diffusion process. If we had taken the impedance of the mass of the fluid in the hose into account, we would have a lumped parameter model of the wave equation, with a second order system. This is mathematical the same as presented in a homework problem of train cars (masses) connected together by springs (Fig. 3.1 p. 210).

Example: The voltage $\phi(x,t) = e^{-\kappa \cdot x} u(t - x/c) \leftrightarrow \frac{1}{s} e^{-\kappa \cdot x}$ [Volts] is an important case since it represents one of d’Alembert two solutions (Eq. 1.95, p. 112) of the wave equation (Eq. 1.24, p. 57), as well as an eigen-function of the gradient operator $\nabla$. From the definition of the scalar (dot) product of two vectors (Fig. 1.16, p. 82),

$$\kappa \cdot x = \kappa_x x + \kappa_y y + \kappa_z z = ||\kappa|| ||x|| \cos \theta_{\kappa x},$$

where $||\kappa|| = \sqrt{\kappa^2_x + \kappa^2_y + \kappa^2_z}$ and $||x|| = \sqrt{x^2 + y^2 + z^2}$ are the lengths of vectors $\kappa$, and $x$ and $\theta_{\kappa x}$ is the angle between them. As before $s = \sigma + \omega j$ is the Laplace frequency.

To keep things simple let $\kappa = [\kappa_x, 0, 0]^T$ so that $\kappa \cdot x = \kappa_x x \hat{x}$. We shall soon see that $||\kappa|| = 2\pi/\lambda$, follows from the basic relationship between a wave’s radian frequency $\omega = 2\pi f$ and its wavelength $\lambda$

$$\omega \lambda = c_o. \quad (1.123)$$

As frequency increases, the wavelength becomes shorter. This key relationship may have been first research by Galileo (c.1564), followed by (c.1627) Mersenne\(^86\) (Fig. 1.15, p. 82).

Exercise: Show that Eq. 1.122 is an eigen-function of the Gradient operator $\nabla$. Taking the gradient of $\phi(x,t)$ gives

\(^86\) http://www-history.mcs.st-and.ac.uk/Biographies/Mersenne.html

“In the early 1620s, Mersenne listed Galileo among the innovators in natural philosophy whose views should be rejected. However, by the early 1630s, less than a decade later, Mersenne had become one of Galileo’s most ardent supporters.”

\[ \nabla e^{-\kappa x}u(t) = -\nabla(\kappa \cdot x) e^{-\kappa x}u(t) \]
\[ = -\kappa e^{-\kappa x}u(t), \]
or in terms of \( \phi(x,t) \)
\[ \nabla \phi(x,t) = -\kappa \phi(x,t) \leftrightarrow -\frac{s}{c}e^{-\kappa x}. \]

Thus \( \phi(x,t) \) is an eigen-function of \( \nabla \), having the vector eigen-value \( \kappa \). As before \( \nabla \phi \) is proportional to the current, since \( \phi \) is a voltage, and the ratio, the eigen-value acts like a mass. In general the units provide the physical interpretation of the eigen values and their spectrum. A famous example is the Rydberg spectrum of the Hydrogen atom.

**Exercise:** Compute \( \hat{n} \) for \( \phi(x,t) \) (Eq. 1.122). \( \hat{n} = \kappa/||\kappa|| \). This represents a unit vector in the direction of the current.

**Exercise:** If the sign of \( \kappa \) is negative, what are the eigen-vectors and eigen-values of \( \nabla \phi(x,t) \)?
\[ \nabla e^{-\kappa x}u(t) = -\kappa \cdot \nabla(x) e^{-\kappa x}u(t) \]
\[ = -\kappa e^{-\kappa x}u(t). \]

Nothing changes other than the sign of \( \kappa \). Physically this means the wave is traveling in the opposite direction, corresponding to the forward and retrograde d’Alembert waves.

Prior to this section, we have only considered the Taylor series in one variable, such as for polynomials \( P_n(x), x \in \mathbb{R} \) (Sect. 1.4.1, Eq.1.27 p. 59) and \( P_n(s), s \in \mathbb{C} \) (Sect. 1.5.2, Eq.1.47 p. 73). The generalization from real to complex analytic functions led to the Laplace Transform, an the hosts of integration theorems (FTCC, Cauchy I, II, III). What is in store when we generalize from one spatial variable (\( \mathbb{R} \)) to three (\( \mathbb{R}^3 \))?

**Exercise:** If \( E(x,t) = E_x \hat{x} \), express \( E(x,t) \) in terms of the voltage potential \( \phi(x,t) \) [V].

The electric field strength may be found from the voltage as
\[ E(x,t) = -\nabla \phi(x,t) = -\hat{x} \frac{\partial}{\partial x} \phi(x,t) \quad [V/m]. \]

**Exercise:** Find the velocity \( v(t) \) of an electron in a field \( E \). From Newton’s 2d law, \(-qE = m_e \dot{v}(t) \) [Nt], where \( m_e \) is the mass of the electron. Thus we must solve this first order differential equation to find \( v(t) \). This is easily done in the frequency domain \( v(t) \leftrightarrow V(\omega) \).

**Role of Potentials:** Note that the scalar fields (e.g., temperature, pressure, voltage) are all scalar potentials, summarized in Fig. 1.19 (p. 94). In each case the gradient of the potential results in a vector field, just as in the electric case above (Eq. 1.110).

It is important to understand the physical meaning of the gradient of a potential, which is typically a generalized force (Electric field, acoustic force density, temperature flux), which in turn generates a flow (current, velocity, heat flux). The ratio of the gradient over the flow determines the impedance.
1. **Example 1:** The voltage drop across a resistor causes a current to flow, as described by Ohm’s law. Taking the difference in voltage between two points is a crude form of gradient when the frequency $f$ [Hz] is low, such that the wavelength is much larger than the distance between the two points. This is the essence of the quasi-static approximation.

2. **Example 2:** The gradient of the pressure gives rise to a force density in the fluid medium (air, water, oil, etc.), which causes a flow (velocity vector) in the medium.

3. **Example 3:** The gradient of the temperature also causes a flow of heat, which is proportional to the thermal resistance, given Ohm’s law for heat (Feynman, 1970c, p. 3-7).

4. **Example 4:** Nernst’ potential: When a solution contains charged ions, it is called an electro-chemical Nernst’ potential $N(x, t)$.

Thus in the above examples there is a potential, the gradient of which is a force, which when applied to the medium (an impedance) causes a flow (flux or current) proportional to that impedance, due to the medium. This is a very general set of concepts, worthy of some thought. In every case there is a force and a flow. The product of the force and flow is a power, while the ratio an impedance, which we have learned about in Section 1.4.11 in the form of 2x2 ABCD matrices (Eq. 1.70, p. 92).

**Exercise:** Show that the integral of Eq. 1.110 is an anti-derivative.

The solution requires the definition of the anti-derivative, defined by the FTC (Eq. 1.86, p. 109):

$$\phi(x, t) - \phi(x_o, t) = \int_{x_o}^{x} E(x, t) \cdot dx = \int_{x_o}^{x} \nabla \phi(x, t) \cdot dx$$

$$= - \int_{x_o}^{x} \left( \hat{x} \frac{\partial \phi}{\partial x} + \hat{y} \frac{\partial \phi}{\partial y} + \hat{z} \frac{\partial \phi}{\partial z} \right) \phi(x, t) \cdot dx$$

$$= - \int_{x_o}^{x} \left( \hat{x} \frac{\partial \phi}{\partial x} + \hat{y} \frac{\partial \phi}{\partial y} + \hat{z} \frac{\partial \phi}{\partial z} \right) \cdot (\hat{x} dx + \hat{y} dy + \hat{z} dz)$$

$$= - \int_{x_o}^{x} \frac{\partial \phi}{\partial x} dx - \int_{y_o}^{y} \frac{\partial \phi}{\partial y} dy - \int_{z_o}^{z} \frac{\partial \phi}{\partial z} dz$$

$$= - \int_{x_o}^{x} d\phi(x, t)$$

$$= - \left( \phi(x, t) - \phi(x_o, t) \right).$$

is the anti-derivative. We may verify this by taking the gradient of both sides

$$\nabla \phi(x, t) - \nabla \phi(x_o, t) = - \nabla \int_{x_o}^{x} E(x, t) \cdot dx = E(x, t).$$

Applying the FTC (Eq. 1.86, p. 109), the anti-derivative must be $\phi(x, t) = E_x \hat{x} + 0\hat{y} + 0\hat{z}$. This very same point, made by Feynman (1970c, p. 4-1, Eq. 4.28), is worth reading.

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87[https://en.wikipedia.org/wiki/Thermal_conduction#Fourier’s_law](https://en.wikipedia.org/wiki/Thermal_conduction#Fourier’s_law)  
Given that the force on a charge is proportional to the gradient of the potential, from the above Exercise showing that the integral of the gradient only depends on the end points, the work done in moving a charge only depends on the limits of the integral. This is the definition of a conservative field. This only holds when $E$ is determined by Eq. 1.110, and the medium of the charge has no friction (i.e., there are no other forces on the charge).

The conservative field: This leads to an important question: “When is a field conservative?” A field is conservative when the work done by the motion is independent of the path of motion. This seems related to the FTC, that the integral only depends on the end points, for a case where the integral represents the total work done.

A proper answer must wait for the introduction of the fundamental theorem of vector calculus, discussed in Section 1.6.12 (Eq. 1.194, p. 171), but it is insightful to give several specific examples:

Example: A voltage is a scalar potential, the gradient of which defines the electric field (Eq. 1.110), which drives a current (flow) across a resistor (or any impedance). If the impedance is infinite, the flow will be zero, leading to zero power dissipation. When the impedance is lossless, the system is conservative.

Example: The viscosity of air is quite small, thus as a simplifying assumption, is typically assumed to be zero. However at high frequencies, where the wavelength is small (e.g., at 100 [kHz] $\lambda = c_o/f = 345/10^5 = 3.45$ [mm]), the viscosity is significant in sound propagation, and there is an associated power loss. $^89$ When the viscosity is taken into account, the field is lossy thus no longer conservative.

Example: If a temperature field is a time-varying constant (i.e., $T(x, t) = T_o(t)$), there is no “heat flux,” since $\nabla T_o(t) = 0$. When there is no heat flux (i.e., flux, or flow), there is no heat power, since the power is the product of the force times the flow.

Example: The force of gravity is given by the gradient of Newton’s gravitational potential (Eq. 1.22, p. 56)

$$F = -\nabla \phi_g(r).$$

Historically this was the first conservative field, used to explain the elliptic orbits of the planets around the sun.

### 1.6.3 Partial differential equations and field evolution:

There are three main category of partial differential equations (PDEs):

1. **diffusion equations** (Eq. 1.124), which describes the evolution of the scalar temperature $T(x, t)$ (a scalar potential) and gradients of solution concentrations (i.e., ink in water) and Brownian motion.

2. **scalar wave equations** (Eq. 1.23), which describe the propagation of a pressure $p(x, t)$ sound field (a scalar potential),

3. **vector wave equations** (i.e., Maxwell’s equations), which describe the propagation of electric $E(x, t)$ and $D(x, t)$ and magnetic $B(x, t)$ and $H(x, t)$ vector fields.

All of these PDEs describe the evolution of either scalar fields (pressure $p(x, t)$ and temperature $T(x, t)$ or vector fields ($E$, $D$, $B$, $H$) in space, as functions of space $x$ and time $t$.

$^89$https://en.wikipedia.org/wiki/Laminar_flow#Examples
They are distinguished by the order of the derivative with respect to time. The diffusion equation is first order in time, while propagation is second order. When these equations are Laplace transformed, diffusion has a single real root, while the wave equation has pairs of complex conjugate roots. This is the most important distinction between diffusion and propagation. The diffusion equation is thus said to be parabolic (first order in time, second order in space), while the wave equation is said to be hyperbolic (second order in time and space). Laplace’s equation is a special case, where the solution is static, where the solution is said to be elliptic. These categories have descriptive, but not significant mathematical utility.

Partial differential equations (PDEs) define the “evolution” of the field. There are two basic categories of field evolution, diffusion and propagation.

1. **Diffusion**: The simplest and easiest PDE example, easily visualized, is a static (time invariant) scalar temperature field $T(x)$ [°C]). Just like an impedance or admittance, a field has regions where it is analytic, and for the same reasons, $T(x,t)$ satisfies Laplace’s equation

$$\nabla^2 T(x,t) = 0.$$

Since there is no current when the field is static, such systems are lossless, thus are conservative.

When $T(x,t)$ depends on time (is not static), it is described by the diffusion equation

$$\nabla^2 T(x,t) = \kappa \frac{\partial}{\partial t} T(x,t), \quad (1.124)$$

a rule for how $T(x,t)$ evolves with time from its initial state $T(x,0)$. The constant $\kappa$ is called the thermal conductivity which depends on the fluid in the container, with $s \kappa$ being the admittance per unit area.

Note that if $T(x,\infty)$ reaches steady state as $t \to \infty$, it evolves into a static state, thus $\nabla^2 T = 0$. This depends on what is happening at the boundaries. When the wall temperature of a container is a function of time, then so will the internal temperature continue to change, but with a delay, that depends on the thermal conductivity $\kappa$.

Such a system is analogous to an electrical resistor-capacitor series circuit, connected to a battery. The wall temperature and the voltage on the battery represent the potential driving the system, the thermal conductivity $\kappa$ and the electrical resistor are analogous, and the fluid heat capacity and the electrical capacitor, are being heated (charged) by the flux. In both cases Ohm’s law defines the flux.

How this happens can only be understood once the solution to the equations has been established.

2. **Propagation** Pressure and electromagnetic waves are described by a scalar potential (pressure) (Eq. 1.23, p. 56) and a vector potential (electromagnets) (Eq. 1.199, p. 176) resulting in scalar and vector wave equations.

All these partial differential equations, scalar and vector wave equations, and the diffusion equation, depend on the Laplacian $\nabla^2$, which first appeared to us as Laplace’s equation, which we first saw with the Cauchy-Riemann conditions (Eq. 1.94, p. 111).

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Postulate P3, p. 104.
1.6. STREAM 3B: VECTOR CALCULUS (10 LECTURES)

**The vector Taylor series:** Next we shall expand on the concept of the one-dimensional Taylor series, a function of one variable, to \( x \in \mathbb{R}^3 \). Just as we generalized the derivative with respect to the real frequency variable \( \omega \in \mathbb{R} \) to the complex analytic frequency \( s = \sigma + \omega j \in \mathbb{C} \), here we generalize the derivative with respect to \( x \in \mathbb{R} \), to the vector \( x \in \mathbb{R}^3 \).

Since the scalar field is analytic in \( x \), it is a perfect place to start. Assuming we have carefully defined the Taylor series 1.41 (p. 69) in one and two (Eq. 1.91, p. 111) variables, the Taylor series of \( f(x) \) in \( x = 0 \in \mathbb{R}^3 \) may be defined as

\[
f(x + \delta x) = f(x) + \nabla f(x) \cdot \delta x + \frac{1}{2!} \sum_k \sum_l \frac{\partial^2 f(x)}{\partial x_k \partial x_l} \delta x_k \delta x_l + \text{HOT.} \tag{1.125}
\]

From this definition it is clear that the gradient is the generalization of the second term in the 1D Taylor series expansion.

**Vector field:** A vector field is an ordered set of scalar fields. For example

\[
A(x) = [\phi(x), \psi(x), \theta(x)]^T
\]

is a vector field in \( \mathbb{R}^3 \) when each of the three function is differentiable (i.e., analytic). For example \( A(x) = [x, xy, xyz]^T \) is a legal vector field (the components are analytic in \( x \)).

Taking an example from Maxwell’s equations, the magnetic flux vector is given by

\[
B(x,t) = -\nabla \times A(x,t).
\]

We shall see that this is always true because the magnetic charge \( \nabla \cdot B(x,t) \) is always 0.

To verify that a field is a potential, check out the units [V, A, °C]. However a proper mathematical definition is that the potential must be an analytic function of \( x \) and \( t \), so that one may operate on it with \( \nabla () \) and \( \nabla \times () \). The divergence of a scalar field is not a legal operation.

An important discussion of vector potentials, with extensive examples, may be found in Feynman (1970c, p. 14-1 to 14.3). If you need to master vector potentials, Read Ch. 14.1 of Feynman (1970c).

**In summary:** For every potential: a force, proportional to the gradient of a scalar, drives a generalized flow. When the system is linear (P2, p. 104), the product of the force and flow is the power, and the ratio (force/flow) is an impedance (Fig. 1.19, p. 94). This impedance statement is called either Ohm’s law, Kirchhoff’s laws, Laplace’s law, or Newton’s laws. In the simplest of cases, they are all linearized (proportional) relationships, between a force and a flow. Very few impedance relationships are inherently linear over a large range of force or current, but for physically useful levels, they are typically linear (P2, p. 104).

Note that it is the difference in the potential (i.e., voltage, temperature, pressure) that is proportional to the flux. This is a major simplification of the gradient relationship, justified by the quasistatic assumption P10 (p. 105), and is the physical basis of the Fundamental theorem of algebra (p. 75), since the impedance is the ratio of the force (potential drop) over the flow (flux).

In electrical circuits it is common to define a zero potential ground point, that all voltages use as the referenced potential. This results in abstracting away (hiding) the difference in voltage. The ground is useful convention, as a simplifying rule, but it obscures the physics, and obscures the fact that the voltage is not the force. Rather the force is the voltage drop, referenced to the ground, which is defined as zero volts. More precisely, \( F = -\nabla \Phi \). It seems misleading (more
precisely it is wrong) to state Ohm’s law as the voltage over the current, since Ohm’s law actually says that the voltage drop (i.e., voltage gradient) over the current, that defines the impedance. Like a voltage, the pressure is the potential, the gradient of which is a force density, which drives the flow. More on this in section 1.6.12 (p. 168), where we introduce The Fundamental theorem of vector calculus (aka Helmholtz’ decomposition theorem), which generalizes Ohm’s law to include circulation (e.g., EM magnetic) effects.

1.6.4 Lec 35: Scalar Wave Equation (Acoustics)

In this Section we discuss the general solution to the wave equation. In general the wave equation has two forms, scalar waves (acoustics) and vector waves (electromagnetics). These have an important mathematical distinction, but have a similar solution space, one scalar and the other vector. To understand the differences we start with the scalar wave equation.

The scalar wave equation: The starting point for understanding PDEs is to explore the scalar wave equation, thus we shall limit our analysis to acoustics, the classic case of scalar waves. Acoustic wave propagation was first analyzed mathematically by Isaac Newton (electricity had yet to be discovered) in his famous book Principia (1687), where he was the first to calculate the speed of sound, based on conservation of mass and momentum. His early development understandably ignored viscous and thermal losses, which can be significant in a thin region called the boundary layer, at acoustic frequencies when the radius of the container (i.e., the horn) becomes less than the viscous boundary layer (e.g., much less than 1 mm) as first described in Helmholtz (1863b) and extended to include thermal losses by Kirchhoff (1868, 1974, English translation), as succinctly summarized by Rayleigh (1896). The experimental verification of these theoretical studies was ultimately verified by Mason (1927, 1928).

Loss-less horns include an important generalization of the solution of the 1-dimensional (1D) wave equation, in regions where the area \( A(r) \) of the horn varies along the range axial (e.g., along the direction of wave propagation), as depicted in Fig. 1.33. Classic applications of horns include vocal tract acoustics, loudspeaker design, cochlear mechanics, and cases having wave propagation in periodic media (Brillouin, 1953).

For the 1D scalar wave equation (guided waves, aka, acoustic horns) the Laplacian is

\[
\nabla^2 r \phi(r, t) \equiv \nabla \cdot \nabla \phi(r, t) = \frac{1}{A(r)} \frac{\partial}{\partial r} A(r) \frac{\partial}{\partial r} \phi(r, t), \tag{1.126}
\]

which assumes that the horn’s diameter \( d \) is less than first allowed “radial mode’s” (a half wavelength) (i.e., \( d < \lambda/2 \)).\(^91\) The term on the right is an approximation that applies for thin tubes (e.g., spherical, cylindrical coordinates, etc.), with angular symmetry, where the potential \( \psi(x, t) \leftrightarrow \Psi(x, s) \) is averaged over the iso-potential cross-sectional area (Sect. 1.6.7, p. 149). Such formulations, which explicitly depend on the area of the horn, are called Webster horn equations.

It follows that the 1D acoustic wave equation is

\[
\frac{1}{A(r)} \frac{\partial}{\partial r} A(r) \frac{\partial}{\partial r} \phi(r, t) = \frac{1}{c_0^2} \frac{\partial^2}{\partial t^2} \phi(r, t) \leftrightarrow \frac{s^2}{c_0^2} P(r, s), \tag{1.127}
\]

\(^{91}\)This condition may be written in forms, the most common being that for a ridged walled tube \( ka < 1 \), where \( k = 2\pi/\lambda \) and \( a \) is the radius. This may be expressed in terms of the diameter as \( \frac{2\pi a}{\lambda} < 1 \), which is \( d < \lambda/\pi < \lambda/2 \). Thus \( d < \lambda/2 \) is a more precise metric, by a factor of \( \pi/2 \approx 1.6 \).
where \( g(r, t) \leftrightarrow P(r, s) \) is the average pressure (Hanna and Slepian, 1924; Mawardi, 1949; Morse, 1948), Olson (1947, p. 101), Pierce (1981, p. 360). Extensive experimental analysis for various types of horns (conical, exponential, parabolic) along with a review of horn theory is provided by Goldsmith and Minton (1924).

**The limits of the Webster horn equation:** It is frequently (i.e., always) stated that the Webster Horn equation (WHEN) is a fundamentally limited, and is an approximation that only applies at low frequencies. However in all these discussions it is assumed that the area function \( A(r) \) is the horn’s cross-sectional area, not the area of the iso-pressure wave-front (Morse, 1948; Shaw, 1970; Pierce, 1981).

In Section 5.4 (p. 239, 181) it is shown that these “limitations” may be totally avoided, making the Webster theory of the horn equation exact (but still subject to the guided wave assumption, \( ka < 1 \)). In this case acoustic variables are defined as the average pressure and the corresponding volume velocity, each defined on the iso-pressure wave-front boundary (Webster, 1919; Hanna and Slepian, 1924).

**Solution methods:**

There are two distinct mathematical methods used to describe physical systems differential equations and lumped models.

1. The *scalar wave equation* describes the evolution of a scalar field, such as a pressure or voltage, or equivalently, the flow. The best known examples are electrical and acoustic transmission lines. Such systems are loosely referred to as the telegraph or telephone equations, referring back to early days of their discovery. In acoustics, guided waves are called horns, such as the horn connected to the first phonographs from around the turn of the century (Webster, 1919). Thus the names can reflect their historical development, with the mathematics and the applications frequently running in close parallel.

2. A system may also be represented in terms of lumped elements, such as electrical inductors, capacitors and resistors, and their mechanical counterparts, masses, springs and dash-pots. Such systems are represented by systems of matrix equations rather than differential equations, the size of which is equal to the number of elements in the network. When the system of lumped element networks contain only resistors and capacitors, they do not support waves, thus are related to the diffusion equation in their solution. Depending on the elements in the system of equations, there can be an overlap between a diffusion process and scalar waves, represented as transmission lines, both modeled as lumped networks of 2x2 matrices (Section 1.4.11, Eq. 1.70, p. 92).

3. When lumped elements are used, the equations approximate the transmission line equations below a cutoff frequency, which depends on frequency. When the wavelength is longer than the physical distance between the elements making of each matrix, the approximation is equivalent to the transmission line. But as the frequency in increased, the wavelength eventually becomes equal, and then shorter than the element spacing, and the lumped element model breaks down. This is under the control of the modeling process, as more elements are required to represent higher frequencies (shorter wavelengths). If the nature of the solution at high frequencies is desired, the lumped parameter model fails and one must use the differential equation method. However for many (perhaps most) problems, lumped elements are
Figure 1.33: Experimental setup showing a large pipe on the left terminating the wall containing a small hole with a balloon, shown in green. At time $t = 0$ the balloon is pricked and a pressure pulse is created. The baffle on the left is meant to represent a semi-$\infty$ long tube having a large radius compared to the horn input diameter $2a$, such that the acoustic admittance looking to the left ($A/\rho_0 c_0$ with $A \to \infty$), is very large compared to the horn’s throat admittance (Eq. 1.155). At time $T$ the outbound pressure pulse $p(r,T) = \delta(t - T/c_o)/r$ has reached radius $T/c_o$. Here $r = x + r_o$, where $x$ is the location of the source at the throat of the horn and $r$ is measured from the vertex.

easy to use, and accurate, for frequencies below the cutoff (where the wavelength approaches the element spacing). These relations are nicely explained in Brillouin (1953).

**Separable coordinate systems:** Classically PDEs are often solved by a technique called *separation of variables*, which is limited to a small number coordinate systems. By the use of the quasistatic approximation (P10, 105), the limitations of separation of variables may be avoided, at the price of a key assumption, *quasi-statics* (P10, p. 105). The most common examples of separable coordinate systems are rectangular, cylindrical and spherical coordinates (Morse, 1948, p. 296-7). Even a slight deviation from separable specific coordinate systems represents a major barrier to further understanding, blocking insight into the general case. These few separable coordinate systems are special cases, which have high degrees of symmetry, while the wave equation is not tied to a specific coordinate system. Quasi-statics can proved solutions over a much wider class of geometries.

When the pressure (i.e., a force potential) is not constrained to a “separable” coordinate system, the PDE is called the Sturm-Liouville equation, which is a 19 century, general description of “modern” Horn theory (Webster, 1919; Morse, 1948; Pierce, 1981). Horn theory add physics to the mathematical approach of Sturm-Louville, which does not attempt to identify the underlying physical principles. As is common in mathematical physics, it is the physical applications, not the mathematics, that makes a theory powerful. The mathematics provides the rigor, while physics is the soul of most theories. Both are critical, however the relative importance is open for discussion, and strongly depends on your point of view. I suspect most mathematicians would disagree with my view. Having an opinion drives science forward.

**Early history:** The study of wave propagation begins at least as early as Huygens (ca. 1678), followed soon after (ca. 1687) by Sir Isaac Newton’s calculation of the speed of sound (Pierce, 1981). To obtain a wave, one must include two basic components, the stiffness of air, and its mass. These two equations shall be denoted (1) Newton’s 2nd law ($F = ma$) and (2) Hooke’s law ($F = kx$), respectively. In vector form these equations are (1) Euler’s equation (i.e., conservation of momentum density)

$$-\nabla \rho(x,t) = \rho_0 \frac{\partial}{\partial t} u(x,t) \leftrightarrow \rho_0 s V(x,s),$$

(1.128)
assuming the density $\rho_0$ is independent of time and position $x$, and (2) the *continuity equation* (i.e., conservation of mass density)

$$-\nabla \cdot u(x, s) = \frac{1}{\eta_0 P_0} \frac{\partial}{\partial t} \rho(x, t) \leftrightarrow \frac{s}{\eta_0 P_0} \mathcal{P}(x, s) \quad (1.129)$$

(Pierce, 1981, page 15). Here $P_0 = 10^5 \text{ [Pa]}$ is the barometric pressure and $\eta_0 P_0$ is the dynamic (adiabatic) stiffness. Combining Eqs. 1.128 and 1.129 (removing $u(x, t)$) results in the 3-dimensional (3D) scalar pressure wave equation

$$\nabla^2 \rho(x, t) = \frac{1}{c_o^2} \frac{\partial^2}{\partial t^2} \rho(x, t) \leftrightarrow \frac{s^2}{c_o^2} \mathcal{P}(x, s), \quad (1.130)$$

where $c_o$ is the sound velocity.

**Exercise:** Show that Eqs. 1.128 and 1.129 can be reduced to Eq. 1.130. Take the divergence of Eq. 1.128 gives

$$-\nabla \cdot \nabla \rho(x, t) = \rho_0 \frac{\partial}{\partial t} \nabla \cdot u(x, t). \quad (1.131)$$

Note that $\nabla \cdot \nabla = \nabla^2$. Next substitute Eq. 1.129 into the above relation results in the scalar wave equation Eq. 1.130, since $c_o^2 = \eta_0 P_0 / \rho_0$.

**Eigen-solutions $\rho^{\pm}(r, t)$**

Because the wave equation (Eq. 1.130) is 2d order in time, there are two causal independent eigen-solutions of the homogeneous (i.e., un-driven) Webster horn equation, an *outbound* (right-traveling) and an *inbound* (left-traveling) wave, denoted as the Laplace transform pair

$$\rho^\pm(r, s) \leftrightarrow \mathcal{P}^\pm(r, s) = \int_0^\infty \rho^\pm(r, t) e^{-st} dt,$$

typically normalized so that $\mathcal{P}^\pm(r_o, s) = 1$, where $r_o$ defines the input excitation point. This construction is equivalent to a separation of time and space variables.

These two primitive solutions depend on $A(r)$, which causes local reflections with any area variations. Thus there is a tight relationship between the area change $dA(r) / dr$, internal resonances (eigen-modes), and the primitive wave solutions $\mathcal{P}^\pm(r, s)$. These relationships will be explored further in Chapter 5.

**Complex vs. real frequency:** We shall continue to maintain the distinction that functions of $\omega$ are Fourier transforms and functions of Laplace frequency $s$ correspond to Laplace transforms, which, because they are causal, are necessarily complex analytic in $s$ in the *right half plane* (RHP) *region of convergence* (ROC). This distinction is critical, since we typically describe impedance $Z(s)$ and admittance $Y(s)$, which must be causal, in terms of their poles and zeros, as complex analytic functions in $s$.\(^{92}\)

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\(^{92}\)When an analytic function of complex variable $s$ includes the pole it is called a *Laurent series* in $s$. For example, the impedance of a capacitor $C$ is $Z_c(s) = 1/sC$, which is analytic in $s$ everywhere other than $s = 0$. The capacitor has a voltage time response given by the integral of the current, i.e., $v(t) = \frac{1}{C} \int_0^t i(t) dt = \frac{1}{C} u(t) \ast i(t)$, where $u(t)$ is the Heaviside step function and $\ast$ represents convolution.
**Eigen-function solutions of the 1D wave equation:** As best we know, Huygen (1690) was the first to gain insight into wave propagation. While his concept, today known as “Huygens’ principle” showed a deep insight, it was seriously flawed, as it ignored the backward traveling wave (Miller, 1991). In 1747 a famous French intellectual and mathematician, Jean le Rond d’Alembert, published the general form of the solution to the 1D scalar wave equation

\[
\rho(x,t) = f(t - x/c_o) + g(t + x/c_o),
\]

(1.132)

where \(f(\cdot)\) and \(g(\cdot)\) are quite general functions of their argument. That this is the solution may be easily shown by use of the chain rule, by taking partials with respect to \(x\) and \(t\). Surprisingly, this is the solution even when the functions are not differentiable. For example \(u(t \mp x/c_o)\), and \(\delta(t \mp x/c_o)\), are valid solutions, even though the proof is more difficult.

**Exercise:** By the use of the chain rule, prove that d’Alembert’s formula satisfies the 1D wave equation. Taking a derivatives with respect to \(t\) and \(r\) give

- \(\partial_t \rho(r,t) = -c_o f'(r - c_o t) + c_o g'(r + c_o t)\)
- \(\partial_r \rho(r,t) = f'(r - c_o t) + g'(r + c_o t)\),

and a second derivatives gives

- \(\partial_{tt} \rho(r,t) = c_o^2 f''(r - c_o t) + c_o^2 g''(r + c_o t)\)
- \(\partial_{rr} \rho(r,t) = f''(r - c_o t) + g''(r + c_o t)\).

From these last two equations we have the 1D wave equation

\[
\partial_{rr} \rho(r,t) = \frac{1}{c_o^2} \partial_{tt} \rho(r,t),
\]

having solutions Eq. 1.132, in disagreement with Huygens’ 1690 “principle” (Miller, 1991).

In terms of the physics, d’Alembert’s general solution describes two arbitrary wave-forms \(f(\cdot)\), \(g(\cdot)\), traveling at a speed \(c_o\), one forward, and one reversed. Thus his solution is quite easily visualized. As previously discussed, Newton (1687) was the first to calculate the speed of sound \(c_o\), be it with an error of \(\sqrt{1.4}\). This error was not corrected for over 200 years, following the creation of thermodynamics, and the concept of an adiabatic process.

**Example:** Assuming \(f(\cdot), g(\cdot)\) are \(\delta(\cdot)\), find the Laplace transform of the solution. Using Table F.3 (p. 278) of Laplace Transforms on Eq. 1.132 gives

\[
\rho(x,t) = \delta(t - x/c_o) + \delta(t + x/c_o) \leftrightarrow e^{-sx/c_o} + e^{sx/c_o}.
\]

(1.133)

Note that the delay is \(T_o = \pm x/c_o\).
3D d’Alembert waves: The d’Alembert solution generalizes to 3D waves by changing the area function in Eq. 1.127 with the 3D area $A(r) = A_0 r$, as explained in Appendix 5.2.1, Eq. 5.11 (p. 237), resulting in the 3D wave equation

$$\nabla^2 r \rho(r, t) = \frac{1}{r} \frac{\partial}{\partial r} r \rho(r, t) = \frac{1}{c_o^2} \frac{\partial^2}{\partial t^2} \rho(r, t).$$

Multiplying by $r$ results in the general spherical (3D) d’Alembert wave equation solution

$$\rho(r, t) = \frac{f(t - r/c_o)}{r} + \frac{g(t + r/c_o)}{r},$$

for arbitrary wave-forms $f(·)$ and $g(·)$. These are the primitive wave solutions for the spherical scalar wave equation.

1.6.5 Related partial differential equations:

Historically the wave equation was seen to be related to several other important partial differential equations, as the theory evolved, resulting in the

1. **Fourier diffusion equation**

$$D_o \frac{\partial T(x, t)}{\partial t} = \nabla^2 T(x, t) \tag{1.134}$$

which describes, for example, the temperature $T(x, t)$ as proposed by Fourier in 1822, or for the diffusion of two miscible liquids (Fick, 1855) and Brownian motion (Einstein, 1905). Diffusion is not a wave.

2. **Poisson’s equation**

$$\nabla^2 \Phi(x, t) = \rho(x, t) \tag{1.135}$$

which holds for gravitational fields, or the voltage around a charge.

3. **Laplace’s equation**

$$\nabla^2 \Phi(x, t) = 0, \tag{1.136}$$

which describes, for example, the voltage inside a closed chamber with various voltages on the walls, or the steady state temperature within a closed container, given a specified temperature distribution on the walls.

Each of these four equations has properties that may be simply explained, and visualized, in simple geometries, and all contain the Laplacian $\nabla^2() = \nabla \cdot \nabla()$.

**The Laplacian $\nabla^2$:** We first discussed the Laplacian as a 2D operator in Section 1.5.2 (p. 110), when we studied complex analytic functions. Then an approximation for horns (variable tubes) was presented as Eq. 1.6.2. In 3D rectangular coordinates it is defined as

$$\nabla^2 T(x) = \left( \frac{\partial^2}{\partial x^2} + \frac{\partial^2}{\partial y^2} + \frac{\partial^2}{\partial z^2} \right) T(x). \tag{1.137}$$

In summary, the Laplacian operator is ubiquitous in mathematical physics, starting with simple complex analytic functions (Laplace’s equation), and progressing to the Poisson’s equation, the diffusion equation, and finally the wave equation. Only the wave equation expresses delay. The diffusion equation “wave” has an instantaneous spread (the effective “wave” velocity is infinite, i.e., its not a wave).
1.6.6 Lec 36: Properties of the Generalized impedance

Up to now the impedance has been a simple quantity, defined as the force over the flow

\[ \text{flow} = \frac{\text{force}}{\text{flow}}. \]

This works well for an electrical circuit \( Z(s) = -\nabla \Phi(s)/I(s) \), or a mechanical system, such as a mass \( Z(s) = msU(\omega)/\rho_o U(\omega) \), or heat circuit, such as a thermal conductivity \( \kappa \) \( Z_{\text{thermal}} = -T/\kappa \nabla T \). In general these are vector quantities that depend on space and time. This leads a generalized force and flow, with impedance as a matrix. But even in a 1-dimensional system we have not yet fully characterized impedance.

**Impedance:** Impedance plays a crucial role in the solution of differential equations, and in the theory and practice of complex variables. The concept broadly applies to electricity, mechanics, acoustics and heat (Table 1.19, p. 94). Impedance is one of the best examples of a complex analytic function, thus plays a key role in potential theory.

Theoretically, impedance plays the role of a boundary condition in the solution of differential equations, accounting for the reflectance of a wave at the boundary. It plays a critical role in conservation of energy, since if energy at the boundary (surface) must always be lost, given a passive boundary.

The first person to fully appreciate the importance of impedance at the boundary may have been Arnold Sommerfeld, who expressed this in what today is known as the **radiation impedance**, which is the impedance boundary condition as a point source wave radiates to \( |R| \to \infty \).

**Brune impedance:** As the concept of impedance evolved (it is a linear relation between a force and a flow), it was incorporated into more theories, such as electrical circuit theory and mechanics. The first persons to quantify the concept of impedance was Ohm, followed by Kirchhoff and Heaviside. Kennelly, not Heaviside, was the first to express the idea as a complex variable of a complex function of frequency (Kennelly, 1893). Perhaps Heaviside fully appreciated the concept, and has been given proper credit.

There are several important theorems here that follow from Brune's Theorem on **positive-real functions**, defined as

\[ \Re\{Z(\sigma > 0)\} \geq 0, \quad (1.138) \]

where \( Z(s) = R(s) + jX(s) \in \mathbb{C} \) is a Brune impedance having real part \( R(s) = \Re\{Z(s)\} \) and imaginary part \( X(s) = \Im\{Z(s)\} \), each as functions of the Laplace frequency \( s = \sigma + \omega j \) (Brune, 1931b; Van Valkenburg, 1964b). Thus a Brune impedance has a positive analytic real part in the right-half plane \( \sigma > 0 \). This condition has several non-obvious important ramifications.

1. \( Z(s) \leftrightarrow z(t) \) is real and causal
2. \( Y(s) = 1/Z(s) \leftrightarrow y(t) \) is real and causal
3. The phase \( \angle Z(\omega j) \) lies between \( \pm \pi/2 \) (i.e., \( Z(s) \) is minimum–phase)
4. All the poles and zeros are first degree (all the poles and zeros are strictly first degree)
5. If two poles approach each other, there must be a zero between them.
6. All poles and zeros lie in the left half plane LHP \( (\sigma \leq 0) \)

7. \( Z(\sigma > 0) \) is complex-analytic \([Z(s) \text{ obeys the Cauchy-Riemann conditions in the RHP } (\sigma > 0)]\)

8. All poles and zeros of \( Z(s) \) on the \( \omega j \) or on the \( \sigma \) axis must alternate \([\text{Foster’s Theorem (Van Valkenburg, 1964b)}]\)

9. All Brune impedances are quasi-static (lumped-parameter, where the size of the impedance element is much smaller than the wavelength).

10. The Brune impedance is a subset of impedance, where delay has been approximated away in the long-wavelength limit. For example, transmission lines are not in the class of Brune impedances.

In general it is difficult to test for Eq. 1.99 because it must be true everywhere in the RHP \((\sigma > 0)\). Once simple test we have access to today is \( zviz.m \). A plot of any given \( Z(s) \), and its reciprocal \( Y(s) \) must be positive in the RHP, which is easily visualized with \( zviz.m \).

A Brune impedance is defined as the ratio of the force \( F(\omega) \) over the flow \( U(\omega) \), and may be expressed in residue form as

\[
Z(s) = sL_o + R_o + \sum_{k=1}^{K} \frac{c_k}{s - s_k} = \frac{N(s)}{D(s)},
\]

(1.139)

known as the partial fraction expansion. The term \( L_o \) represents an inductance, and \( R_o \) a resistor. Coefficients \( c_k \) are the residues, and \( s_k \) the roots of \( D(s) \). It follows that

\[
D(s) = \prod_{k=1}^{K} (s - s_k) \quad \text{and} \quad c_k = \lim_{s \to s_k} (s - s_k)Z(s) = \prod_{n'=1}^{K-1} (s - s_n),
\]

where the prime on index \( n' \) means that \( n = k \) is not included in the product. Every Brune impedance may be expanded in a partial fraction expansion of the form of Eq. 1.139, and every partial fraction expansion may be converted back into the ratio of two polynomials, using the Matlab/Octave commands \([C, P, K]=\text{residue}(N, D)\) and \([N, D]=\text{residue}(C, P, K)\). These two representations are interchangeable.

Given a 2-port network, the input impedance and the transfer function share the same poles. In fact the transfer function must be all-pole.

Exercise: Find the Laplace transform \((\mathcal{L}T)\) of the three impedance relations in terms of the force \( F(s) \) and the velocity \( V(s) \), along with the electrical equivalent impedance: Each classical law is a linear relation

1. Hooke’s law \( f(t) = K x(t) \). Sol: First \( \mathcal{L}T \) Hooke’s law and then write it in terms of force and velocity

Taking the \( \mathcal{L}T \) gives

\[
F(s) = K X(s).
\]

\[\text{Is } C(s) \text{ of the ABCD matrix the same as the impedance denominator } D(s)\? \text{ Verify!}\]
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Since
\[ v(t) = \frac{d}{dt}x(t) \leftrightarrow V(s) = sX(s). \]

Thus the impedance of the spring is
\[ Z_s(s) = \frac{F(s)}{V(s)} = \frac{K}{s}, \]

which is analogous to the impedance of an electrical capacitor. If we specifying the compliance \( C \) of a spring as \( C = 1/K \), the relation looks just like the electrical case.

While Hooke’s law says the force and displacement are proportional, in terms of impedance variables force and flow,
\[ f(t) = K \int_0^t v(t) dt. \]

2. Dash-pot resistance \( f(t) = Rv(t) \). Sol: From the \( \mathcal{L} \mathcal{T} \) this becomes
\[ F(s) = RV(s) \]
and the impedance of the dash-pot is then
\[ Z_r = R, \]
analogous to that of an electrical resistor.

Exercise: If \( Z_r \) is the impedance of a resistor, find the formula time domain impedance. Sol: \( Z_r = R \Rightarrow z_r(t) = R\delta(t) \).

3. Newton’s 2d law for a constant mass \( M \): \( f(t) = Mdv(t)/dt \). Sol: Taking the \( \mathcal{L} \mathcal{T} \) gives
\[ F(s) = sMV(s) \]
thus
\[ Z_m(s) = \frac{F(s)}{V(s)} = sM, \]
analogous to an electrical inductor.

4. Find the total impedance seen by the net force \( f(t) \) applied to the mass \( M \). Sol: Summing the forces must equal the applied force, Eq. 1.81, p. 102.

5. Take the Laplace transform (\( \mathcal{L} \mathcal{T} \)) of Eq. 1.81 (p. 102), and evaluate the total impedance \( Z(s) \) of the mechanical circuit. Sol: From the properties of the \( \mathcal{L} \mathcal{T} \) that \( dx/dt \leftrightarrow sX(s) \), we find
\[ Ms^2X(s) + RsX(s) + KX(s) = F(s). \]
In terms of velocity this is \( (Ms + R + K/s)V(s) = F(s) \). Thus the circuit impedance is
\[ Z(s) = \frac{F}{V} = \frac{K + Rs}{s} + Ms^2. \]
6. What are $N(s)$ and $D(s)$ (e.g. Eq. 1.139, p. 147)? Sol: $D(s) = s$ and $N(s) = K + Rs + Ms^2$.

7. Assume that $M = R = K = 1$. Find the residues (e.g. Eq. 1.139, p. 147) of the admittance $Y(s) = 1/Z(s)$, in terms of the roots $s_{\pm}$. Check your answer with the Matlab/Octave command `residue`. Sol: First find the roots of the numerator of $Z(s)$ (the denominator of $Y(s)$):

$$s_{\pm}^2 + s_{\pm} + 1 = (s_{\pm} + 1/2)^2 + 3/4 = 0,$$

which is

$$s_{\pm} = \frac{-1 \pm j\sqrt{3}}{2}.$$

Next form the partial fraction expansion of the admittance

$$Y(s) = \frac{s}{1 + s + s^2} = c_0 + \frac{c_+}{s - s_+} + \frac{c_-}{s - s_-} = \frac{s(c_+ + c_-) - (c_+ s_- + c_- s_+)}{1 + s + s^2}.$$

Comparing the two sides requires that $c_0 = 0$. We also have two equations for the residues $c_+ + c_- = 1$ and $c_+ s_- + c_- s_+ = 0$. The best way to solve this is to set up a matrix relation and take the inverse

$$\begin{bmatrix} 1 & 1 \\ 1 & s_- \\ s_- & s_+ \end{bmatrix} \begin{bmatrix} c_+ \\ c_- \end{bmatrix} = \begin{bmatrix} 1 \\ 0 \end{bmatrix},$$

thus:

$$\begin{bmatrix} c_+ \\ c_- \end{bmatrix} = \begin{bmatrix} 1 & 1 \\ 1 & s_- - s_+ \end{bmatrix}^{-1} \begin{bmatrix} s_+ & -1 \\ s_- & -1 \end{bmatrix} \begin{bmatrix} 1 \\ 0 \end{bmatrix},$$

which gives $c_\pm = \pm \frac{s_+}{s_+ - s_-}$. The denominator is $s_+ - s_- = j\sqrt{3}$ and the numerator is $\pm 1 + j\sqrt{3}$. Thus

$$c_\pm = \pm \frac{s_\pm}{s_+ - s_-} = \frac{1}{2} \left( 1 \pm \frac{j}{\sqrt{3}} \right).$$

As always, finding the coefficients is always the most difficult part. Using 2x2 matrix algebra can really simplify up the process, and will more likely give the correct answer.

8. By applying the CRT, find the inverse Laplace transform ($LT^{-1}$). Use the residue form of the expression that you derived in the previous exercise. Sol:

$$z(t) = \frac{1}{2\pi j} \oint_C Z(s)e^{st}ds.$$

were $C$ is the Laplace contour which encloses the entire left-half $s$ plane. Applying the CRT

$$z(t) = c_+ e^{s_+ t} + c_- e^{s_- t},$$

where $s_{\pm} = -1/2 \pm j\sqrt{3}/2$ and $c_{\pm} = 1/2 \pm j/(2\sqrt{3})$.

### 1.6.7 Lec 37: Webster Horn Equation: Derivation

Here we transform the acoustic equations Eq. 1.128 and 1.129 (p. 142) into their equivalent integral form Eq. 1.127 (p. 140). This derivation is similar (but not identical) to that of Hanna and Slepian (1924); Pierce (1981, p. 360).
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Conservation of momentum: The first step is an integration the normal component of Eq. 1.128 over the pressure iso-surface $S$, defined by $\nabla p = 0$,

$$-\int_S \nabla p(x, t) \cdot dA = \rho_0 \frac{\partial}{\partial t} \int_S u(x, t) \cdot dA. \quad (1.140)$$

The average pressure $\bar{\rho}(x, t)$ is defined by dividing by the total area

$$\bar{\rho}(x, t) \equiv \frac{1}{A(x)} \int_S p(x, t) \hat{n} \cdot dA. \quad (1.141)$$

From the definition of the gradient operator

$$\nabla p = \frac{\partial p}{\partial x} \hat{n}, \quad (1.142)$$

where $\hat{n}$ is a unit vector perpendicular to the iso-pressure surface $S$. Thus the left side of Eq. 1.128 reduces to $\frac{\partial \bar{\rho}(x, t)}{\partial x}$.

The integral on the right side defines the volume velocity

$$\nu(x, t) \equiv \int_S u(x, t) \cdot dA. \quad (1.143)$$

Thus the integral form of Eq. 1.128 becomes

$$\frac{\partial}{\partial x} \bar{\rho}(x, t) = -\rho_0 \frac{\partial}{\partial t} \nu(x, t). \quad (1.144)$$

Conservation of mass: Integrating Eq. 1.129 over the volume $V$ gives

$$-\int_V \nabla \cdot u \, dV = \frac{1}{\eta_o P_o} \frac{\partial}{\partial t} \int_V p(x, t) \, dV.$$

Volume $V$ is defined by two iso-pressure surfaces between $x$ and $x + \delta x$ (Fig. 1.34, shaded blue). On the right hand side we use our definition for the average pressure (i.e., Eq. 1.141), integrated over the thickness $\delta x$.

Applying Gauss’ law to the left hand side, and using the definition of $\bar{\rho}$ (on the right), in the limit $\delta x \to 0$, gives

$$\frac{\partial \nu}{\partial x} = -\frac{A(x)}{\eta_o P_o} \frac{\partial \bar{\rho}}{\partial t} \nu(x, t). \quad (1.145)$$

Finally, writing Eq. 1.144 and Eq. 1.145 in matrix form, in the frequency domain ($\partial_t \leftrightarrow s$), results in Eq. 1.160 (p. 156).

Boundary Conditions: The pressure and velocity at any point $x$ be written in terms of a superposition of the two homogeneous solutions $P^+(x, s)$ and $P^-(x, s)$ (aka, primitive solutions) of Eq. 1.160 (p. 156), since by d’Alembert principle $V = V^+ - V^-$ and $P = P^+ + P^-$. The formula for the input admittance at any point $x$ on the line is given in terms of the forward and reverse primitive solutions

$$Y_{in}(r) \equiv \frac{Y(r, \omega)}{P(r, \omega)} = \frac{V^+ - V^-}{P^+ + P^-} = \frac{V_+}{P_+} \left( \frac{1 - V^- / V^+}{1 + P^- / P^+} \right) = \frac{\Gamma(1 - \Gamma(r, s))}{1 + \Gamma(r, s)}, \quad (1.146)$$

As shown in Fig. 1.34, we convert the divergence into the difference between two volume velocities, namely $\nu(x + \delta x) - \nu(x)$, and $\partial \nu/\partial x$ as the limit of this difference over $\delta x$, as $\delta x \to 0$.
Here the reflectance $\Gamma(r, s)$ is defined as the ratio of the forward and retrograde traveling waves

$$\Gamma(r, s) = \frac{\mathcal{V}^-(r, \omega)}{\mathcal{V}^+(r, \omega)} = \frac{\mathcal{P}^-(r, \omega)}{\mathcal{P}^+(r, \omega)}.$$

By a rearrangement of the above terms we may recover the definition of the characteristic admittance

$$\mathcal{Y}(r) = \frac{\mathcal{V}^+(r, \omega)}{\mathcal{P}^+(r, \omega)} = \frac{\mathcal{V}^-(r, \omega)}{\mathcal{P}^-(r, \omega)},$$

where $\mathcal{Y}(r) = \rho_o c_0 / A(r)$ for lossless horns.

It follows that in matrix notation d’Alembert’s principle may be written as

$$\begin{bmatrix} \mathcal{P}(x) \\ \mathcal{V}(x) \end{bmatrix} = \begin{bmatrix} 1 & 1 \\ \mathcal{Y}(x) & -\mathcal{Y}(x) \end{bmatrix} \begin{bmatrix} \mathcal{P}^+(x) \\ \mathcal{P}^-(x) \end{bmatrix}. \quad (1.147)$$

**Complex-analytic nature of $\Gamma(s)$ and $Z_{in}(s)$:** A very important assumption has been made here when expressing the complex reflectance $\Gamma(s)$ as a function of the complex frequency $s = \sigma + j\omega$, even though it is defined by the ratio of two functions of real (radian) frequency $\omega$. This assumption is based on the fact that, like the impedance, the reflectance must be causal. Namely $\gamma(t) \leftrightarrow \Gamma(s)$ is zero for $t < 0$. The same may be assumed of the time-domain impedance $\zeta(t) \leftrightarrow Z_{in}(s)$. That $\gamma(t)$ and $\zeta(t)$ are causal is require by the physics. It follows that both $\Gamma(s)$ and $Z_{in}(s)$ are complex analytic functions of $s$, which means they must have a Taylor series expansion in $s$ everywhere in the right-half $s$ plane ($\sigma > 0$). This follow from the Cauchy’s integral theorem (aka The fundamental theorem of complex calculus.), which is a special case of the Cauchy’s integral formula (aka, The Residue theorem.).

The forward and retrograde waves may be any function of frequency $\omega$, since they depend on the source pressure (or velocity) at the input to the horn. The reflectance is a transfer function (thus the source term cancels), that only depends on the impedance (or reflectance) looking into the system (at any position $x$).

As an alternative way to derive $\Gamma(r, s)$, invert Eq. 1.147

$$\frac{\mathcal{P}^+(x)}{\mathcal{P}^-(x)} = \frac{1}{2\mathcal{V}(x)} \begin{bmatrix} \mathcal{Y}(x) & 1 \\ \mathcal{Y}(x) & -1 \end{bmatrix} \begin{bmatrix} \mathcal{P} \\ \mathcal{V} \end{bmatrix} = \frac{1}{2} \begin{bmatrix} 1 & Z(x) \\ 1 & -Z(x) \end{bmatrix} \begin{bmatrix} \mathcal{P} \\ \mathcal{V} \end{bmatrix} = \frac{\mathcal{P} - Z\mathcal{V}}{\mathcal{P} + Z\mathcal{V}} \quad (1.148)$$

and form the ratio of reflected to incident waves

$$\Gamma(s) = \frac{\mathcal{P}^-}{\mathcal{P}^+} = \frac{\mathcal{P} - Z\mathcal{V}}{\mathcal{P} + Z\mathcal{V}} = \frac{Z_{in} - Z}{Z_{in} + Z}.$$
It is convenient to define the normalized input impedance \( Z_{in}/Z \) when working with this form of \( \Gamma(s) \).

Given some experience with these two forms, \( Z_{in} \equiv 1/Y_{in} \) and \( \Gamma(s) \), one may quickly begin to appreciate the advantage of working with the reflectance over the radiation impedance/admittance (aka immittance). The impedance has so many different forms, each of which are complicated, whereas the reflectance is easily understood, as it is closer to the physics.

**Finite length Horns:** For a horn of fixed length \( L \) these expressions may be rewritten in terms of re-normalized primitive waves. If we define the forward wave \( \mathcal{P}^+(x) \) as launched from \( x = 0 \) and the retrograde wave \( \mathcal{P}^-(x) \) as launched from \( x = L \), we may also write the pressure and velocity in terms of the primitives.

\[
\begin{bmatrix}
\mathcal{P}(x) \\
\mathcal{V}(x)
\end{bmatrix} = \begin{bmatrix}
\mathcal{P}^+(x) & \mathcal{P}^-(x - L) \\
\mathcal{V}^+(s)\mathcal{P}^+(x) & -\mathcal{V}^-(s)\mathcal{P}^-(x - L)
\end{bmatrix} \begin{bmatrix}
\alpha \\
\beta
\end{bmatrix}.
\]

(1.149)

The characteristic admittance \( \mathcal{Y}^\pm(x, s) \) depends on the range \( x \) and Laplace frequency \( s \). Coefficients \( \alpha(\omega) \) and \( \beta(\omega) \), which depend on frequency \( \omega \) (but not \( x \)), are determined by the boundary condition at \( x = L \). To find the four parameters \([A(s), B(s), C(s), D(s)]\) we evaluate the inverse of Eq. (1.149) at \( x = L \), substitute this result into 1.149, and then evaluated the matrix product at \( x = 0 \).

For compactness we adopt the following simplified subscript notation: \( \mathcal{P}_o^\pm \equiv \mathcal{P}^\pm(x = 0, s) \), \( \mathcal{P}_L^\pm \equiv \mathcal{P}^\pm(x = L, s) \), i.e., \( \mathcal{P}_L \equiv \mathcal{P}(x = L) \) and \( \mathcal{V}_L \equiv \mathcal{V}(x = L) \). The normalization of the primitive solutions are taken as \( \mathcal{P}_o^+ = 1 \) and \( \mathcal{P}_o^- = 1 \).

To compute \( \Delta_L \) we start with

\[-\Delta_T(x, s) = \mathcal{P}^+(x) \mathcal{Y}^-\mathcal{P}^-(x - L) + \mathcal{P}^-(x - L) \mathcal{Y}^\mathcal{P}^+(x) \mathcal{P}^-(x) \]

evaluated at \( x = L \) (recall \( \mathcal{Y}^-\mathcal{P}^+(x = L) = \mathcal{P}^+_L = 1 \)),

\[-\Delta_L = \mathcal{Y}^-\mathcal{P}_L^+ \mathcal{P}_o^+ + \mathcal{P}_o^- \mathcal{P}_L^- ,
\]

thus

\[-\Delta_L = \mathcal{Y}^-\mathcal{P}_L^+ \mathcal{P}_o^- + \mathcal{P}_o^- \mathcal{P}_L^- ,
\]

(1.150)

\[
\begin{bmatrix}
\alpha \\
\beta
\end{bmatrix} = -\frac{1}{\Delta_L} \begin{bmatrix}
\mathcal{Y}^-\mathcal{P}^-(x - L) & \mathcal{P}^-(x - L) \\
\mathcal{Y}^\mathcal{P}^+(x) & -\mathcal{P}^+(x)
\end{bmatrix} \begin{bmatrix}
\mathcal{P}_L \\
\mathcal{V}_L
\end{bmatrix}.
\]

(1.151)

This detailed set of computations results in the following:

\[
\begin{bmatrix}
\mathcal{P}_o \\
\mathcal{V}_o
\end{bmatrix} = -\frac{1}{\Delta_L} \begin{bmatrix}
1 & \mathcal{P}_L^- \mathcal{P}_o^- \mathcal{P}_L^+ \\
\mathcal{Y}^-\mathcal{P}^+_o & -\mathcal{Y}^-\mathcal{P}^+_o \mathcal{Y}^-\mathcal{P}^-\mathcal{P}_L^+
\end{bmatrix} \begin{bmatrix}
\mathcal{P}_L \\
\mathcal{V}_L
\end{bmatrix}.
\]

(1.152)

The subscript to the right of each matrix indicates it is evaluated at \( x = 0 \) or \( x = L \). Here \( \mathcal{P}_L^- \) is \( \mathcal{P}^-(x - L) \) at \( x = 0 \). The sign of \( \mathcal{V}_L \) must be negative to satisfy the definition of every ABCD matrix, that the output velocity (i.e., \( -\mathcal{V}_L \)) is out of the port.
The relationship of \( \beta/\alpha \) has special significance because it specifies the ratio of the reflected wave amplitude \( \beta(\omega) \) in terms of the incident wave amplitude \( \alpha(\omega) \). This ratio is known as the reflectance

\[
\gamma_L(t) \leftrightarrow \Gamma_L(s) = \frac{\beta}{\alpha}. \tag{1.153}
\]

It has a critical role in the theory of horns, as we shall see as it is determined by the relative rate of change of the impedance (i.e., area) with range (i.e., \( d\ln(Z)/dx \)).

**Examples:** Given the equation for the two d’Alembert pressure waves \( P^\pm(x,s) \) for the case of the uniform horn (Eq. 1.133, p. 144), with \( \kappa(s) = s/c_o \), the volume velocity is

\[
\mathcal{V}^\pm(x,\omega) = \mp \frac{1}{2}(x,s) \frac{d}{dx} e^{\pm \kappa(s)x}
\]

\[
= \mp \frac{\kappa(s)}{2} P^\pm(x,s)
\]

\[
= \mathcal{Y}_o P^\pm(x,s)
\]

\[
= \frac{A_o}{\rho_o c_o} P^\pm(x,s).
\]

Thus

\[
\mathcal{Y}^\pm = \frac{A_o}{\rho_o c_o} > 0 \tag{1.154}
\]

is the characteristic admittance of the uniform horn, and independent of the wave direction. This is the simplest case.

For the spherical geometry Eq. 1.6.4 (p. 145):

\[
\mathcal{Y}^\pm(r,\omega) = -\frac{1}{2}(r,s) \frac{d}{dr} \frac{1}{r} e^{\mp \kappa(s)r}
\]

\[
= -\frac{1}{2}(r,s) \left[ \mp \kappa(s) \frac{e^{\mp \kappa(s)r}}{r} - \frac{e^{\mp \kappa(s)r}}{r^2} \right]
\]

\[
= \frac{1}{2}(r,s) \left[ \pm \kappa(s) \frac{e^{\mp \kappa(s)r}}{r} + \frac{e^{\mp \kappa(s)r}}{r^2} \right]
\]

\[
= \pm \frac{1}{2}(r,s) \left[ \kappa(s) P^\pm \pm \frac{1}{r} P^\pm \right]
\]

\[
= \pm \frac{1}{2}(r,s) \left[ \kappa(s) \pm \frac{1}{r} \right] P^\pm(r,s)
\]

\[
= \frac{\kappa(s)}{2}(r,s) \left[ 1 \pm \frac{1}{\kappa(s)r} \right] P^\pm(r,s)
\]

\[
= \mathcal{Y}(r) \left[ 1 \pm \frac{1}{\kappa(s)r} \right] P^\pm(r,s),
\]

where \( A(r) = A_o(r/a)^2 \), \( a \) is the radius at the mouth, \( \mathcal{Y}(r) = A_o r^2/\rho_o c_o \) and \( \kappa(s) = s/c_o \). Thus the input admittance is

\[
\mathcal{Y}^\pm_{in}(r,s) = \frac{A_o r^2}{\rho_o c_o} \frac{1 \pm \frac{c_o}{sr}}{a^2 \rho_o c_o} \left[ \delta(t) \mp \frac{c_o}{r} u(t) \right]. \tag{1.155}
\]
Since the input impedance is the reciprocal of the admittance,
\[
Z_{in}^\pm (r, s) = \frac{\rho_o c_o}{A_o r^2} \frac{1}{\pm e^{x/s}} = \frac{\rho_o c_o}{A(r)} \frac{sr/c_o}{\pm e^{x} + 1} \leftrightarrow \frac{\rho_o c_o}{A(r)} e^{\mp rt/c_o} u(t). \tag{1.156}
\]

Note that both \(\Re\{Y_{in}\} = \mathcal{Y} > 0\) and \(\Re\{Z_{in}\} \geq 0\), as is physically required.

Writing \(Y_{in}(r, s)\) in terms of \(\ln P^\pm\) greatly simplifies the algebra, and thus provides improved insight into the properties of the radiation admittance. For example, for the spherical wave,
\[
\frac{\partial \ln P^\pm}{\partial r} = \frac{\partial}{\partial r} (\mp \kappa(s) r - \ln r) = \mp \kappa(s) - \frac{1}{r}. \leftrightarrow \mp \tag{1.157}
\]

Frequently the velocity and pressure are specified as functions of frequency \(\omega\), not complex frequency \(s = \sigma + \omega j\). However since \(Y_{in}(x, s)\) must be causal and be non-negative, we indicate it as a function of \(s\). Since the primitive solutions must be causal and stable functions, they also must be analytic functions of \(s\) for \(\sigma > 0\).

Summary of four classic horns: Table 1.4 is a summary of the properties for four different cases of \(A(r)\) for the uniform, parabolic, conical and exponential horns.

Table 1.4: Table of horns and their properties for 1, 2 and 3 dimensions, and the exponential horn. In this table the horn’s range variable is \(x\), having area \(A(x)\), radius \(r_o(x) = \sqrt{A(x)/\pi}\). \(F(x)\) is the coefficient on \(P_o\) (Eq. 5.12, p. 237), \(\kappa(s) = s/c_o\), where \(c_o\) is the speed of sound. A dimensionless range variable is defined as \(x = (\xi - \xi_o)/(L - \xi_o)\), with \(\xi\) the linear distance along the horn axis, from \(x = \xi_o\) to \(L\) corresponding to \(x = 0\) to 1. The horn’s primitive solutions are \(P^\pm(x, \omega) \leftrightarrow \phi^\pm(x, t)\). When \(\pm\) is indicated, the outbound solution corresponds to the negative sign. The last column is the radiation admittance normalized by \(A(x)/\rho c_o\) where \(s = \sigma + \omega j\) is the Laplace frequency.

<table>
<thead>
<tr>
<th>#D</th>
<th>Name</th>
<th>radius</th>
<th>Area/A_o</th>
<th>(F(x))</th>
<th>(P^\pm(x, s))</th>
<th>(\phi^\pm(x, t))</th>
<th>(Y_{in}^\pm/\mathcal{Y})</th>
</tr>
</thead>
<tbody>
<tr>
<td>1D</td>
<td>plane</td>
<td>1</td>
<td>1</td>
<td>0</td>
<td>(e^{\mp \kappa(s)x})</td>
<td>(\delta(t \mp x/c))</td>
<td>1</td>
</tr>
<tr>
<td>2D</td>
<td>parabolic</td>
<td>(x/x_o)</td>
<td>(x/x_o)</td>
<td>(1/x)</td>
<td>(H^\pm_o(-j \kappa(s)x))</td>
<td>(\delta(t \mp x/c)/x)</td>
<td>(-jxH^\pm_o/H^\pm_o)</td>
</tr>
<tr>
<td>3D</td>
<td>conical</td>
<td>(x)</td>
<td>(x^2)</td>
<td>(2/x)</td>
<td>(e^{\mp \kappa(s)x}/x)</td>
<td>(\delta(t \mp x/c)/x)</td>
<td>(1 \pm c/sx)</td>
</tr>
<tr>
<td>EXP</td>
<td>exponential</td>
<td>(e^{mx})</td>
<td>(e^{2mx})</td>
<td>(2m)</td>
<td>(e^{-(m \pm \sqrt{m^2 + \kappa^2})x})</td>
<td>(e^{-mx}E(t))</td>
<td>—</td>
</tr>
</tbody>
</table>

The quasi-static approximation: Since the velocity perpendicular to the walls of the horn must be zero, any radial wave propagation is exponentially attenuated \((\kappa(s)\) is real and negative, i.e., Eq. 1.167 will not describe radial wave propagation), with a space constant of about 1 diameter. The assumption that these radial waves can be ignored (i.e., more than 1 diameter from their source) is called the quasi-static approximation. As the frequency is increased, once \(f \geq f_c = 2c_o/\lambda\), the radial wave can satisfies the zero normal velocity wall boundary condition, and therefore will not be attenuated. Thus above this critical frequency, radial waves (aka, higher order modes) are supported \((\kappa\) becomes imaginary). Thus for Eq. 1.127 to describe guided wave propagation, \(f < f_c\). But even under this condition, the solution will not be precise within a diameter (or so) of any discontinuities (i.e., rapid variations) in the area.

Each horn, as determined by the area function \(A(r)\), has a distinct wave equation, and thus a distinct solution. Note that the area function determines the upper cutoff frequency via the quasi-static approximation since \(f_c = c_0/\lambda_c, \lambda_c/2 > d, A(r) = \pi(d/2)^2\). Thus to satisfy the quasi-static
approximation, the frequency $f$ must be less than the cutoff frequency

$$f < f_c(r) = \frac{c_o}{4} \sqrt{\frac{\pi}{A(r)}} \quad (1.158)$$

We shall discuss two alternative matrix formulations of these equations, the ABCD transmission matrix, use for computation, and the impedance matrix, used when working with experimental measurements (Pierce, 1981, Chapter 7). For each formulation reciprocity and reversibility show up as different matrix symmetries, as addressed in Section 1.4.17 (p. 103) (Pierce, 1981, p. 195-203).

**Finite horns:** When a finite section of horn is terminated in a load impedance, the forward wave $\mathcal{P}^+(r, s)$ is reflected at the load, producing a backward traveling wave $\mathcal{P}^-(r, s)$. The way this is treated is by introducing boundary conditions at the termination, a key concept that needs an explanation.

**Exercise:** Prove that $Y^\pm(r)$ is independent of frequency even when Eq. 1.160 has losses. **Sol:**

This proof rests on the fact that $\gamma(t) \leftrightarrow \Gamma(s)$ is strictly causal, that is $\gamma(t) = 0$ for $t \leq 0$. Since $\Gamma(s)$ is complex analytic, it is causal (i.e., $\gamma(t < 0) = 0$). Thus we need to look at $t = 0$. The magnitude of any delta function term in $\gamma(t) \leftrightarrow \Gamma(s)$, that is

$$\gamma(t) = \gamma_0 \delta(t) + \tilde{\gamma}(t)$$

is zero. Lundberg et al. (2007) have shown that

$$\gamma_0 = \gamma(0^+) - \gamma(0^-). \quad (1.159)$$

For any causal (complex analytic) function, $\gamma(0^-) = 0$. Furthermore from the initial-value theorem,

$$\gamma(0^+) = \lim_{\sigma \to \infty} s \hat{F}(s)$$

where $\hat{F}(s)$ is that part of $F(s)$ which converges to $\gamma(0^+)$ as $\sigma \to \infty$.

But since $\Gamma(s)$ is complex analytic and bounded in magnitude ($|\Gamma(\omega)| \leq 1$), this limit must be zero. This proof is similar to that of Brune (1931a) for positive real impedances.

Imagine a function that is complex analytic for $\sigma \geq 1$, that is bounded by 1 on the $\sigma = 0$ axis. Since it is causal, the condition of Eq. 1.159 must be true. The corresponding physical argument is interesting. When a plane wave hits a junction, the only reflection is due to the change in impedance. If there is no change, then there is no reflection. If the change is only due to the losses, then there must be a delay in that reflection, which physically means $\gamma(0) = 0$.

While such cases are interesting, and realistic, they lie outside the traditional loss-less Webster horn equation formulation.

Newton’s conservation of momentum law (Eq. 1.128, p. 142), along with conservation of mass (Eq. 1.129, p. 143), are modern versions of Newton’s starting point for calculating the speed of sound. Following the simplification of averaging the normal component of the particle velocity over the iso-pressure wave front (defining the volume velocity), the two equations may be written as a 2x2 matrix (Ramo et al., 1965) in terms of acoustic variables average pressure $\mathcal{P}(r, \omega)$ and
volume velocity $\mathcal{V}(r, \omega)$, sometimes referred to as conjugate variables.\footnote{\url{https://en.wikipedia.org/wiki/Conjugate_variables_(thermodynamics)} The product of conjugate variables defines an intensity, and their ratio, an impedance (Pierce, 1981, p. 37-41).} 

$$\frac{d}{dr} \left[ \mathcal{P}(r, \omega) \right] = - \begin{bmatrix} 0 & \mathcal{Z}(s, r) \\ \mathcal{Y}(s, r) & 0 \end{bmatrix} \begin{bmatrix} \mathcal{P}(r, \omega) \\ \mathcal{V}(r, \omega) \end{bmatrix}. \tag{1.160}$$

This key transformation, from particle to volume velocity, is discussed in Section ?? (Eqs. 1.144,1.145, p. ??). The Fourier-transform pair of the average pressure and volume velocity are denoted $\mathcal{P}(r, t) \leftrightarrow \mathcal{P}(r, \omega)$ and $\mathcal{V}(r, t) \leftrightarrow \mathcal{V}(r, s)$\footnote{Notation: Lower case variables (i.e., $g(r, t), \nu(r, t)$) denote time-domain functions while upper case letters (i.e., $\mathcal{P}(r, \omega), \mathcal{V}(r, \omega), \mathcal{Z}(r, s), \mathcal{Y}(r, s)$) indicate frequency domain Fourier ($\omega$) or Laplace ($s$) transforms, the latter being analytic in $s = \sigma + j\omega$ for $\sigma > 0$. An exception to this notional rule is $\kappa(s)$.}. The complex Laplace frequency $s = \sigma + j\omega$ is required when defining the per-unit-length impedance

$$\mathcal{Z}(s, r) \equiv \frac{s \rho_o}{A(r)} = sM(r) \tag{1.161}$$

and per-unit-length admittance

$$\mathcal{Y}(s, r) \equiv \frac{sA(r)}{\eta_oP_o} = sC(r), \tag{1.162}$$

to indicate that these functions are causal, and except at their poles, analytic in $s$\footnote{A function $F(s)$ is said to be complex analytic in $s = \sigma + \omega j$ at point $\alpha$ if it may be represented by a convergent Taylor series in $s$ the neighborhood of $s = \alpha$. That is if $F(s) = \sum_{n=0}^{\infty} F^{(n)}(\alpha)(s-\alpha)^n$, where $F^{(n)}(\alpha) = \frac{d^n F(s)}{ds^n}\bigg|_{s=\alpha}$.}. Here $M(r) = \rho_o/A(r)$ is the horn’s per-unit-length mass, $C(r) = A(r)/\eta_oP_o$ per-unit-length compliance, $\rho_o$ is the density of air (i.e., 1.2 [kg/m$^3$]), $P_o$ is the atmospheric pressure (i.e., 10$^5$ [N/m$^2$]), $\eta_o = c_p/c_v \approx 1.4$ (air). It is helpful to express the bulk properties of the waves in terms of $\mathcal{Z}$ and $\mathcal{Y}$, which is provided next. Equation 1.127 gives the full pressure wave equation when Eqs. 1.161 and 1.162 are reduced to a single equation for the pressure.

Exercise: Show that Eq. 1.162 is the same as the 2d order equation in the pressure results in the traditional Webster Horn equation

$$\nabla^2 \mathcal{P}(r, \omega) \equiv \frac{1}{A(r)} \frac{\partial}{\partial r} A(r) \frac{\partial}{\partial r} \mathcal{P}(r, \omega) = \frac{s^2}{c_o^2} \mathcal{P}(r, \omega). \tag{1.163}$$

Namely the Laplacian in the range variable $r$ depends on the area function $A(r)$, which in turn depends on the geometry of the horn. For example, for a uniform horn, $A(r) = A_o$, and for a conical horn, $A(r) = A_o r^2$. These cases are specifically dealt with in the next section.

Two fundamental parameters fall out of this formulation, the speed of sound and the characteristic admittance and impedance.

Bulk acoustic parameters: The bulk acoustic parameters are the free-field speed of sound

$$c_o = \sqrt{\frac{\eta_o P_o}{\rho_o}} = \sqrt{\frac{\text{stiffness}}{\text{mass}}}, \tag{1.164}$$

the specific acoustic resistance is

$$\rho_o c_o = \sqrt{\rho_o \eta_o P_o} = \sqrt{\text{stiffness-mass}}. \tag{1.165}$$
Characteristic admittance $\mathcal{Y}(r)$ and impedance $\mathcal{Z}(r)$: An important definition is each primitive wave’s characteristic admittance $\mathcal{Y}(r)$, defined as the square root of the ratio of $\mathcal{Y}$ over $\mathcal{Z}$ (Eq. 2.8, p. 189 and Fig. 2.2, p. 189),

$$\mathcal{Y}^{\pm}(r) = \sqrt{\frac{\mathcal{Y}(r,s)}{\mathcal{Z}(r,s)}} = \sqrt{sA(r)A(r)} \frac{A(r)}{\rho_o c} > 0,$$

(1.166)

which depends specifically on $A(r)$, but not on frequency $s$ (Campbell, 1903, 1910, 1922). Based on physical requirements that the admittance must be positive, only the positive square root is allowed. The characteristic impedance $\mathcal{Z}(r) = 1/\mathcal{Y}(r)$.

Since the horn (Eq. 1.127, p. 140) is lossless, $\mathcal{Y}^{\pm}(r)$ must be real (and positive). If losses are introduced, $\kappa(s)$ and $\mathcal{Y}(r,s)$ can become complex analytic functions of the Laplace frequency $s$ (Kirchhoff, 1974; Mason, 1928; Ramo et al., 1965; Pierce, 1981, p. 532-4). This more difficult case is addressed in Section 5 (p. 233).

Propagation function $\kappa(s)$: The primitive solutions of the horn equation depend on the complex wave propagation function $\kappa(s)$, defined as the square root of the product of $\mathcal{Z}$ and $\mathcal{Y}$ (Eq. 2.7, p. 189 and Eq. 5.15, p. 239):

$$\kappa(s) \equiv \sqrt{\mathcal{Z}(r,s)\mathcal{Y}(r,s)} = \sqrt{\frac{sA(r)}{A(r)}} \frac{sA(r)}{\eta_o P_o} = \frac{s}{c_o}.$$

(1.167)

Note that since the area $A(r)$ cancels in this expression, $c_o$ is independent of the range $r$. While horns are generally dispersive, namely the solution can have a group velocity (and thus a group delay) that depends on frequency, the wave-front speed is always constant, defined by the free-field sound velocity $c_o$. These relations between the phase, group and wave-front velocities (delay) were first demonstrated between 1907–1914 by Brillouin (1960, English version), while working with Arnold Sommerfeld (Pleshko and Pla`cz, 1969).

Horn input admittance: A horn’s acoustic input admittance $Y_{in}(r,s)$ is the admittance looking into a horn (Fig. 1.33, p. 142), at any point $r$. It is defined as the ratio of the volume velocity over the average pressure

$$Y_{in}(r,s) \equiv \frac{V(r,s)}{P(r,s)},$$

(1.168)

where the direction of flow $V(r,s)$ is into the admittance. $Y_{in}(s)$ depends on the entire horn, and any terminating admittance, in the case of a finite length horn.

The radiation admittance $Y_{rad}(s)$ is defined for an infinite horn, when there is no terminating load, as

$$Y_{rad}(s) = \lim_{r \to \infty} Y_{in}(r,s).$$

(1.169)

This may be viewed as the horn’s termination at $r = \infty$.

It is a physical requirement that the real part of every physical admittance (impedance) be positive (non-negative). To guarantee this physical requirement, care must be taken when fixing

---

98 The wave propagation function is commonly called the wave-number (Sommerfeld, 1952, p. 152) or simply the propagation constant. However since $\kappa(s)$ is neither a number, nor constant, we have appropriately rename it. The meaning of a complex $\kappa(s)$ is addressed in (Sommerfeld, 1952, p. 154). The units of $\kappa(s)$ are reciprocal length, namely $\kappa(s) = 2\pi/\lambda(s)$. 
the flow direction. Thus the velocity (or current) for any admittance must be defined as directed into the admittance, or else the real part will automatically be negative.

From Eq. 1.160 (p. 156) (top equation), the volume velocity is

\[
V(r, \omega) = -\frac{1}{Z(r,s)} \frac{d}{dr} P(r, \omega).
\]

(1.170)

Note that \( V \) is always defined looking into the horn, to guarantee that the resistance must be positive, (i.e., \( \Re Y_{in}(s) \geq 0 \)), as required by the physics (Postulate P3, Section 3.5.1, p. 103).

### 1.6.8 General 1D solution of the horn equation

In the mathematical literature, equations of this form are known as Sturm-Liouville problems, the solution of which may be obtained using integration by parts, following a multiplication by an “integration factor” \( \sigma(r) \):

\[
\frac{1}{\sigma(r)} \partial_r [\sigma(r) P_r] = \partial_{rr} + \left\{ \frac{1}{\sigma(r)} \partial_{rr} \right\} \partial_r P(r, s).
\]

(1.171)

**Exercise:** Write out the above using standard notation (i.e., use the chain rule). **Sol:**

\[
\frac{1}{\sigma(r)} \partial \left\{ \sigma(r) \partial_r P(r, s) \right\} \equiv \frac{\partial^2}{\partial r^2} P(r, s) + \left\{ \frac{1}{\sigma(r)} \partial_{rr} \right\} \partial_r P(r, s).
\]

By a direct comparison of the Eq. 1.171 and Eq. 5.1 (p. 233), we see that \( \sigma_r/\sigma \equiv -Z_r/Z \). From the definition of \( Z(r,s) = s\rho_0/A(r) \) (Eq. 1.161) it follows that

\[
\sigma(r) \equiv A(r).
\]

(1.172)

Thus one may conclude that the integration factor is physically the area function. This explains why the integration factor in the Sturm-Liouville theory must be strictly positive.\(^{99}\) Thus the Webster horn equation Eq. 1.127 (Morse, 1948, p. 269) directly follows from Eq. 1.127, due to Eq. 1.172.

In summary, if one starts with the 3D wave equation, and by the application of Gauss’ law, transforms it into a “1-dimensional” (i.e., single range variable) equivalent horn equation, then a Sturm-Liouville equation (a generalized scalar differential equation) results. This may be integrated by the application of the area function as the integration factor resulting in the Webster horn equation

\[
\frac{1}{A(r)} \partial_r \left( A(r) \partial_r P(r, s) \right) = \frac{s^2 \rho}{c^2} P(r, s), \leftrightarrow \frac{1}{c^2} \partial_{tt} \rho
\]

(1.173)

### 1.6.9 Webster horn equation

The Webster horn equation was first proposed in by Webster in 1919. It is a special but important case of the Strum-Liouville equation, where \( p(x) \) is the area of the horn. (Webster, 1919).

In 1946 Salmon formalized a special case, by transforming Eq. 1.127 to the Sturm-Liouville equation, which has constant coefficients, thereby making it is integrable. Here we discuss this special class of Salmon Horns (Salmon, 1946a,b).

\(^{99}\)We are not aware of this observation in the existing literature. It seems likely that Morse (1948) must have been aware of this connection, however to our knowledge he did explicitly say so.
Exercise:

1. Transform Eq. 1.127 by assuming an effective pressure \( E(x) = p(x)/\sqrt{A} \) (Salmon, 1946a, Eq. 2.2), resulting in Salmon’s Eq. 2.3

\[
\frac{d^2 E}{dx^2} + \left( k^2 - \frac{1}{\sqrt{A(x)}} \frac{d^2 \sqrt{A(x)}}{dx^2} \right) E = 0. \tag{1.174}
\]

This transformation assumes a constant energy \( A(x)|P|^2 \), in dependent of position \( x \), given that the effective radius is \( R(x) \equiv \sqrt{A} \) (Morse, 1948, p. 270). **Sol:** If we substitute \( P = E/\sqrt{A} \) into Eq. 1.127 we obtain

\[
-k^2 \frac{E}{\sqrt{A}} = \frac{1}{A} \frac{d}{dx} \left[ A \frac{d}{dx} \frac{E}{\sqrt{A}} \right].
\]

Expanding within the brackets, and canceling terms, gives

\[
-k^2 E = \frac{d^2 E}{dx^2} - \frac{E}{\sqrt{A}} \frac{d^2 \sqrt{A}}{dx^2},
\]

which is the same as Eq. 1.174.

2. Show that the equation is equivalent to the formula given by Pierce page 360, (7-8.5a)

\[
\left[ \frac{\partial^2}{\partial x^2} - m^2 - \frac{1}{c^2} \frac{\partial^2}{\partial t^2} \right] A(x)^{\frac{1}{2}} p(x, t) = 0
\]

with \( m \) a constant

\[
-m^2 = \frac{1}{4A^2} \left[ (A')^2 - 2AA'' \right].
\]

**Sol:** First Fourier transform this equation and express in terms of \( E(x, k) \):

\[
\left[ \frac{\partial^2}{\partial x^2} + k^2 - m^2 \right] E(x, k) = 0.
\]

This follows from the relation

\[
m^2 = \frac{1}{\sqrt{A(x)}} \frac{d^2 \sqrt{A(x)}}{dx^2} = \frac{1}{4A^2} \left[ 2AA'' - (A')^2 \right].
\]

3. Show that if \( A(x) = A_0 e^{\pm 2mx} \), \( m \) is constant. **Hint:** Use \( (\ln f(x))' = f'/f \). **Sol:** Assume that \( m \) is a constant and then substitute \( A(x) = A_0 e^{\pm 2mx} \) into

\[
\frac{1}{4A^2} \left[ (A')^2 - 2AA'' \right] = \left( \frac{A'}{2A} \right)^2 - \frac{A''}{2A} = m^2 \pm \frac{2m}{2A} \left( \frac{d}{dx} e^{\pm 2mx} \right) = m^2 \mp m (\pm 2m) = m^2 \mp m^2.
\]

Not surprisingly, it follows that when \( m \) is a constant, the relationship holds.
4. If \( m = 1 \) what is \( A(x) \)?

\[
\text{Sol: Based on the equation for the exponential horn: If } A(x) = A_0 e^{\pm 2mx} \text{ the WHEN is a constant coefficient equation. Thus if } m = 1, \ A(x) = A_0 e^{\pm 2x}.
\]

There is a more general solution of the form
\[
A(x) = a e^{-2x} \left(b + e^{2x}\right)^2
\]
where \( a, b \) are consonants. This example is a special case of the more general Fourier transform relationship of the area function
\[
A(x) = \int_{-\infty}^{\infty} A(k)e^{2x}dk
\]
(1.175)

Alternatively, as discussed by Salmon (1946b), expressing the area as an analytic function
\[
A(x) = \sum_{k=0}^{\infty} a_k x^k.
\]

### 1.6.10 Lec 38 Three examples of horns

#### The uniform horn

The 1D wave equation \([A(r) = A_o]\)
\[
\frac{d^2}{dr^2}P = \frac{s^2}{c^2_o}P.
\]

**Solutions:** The two primitive solutions of this equation are the two d’Alembert waves (Eq. 1.95, p. 112)
\[
\phi(x,t) = \phi^+(t - x/c) + \phi^-(t + x/c) \leftrightarrow e^{-\kappa x} + e^{\kappa x},
\]
where \( \mathcal{P}^\pm(r_o, s) \) are Laplace transform pairs representing the causal forward and retrograde traveling wave pressure amplitudes. It is convenient to normalize \( \mathcal{P}^\pm(r_o, s) = 1 \). Doing so gives the normalized primary solutions
\[
\phi^+(x,t) = \delta(t - x/c_o) \leftrightarrow \mathcal{P}^+(r_o, s)e^{-\kappa x}
\]
and
\[
\phi^-(x,t) = \delta(t + x/c_o) \leftrightarrow \mathcal{P}^-(r_o, s)e^{\kappa x}.
\]

As before, separation of time and place variables explains this relationship.

The characteristic admittance (Eq. 1.166) is independent of direction. The signs must be “physically chosen,” with the velocity \( \mathcal{Y}^\pm \) *into* the port, to assure that \( \mathcal{Y} > 0 \), for both waves.

**Applying the boundary conditions:** The general solution is compactly formulated as an ABCD matrix (i.e., Section 1.4.11, p. 92), starting from
\[
\begin{bmatrix}
\mathcal{P}(x) \\
\mathcal{V}(x)
\end{bmatrix} =
\begin{bmatrix}
e^{-\kappa x} & e^{\kappa x} \\
\mathcal{Y} e^{-\kappa x} & -\mathcal{Y} e^{\kappa x}
\end{bmatrix}
\begin{bmatrix}
\alpha \\
\beta
\end{bmatrix}_x,
\]
(1.176)
where \( \alpha, \beta \) are the relative weights on the two unknown primitive solutions that must be determined by the boundary conditions at \( x = 0, L \), and \( \kappa = s/c, \mathcal{Y} = 1/\mathcal{Z} = A_0/\rho_0 c \).

Solving Eq. 1.176 for \( \alpha \) and \( \beta \) (with \( \mathcal{Z} = 1/\mathcal{Y} \) and determinant \( \Delta = -2\mathcal{Y} \)), at \( x = L \)
\[
\begin{bmatrix}
\alpha \\
\beta
\end{bmatrix}
= \frac{1}{2\mathcal{Y}}
\begin{bmatrix}
-\mathcal{Y} e^{\kappa L} & -e^{\kappa L} \\
-\mathcal{Y} e^{-\kappa L} & e^{-\kappa L}
\end{bmatrix}
\begin{bmatrix}
P \\
-V
\end{bmatrix}
= \frac{1}{2}
\begin{bmatrix}
e^{\kappa L} & \mathcal{Z} e^{\kappa L} \\
e^{-\kappa L} & -\mathcal{Z} e^{-\kappa L}
\end{bmatrix}
\begin{bmatrix}
P \\
-V
\end{bmatrix}
\]  

We may uniquely determine these two weights given the pressure and velocity at the boundary \( x = L \), which is typically determined by the load impedance \( (P_L/V_L) \).

Once the weights have been determined, they may be substituted back into Eq. 1.176, to determine the pressure and velocity amplitudes at any point \( 0 \leq x \leq L \).

\[
\begin{bmatrix}
P \\
V
\end{bmatrix}
= \frac{1}{2}
\begin{bmatrix}
e^{-\kappa x} & e^{\kappa x} \\
\mathcal{Y} e^{-\kappa x} - \mathcal{Y} e^{\kappa x} & e^{-\kappa x} - \mathcal{Z} e^{-\kappa x}
\end{bmatrix}
\begin{bmatrix}
P \\
-V
\end{bmatrix}
\]  

Multiplying these out gives the final transmission matrix

\[
\begin{bmatrix}
P \\
V
\end{bmatrix}
= \frac{1}{2}
\begin{bmatrix}
e^{\kappa(L-x)} + e^{-\kappa(L-x)} & \mathcal{Z}(e^{\kappa(L-x)} - e^{-\kappa(L-x)}) \\
\mathcal{Y}(e^{\kappa(L-x)} - e^{-\kappa(L-x)}) & e^{\kappa(L-x)} + e^{-\kappa(L-x)}
\end{bmatrix}
\begin{bmatrix}
P \\
-V
\end{bmatrix}
\]  

thus

Applying the last boundary condition, we evaluate Eq. 1.177 to obtain the ABCD matrix at the input \( (x = 0) \) (Pipes, 1958)

\[
\begin{bmatrix}
P \\
V
\end{bmatrix}
= \begin{bmatrix}
cosh(\kappa L) & \mathcal{Z} \sinh(\kappa L) \\
\mathcal{Y} \sinh(\kappa L) & \cosh(\kappa L)
\end{bmatrix}
\begin{bmatrix}
P \\
-V
\end{bmatrix}
\]  

**Exercise:** Evaluate this expression in terms of the load impedance. **Sol:** Since \( Z_{load} = P_L/V_L \)

\[
\begin{bmatrix}
P \\
V
\end{bmatrix}
= \begin{bmatrix}
\mathcal{Z}_{load} \cosh(\kappa L) & -\mathcal{Z} \sinh(\kappa L) \\
\mathcal{Z}_{load} \mathcal{Y} \sinh(\kappa L) & -\cosh(\kappa L)
\end{bmatrix}
\]  

**Impedance matrix:** Expressed Eq. 1.181 as an impedance matrix gives (algebra required)

\[
\begin{bmatrix}
P_o \\
P_L
\end{bmatrix}
= \frac{\mathcal{Z}}{\sinh(\kappa L)}
\begin{bmatrix}
cosh(\kappa L) & 1 \\
1 & \cosh(\kappa L)
\end{bmatrix}
\begin{bmatrix}
V_o \\
V_L
\end{bmatrix}
\]

**The input admittance:** Given the input admittance of the horn, it is possible to determine if it is uniform, without further analysis. Namely if the horn is uniform and infinite in length, the input impedance at \( x = 0 \) is

\[
Y_{in}(0, s) \equiv \frac{\mathcal{Y}(0, \omega)}{P(0, \omega)} = \mathcal{Y},
\]

since \( \alpha = 1 \) and \( \beta = 0 \). That is for an infinite uniform horn, there are no reflections.

When the horn is terminated with a fixed impedance \( Z_L \) at \( x = L \), one may substitute pressure and velocity measurements into Eq. 1.177 to find \( \alpha \) and \( \beta \), and given these, one may calculate the reflection at \( x = L \) (see Eq. 1.153, 153)

\[
\Gamma_L(s) \equiv \frac{P^+}{P^+} \bigg|_{x=L} = \frac{\beta}{\alpha} = \frac{P(L, \omega) - \mathcal{Z} V(L, \omega)}{P(L, \omega) + \mathcal{Z} V(L, \omega)} = \frac{Z_L - \mathcal{Z}}{Z_L + \mathcal{Z}}
\]

given sufficiently accurate measurements of the throat pressure \( P(0, \omega) \), velocity \( V(0, \omega) \), and the characteristic impedance of the input \( \mathcal{Z} = \rho_0 c/A(0) \).
Figure 1.35: Throat acoustical resistance \( r_A \) and acoustical reactance \( x_A \), frequency characteristics of infinite parabolic, conical, exponential, hyperbolic and cylindrical horns having a throat area of 1 square centimeter. Note how the “critical” frequency (defined here as the frequency where the reactive and real parts of the radiation impedance are equal) of the horn reduces dramatically with the type of horn. For the uniform horn, the reactive component is zero, so there is no cutoff frequency. For the parabolic horn (1), the cutoff is around 3 kHz. For the conical horn (2), the cutoff is at 0.6 [kHz]. For the exponential horn the critical frequency is around 0.18 [kHz], which is 16 times smaller than for the parabolic horn. For each horn the cross-sectional area of the parabolic, conical exponential and hyperbolic horns is defined as 100 square centimeters at a distance of 100 centimeters from the throat (Olson, 1947, p. 101).

3D conical Horn

For each horn we must find the natural normalization from the range variable to the normalize range variable \( x \). For the conical horn the radius is proportional to the range variable \( r \) thus

\[
A(r) = 4\pi \sin^2 \left( \frac{\Theta}{2} \right) r^2 \ [m^2].
\]

The angle \( \Theta \) is a measure of the solid (cone) angle. When \( \Theta = \pi \) we have the case of the entire sphere so the solid angle is \( 4\pi \) [steradians] and the area is \( 4\pi r^2 \). The formula for the area may be simplified by defining \( A_\theta = 4\pi \sin^2 \left( \frac{\Theta}{2} \right) r_0^2 \ [m^2] \), resulting in the more convenient relation

\[
A(r) = A_\theta \left( \frac{r}{r_0} \right)^2 \ [m^2].
\]

This is a bit tricky because \( A_\theta \) is not a constant since it depends on the place where the area was normalized, in this case \( r_0 \).

Using the conical horn area \( A(r) \propto r^2 \) in Eq. 1.127, p. 140 [or Eq. 1.160 (p. 156)] results in the spherical wave equation (Appendix 5.2.1 p. 237)

\[
P_{rr}(r, \omega) + \frac{2}{r} P_r(r, \omega) = \kappa^2 P(r, \omega) \tag{1.182}
\]

(one must remember the steradian scale factor \( A_\theta \)). Here \( F(r) = \partial_r \ln A(r) = \frac{2}{r} \) (see Table 1.4, p. 154).

Exponential horn: The case of the exponential horn

\[
P^\pm(r) = e^{-mr} e^{\mp j\sqrt{\omega^2 - \omega_c^2} r/c}, \tag{1.183}
\]

which is of special interest because the radiation impedance is purely reactive below the horn’s cutoff frequency \( (\omega < \omega_c = mc_0) \), as may be seen from curves 3 and 4 of Fig. 1.35 since no energy can radiate from an open horn below this frequency.
1.6.11 **Lec 39: Integral forms of \( \nabla() \), \( \nabla \cdot () \) and \( \nabla \times () \)**

There are two definitions for each operator, the point or differential form, and the integral limit form. The integral form gives an intuitive view of what the operator does, and in the limit, converges to the differential form. These two definitions are discussed next.

The divergence and the curl each have fundamental theorems: Gauss’ law (divergence theorem) and Stokes’ law (curl theorem). Without the use of these very fundamental vector calculus theorems, Maxwell’s equations cannot be understood. The history of these important theorems is discussed in the caption of Fig. 1.13, p. 58.

**Gradient:** \( E = -\nabla \phi(x, y, z) \) [V/m]

As briefly summarized on page 130, the differential definition of the gradient maps \( \mathbb{R}^1 \rightarrow \mathbb{R}^3 \). For example, the electric field strength is the gradient of the voltage

\[
E(x) = -[\partial_x, \partial_y, \partial_z]^T \phi(x) = -\left[ \frac{\partial \phi}{\partial x}, \frac{\partial \phi}{\partial y}, \frac{\partial \phi}{\partial z} \right]^T(x).
\]

The negative sign is optional.

The gradient is the slope of the tangent plane of the potential \( \phi(x) \) at \( x \) pointing in the direction of the maximum slope. The gradient (i.e., \( E(x) \)) is \( \perp \) of the constant potential contours of \( \phi(x) \) (the iso-potential lines), since derivatives along any iso-contours must be zero.

As a specific example, consider the paraboloid \( z = 1 - (x^2 + y^2) \) as the potential, with iso-potential circles of constant \( z \) that have radius of zero at \( z = 1 \), and unit radius at \( z = 0 \). The negative gradient

\[
E(x) = -\nabla z(x, y) = 2(x \hat{x} + y \hat{y} + 0 \hat{z})
\]

is \( \perp \) to the circles of constant radius (constant \( z \)), thus points in the direction of the radius.

If one were free-fall skiing this surface, they would be the first one down the hill. Normally skiers try to stay close to the iso-clines (not in the direction of the gradient), so they can stay in control. If you ski an iso-cline, you must walk, since there is no pull due to gravity.

**The integral definition of the gradient:** The gradient may be defined in terms of the integral

\[
\nabla \phi(x, y, z) \equiv \lim_{|S| \rightarrow 0} \left\{ \frac{\int_S \phi(x, y, z) \hat{n} \, d|S|}{|S|} \right\},
\]

over a closed surface \( S \), having area \( |S| \) and volume \( ||S|| \), centered at \( (x, y, z) \)(Greenberg, 1988, p. 773).\(^{100}\) Here \( \hat{n} \) is the unit vector perpendicular to the surface \( S \)

\[
\hat{n} = \frac{\nabla \phi}{||\nabla \phi||}.
\]

The dimensions of Eq. eq:GradInt are in the units of the potential times the area, divided by the volume, as needed for a gradient (e.g., [Volts/m]).

The natural way to define the surface and volume is to place the surface on the iso-potential surfaces, forming either a cube or pill-box shaped volume. As the volume goes to zero, so must

\(^{100}\)See further discussions on pages Greenberg (1988, pp. 778, 791, 809)
the area. One must avoid *irregular volumes* such that the area is finite as the volume goes to zero (Greenberg, 1988, footnote p. 762).

A well known example is the potential

$$\phi(x, y, z) = \frac{Q}{\epsilon_o \sqrt{x^2 + y^2 + z^2}} = \frac{Q}{\epsilon_o R} \text{ [Volts]}$$

around a point charge $Q$ [SI Units of Coulombs]. The constant $\epsilon_o$ is the *permittivity* [Farad/m$^2$].

**How does this work?** To better understand what Eq. 1.184 means, consider a three-dimensional Taylor series expansion of the potential in $x$ about the limit point $x_o$. To compute the *higher order terms* (HOT) one needs the Hessian matrix

$$H_{i,j} = \frac{\partial^2 \phi(x)}{\partial x_i \partial x_j},$$

which will exist if the potential is analytic in $x$ at $x_o$.

We could define the gradient using this relationship as

$$\nabla \phi(x_o) = \lim_{x \to x_o} \frac{\phi(x) - \phi(x_o)}{x - x_o}.$$  

For this definition to apply, $x$ must approach $x_o$ along $\hat{n}$.

The natural form for the surface $|S|$ is to lie along the iso-potential surfaces as much as possible, so that the integral is a constant (the potential) times the area. The remainder of the surface must be perpendicular to these iso-potential surfaces, in the direction of the gradient, or maximum change of the potential. The secret to the integral definition is in taking the limit. As the volume $||S||$ shrinks to zero, the HOT terms are small, and the integral reduces to the first order term in the Taylor expansion, since the constant term integrates to zero. Such a construction is used in the proof of the Webster Horn equation (??, p. ??; Fig. 1.34, p. 151).

**Divergence: $\nabla \cdot D = \rho$ [Col/m$^3$]**

As briefly summarized on page 131, the differential definition of the gradient which maps $\mathbb{R}^3 \mapsto \mathbb{R}^1$ is

$$\nabla \cdot D \equiv [\partial_x, \partial_y, \partial_z] \cdot D = \left[ \frac{\partial D_x}{\partial x} + \frac{\partial D_y}{\partial y} + \frac{\partial D_z}{\partial z} \right] = \rho(x, y, z)$$

The divergence is a direct measure of the flux (flow) of vector field it operates on ($D$), coming from $x$. A vector field is said to be *incompressible* if the divergence of that field is zero. It is therefore compressible when the divergence is non-zero. Compared to air, water is considered to be incompressible. However at very low frequencies, air can also be considered as incompressible. Thus the definition of compressible depends on the wavelength in the medium, so the terms must be used with some awareness of the circumstances.

If we restrict ourselves to the electrical case, then at this point it is helpful to think of the physics of the electron, a negatively charged particle that is a single point in space. Of course this it is nonsense to consider an electron to be a point, but given that it is so small (it is so small it is difficult to estimate its size). This size is called the Lorentz radius, and is estimated to be $2.810^{-15}$ [m]. One could summarize the Lorentz radius as follows: *Here lie many unsolved problems in*
1.6. STREAM 3B: VECTOR CALCULUS (10 LECTURES)

**Figure 1.36: Derivation of Gauss's law**

The divergence of the velocity \( \nabla \cdot \mathbf{u} \), within \( \delta x \), shown as the filled shaded region, is integrated over the closed volume. Then the divergence theorem is applied, transforming the integral to a surface integral normal to the surface, resulting in the difference of the two volume velocities \( \delta \nu = \nu(x + \delta x) - \nu(x) = [\mathbf{u}(x + \delta x) \cdot \mathbf{A}(x + \delta x) - \mathbf{u}(x) \cdot \mathbf{A}(x)] \), as the limit of \( \delta x \to 0 \). Because the ends are in the direction of \( \mathbf{u} \), the contribution to the volume velocity from the ends is zero.

**physics.** More specifically, at dimensions of the Lorentz radius, what exactly is the structure of the electron?

Ignoring the difficulties, if one integrates the charge density of the electron over the Lorentz radius and places the total charge at a single point, then one may make a grossly over simplified model of the electron. For example the Electric Displacement (flux density) around a point charge \( (\mathbf{D} = \epsilon_o \mathbf{E}) \)

\[
\mathbf{D} = -\epsilon_o \nabla \phi(R) = -Q \nabla \left\{ \frac{1}{R} \right\} = -Q \delta(R).
\]

This is a formula taught in many classic texts, but one should remember how crude a model of an electron it is. But it does describe the electric flux in an easily remembered form. However, computationally, it is less nice, due to the delta function. A limitation of this model is that the electron has a magnetic dipole moment (aka, spin), which a simple point charge does not capture. When placed in a magnetic field, due to the dipole, the electron will align itself with the field.

**Divergence and Gauss’ law**

Like the gradient, the divergence of a vector field may be defined as the surface integral of a compressible vector field, as a limit as the volume enclosed by the surface goes to zero. As for the case of the gradient, for this definition to make sense, the surface \( S \) must be a closed, defining volume \( V \). The difference is that the surface integral is over the normal component of the a vector field being operated on. Specifically (Greenberg, 1988, p. 762-763)

\[
\nabla \cdot \mathbf{D} = \lim_{|S| \to 0} \left\{ \frac{\int_S \mathbf{D} \cdot \mathbf{\hat{n}} \ d|S|}{|S|} \right\} = \rho(x, y, z).
\]

(1.185)

As with the case of the gradient we have defined the surface as \( S \), its area as \( |S| \) and the volume within as \( ||S|| \). As the area \( |S| \) goes to zero, so does the volume \( ||S|| \). This is a necessary condition for the integral to converge to the divergence.

Here \( \mathbf{\hat{n}} \) is a unit vector normal to the surface \( S \). The limit as the volume goes to zero, defines the total flux across the surface. Thus the surface integral is a measure of the total flux \( \perp \) to the surface. It is helpful to compare this formula with that for the gradient Eq. 1.184.
GAUSS’ LAW: The above definitions resulted in a major breakthrough in vector calculus, and the first\textsuperscript{102} fundamental theorem of vector calculus, Gauss’ law\textsuperscript{103}

The surface integral over the normal component of the flux (i.e., the total flux) is equal to the volume integral over the divergence of the flux.

For the electrical case this is equivalent to the observation that the total flux across the surface is equal to the net charge enclosed by the surface. Since the volume integral over charge density $\rho(x, y, z)$ is total charge enclosed $Q_{enc}$ [Col]

$$Q_{enc} = \iiint_{||S||} \nabla \cdot D \ ||dS|| = \iiint_{||S||} D \cdot \hat{n} \ d|S|.$$  

(1.186)

It trivially follows that when $Q_{enc} = 0$. That is when the surface integral over the normal component of $D(x)$, and the volume integral the charge density $\nabla \cdot D = \rho(x)$, are both zero.

Taking the derivative with respect to time Eq. 1.186 becomes, the current flowing out of the surface (i.e., the component normal to the surface)

$$I = \iiint_{||S||} D \cdot \hat{n} \ d|S| = \dot{Q}_{enc} = \iiint_{||S||} \dot{\rho}_{enc} \ d||S||.$$  

(1.187)

Paraphrasing Feynman (1970c, p. 13-2)

The current leaving the closed surface $|S|$ equals the rate of the charge leaving that volume $||S||$, defined by that surface.

Of course the surface must be closed to define the volume, a necessary condition for Gauss’s law. This reduces to a common sense summary that can be grasped intuitively, an example of the beauty in Feynman’s understanding.

INTEGRAL DEFINITION OF THE CURL: $\nabla \times H = C$ [AMPS]

As briefly summarized on page 131 (p. 131), the differential definition of the Curl maps $\mathbb{R}^3 \rightarrow \mathbb{R}^3$. For example, the curl of the magnetic field strength $H(x)$ is equal to the total current $C$

$$\nabla \times H \equiv \left| \begin{array}{ccc} \hat{x} & \hat{y} & \hat{z} \\ \partial_x & \partial_y & \partial_z \\ H_x & H_y & H_z \end{array} \right| = C.$$  

As we shall see in Sect. 1.6.13 (p. 173), the curl and the divergence are both key when writing out Maxwell’s equation four equations. Without a full understanding of these two differential operators ($\nabla \cdot$, $\nabla \times$), there is no hope of understanding Maxwell’s basic result, typically viewed as the most critical elements of mathematical physics, and the starting point for Einstein’s relativity theories. Some will say that quantum mechanics fall outside the realm of MEs, but this at least open to debate, if not hotly debated.

The curl is a measure of the rotation of a vector field. If this were water, it would correspond to the angular momentum of the water, such as water going down the drain, as in a whirlpool, or

\textsuperscript{102}As best I know, the three fundamental theorems of vector calculus, Gauss’ law, the Stokes’ law and Helmholtz’s theorem (aka Vector decomposition theorem) are not ordered.

\textsuperscript{103}aka, Gauss’s law
### 1.6. STREAM 3B: VECTOR CALCULUS (10 LECTURES)

The integral definition of the curl is related to that of the divergence (Greenberg, 1988, p. 774), as an integration over the tangent to the surface, except: 1) the curl is defined as the cross product \( \hat{n} \times \mathbf{H} [\text{Amps/m}^2] \), of \( \hat{n} \) with the current density \( \mathbf{H} \), and 2) the surface is open, leaving a boundary \( B \) along the open edge. As with the divergence, which lead to Gauss’ law, this definition leads to a second fundamental theorem of vector calculus: Stokes’ law (aka the Curl theorem).

**Figure 1.37:** The integral definition of the curl is related to that of the divergence (Greenberg, 1988, p. 774), as an integration over the tangent to the surface, except: 1) the curl is defined as the cross product \( \hat{n} \times \mathbf{H} [\text{Amps/m}^2] \), of \( \hat{n} \) with the current density \( \mathbf{H} \), and 2) the surface is open, leaving a boundary \( B \) along the open edge. As with the divergence, which lead to Gauss’ law, this definition leads to a second fundamental theorem of vector calculus: Stokes’ law (aka the Curl theorem).

**Curl and Stokes law:** As in the cases of the gradient and divergence, the curl also may be written in integral form, allowing for the physical interpretation of its meaning.

Surface integral definition of \( \nabla \times \mathbf{H} = \mathbf{C} \) where the current \( \mathbf{C} \) is \( \perp \) to the rotation plane of \( \mathbf{H} \). Stokes law states that the open surface integral over the normal component of the curl of the magnetic field strength \( (\hat{n} \nabla \times \mathbf{H}) [\text{Amps/m}^2] \) is equal to the line integral \( \oint_B \mathbf{H} \cdot dl \) along the boundary \( B \). As summarized in Fig. 1.37, Stokes’ law is

\[
\mathcal{I}_{\text{enc}} = \iint_{|S|} (\nabla \times \mathbf{H}) \cdot \hat{n} \ d|S| = \oint_B \mathbf{H} \cdot dl \ [\text{Amps}],
\]

namely

**The line integral of \( \mathbf{H} \) along the open surface’s boundary \( B \) is equal to the total current enclosed \( \mathcal{I}_{\text{enc}} \) [Amps].**

**Summing it up:** As mentioned earlier (Fig. 1.13, p. 58), the history of the discovery and proof of this theorem was both complex and interesting.

Since integration is a linear process (sums of smaller elements), one may tile, or tessellate the surface, breaking it up into a sum over smaller surfaces, and their boundaries, the sum over which is equal to the integral over the original boundary. This is an important concept, which leads to the proof of Stokes’ law.

The integral formulations of Gauss’ and Stokes’ laws use \( \hat{n} \cdot \mathbf{D} \) and \( \mathbf{H} \times \hat{n} \) in the integrands. The key distinction between the two laws naturally follows from the properties of the scalar \( (\mathbf{A} \cdot \mathbf{B}) \) and vector \( (\mathbf{A} \times \mathbf{B}) \) products, as discussed in Sect. 1.4.8, p. 80, and detailed in Fig. 1.16, p. 82.

---

**Example:** When \( \mathbf{H} = -y\hat{x} + x\hat{y} + 0\hat{z} \), \( \nabla \times \mathbf{H} = 2\hat{z} \), thus has a constant *rotation*; when \( \mathbf{H} = 0\hat{x} + 0\hat{y} + z^2\hat{z} \) then \( \nabla \times \mathbf{H} = 0 \) has a curl of zero, thus has zero rotation thus is said to be *irrotational*. There are rules that precisely govern when a vector field is rotational versus irrotational, and compressible versus incompressible. These classes are dictated by Helmholtz’s Theorem, the third fundamental theorem of vector calculus (Eq. 1.194, p. 171).

---

with wind, a tornado. A spinning top is another an excellent example, given a spinning solid body. While a top (aka gyro-scope) will fall over if not spinning, once it is spinning, it can stability stand on its pointed tip. These structure are stable due to conservation of angular momentum: Once something is spinning, it will continue to spin.
CHAPTER 1. INTRODUCTION

Table 1.5: Summary of the fundamental theorems of integral calculus, each of which deals with integration. There are at least two main theorems related to scalar calculus, and three more for vector calculus.

<table>
<thead>
<tr>
<th>Name</th>
<th>Mapping</th>
<th>p.</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Leibniz (FTC)</td>
<td>$\mathbb{R}^1 \rightarrow \mathbb{R}^0$</td>
<td>108</td>
<td>Area under a real curve.</td>
</tr>
<tr>
<td>FTCC</td>
<td>$\mathbb{R}^1 \rightarrow \mathbb{R}^0$</td>
<td>108</td>
<td>Area under a complex curve.</td>
</tr>
<tr>
<td>Cauchy’s Theorem</td>
<td>$\mathbb{C}^1 \rightarrow \mathbb{C}^0$</td>
<td>121</td>
<td>Close integral over analytic region is zero</td>
</tr>
<tr>
<td>Cauchy’s Integral Formula</td>
<td>$\mathbb{C}^1 \rightarrow \mathbb{C}^0$</td>
<td>121</td>
<td>Residue integration and complex analytic functions.</td>
</tr>
<tr>
<td>residue Theorem</td>
<td>$\mathbb{C}^1 \rightarrow \mathbb{C}^0$</td>
<td>121</td>
<td>Residue integration and complex analytic functions.</td>
</tr>
</tbody>
</table>

To fully appreciate the differences between Gauss’ and Stokes’ laws, these two types of vector products must be mastered.

Paraphrasing Feynman (1970c, 3-12),

1. $V_2 = V_1 + \int_1^2 \nabla \Phi \cdot dS$
2. $\oint D \hat{n} \, d|S| = \oint \nabla \cdot D \, d||S||$
3. $\oint E \, dl = \oint_{|S|} (\nabla \times E) \cdot \hat{n} \, d|S|$

The first is the FTC. The second is Gauss’s law, and the third is Stokes’ theorem.

1.6.12 Lec 40 Second-order operators: Terminology

Besides the above first order vector derivatives, second order combinations exist, the most common being the scalar Laplacian $\nabla \cdot \nabla (\cdot) = \nabla^2 (\cdot)$ (Table 1.3, p. 130; Appendix 5.2.1, p. 237).

There are other important second-order combinations of $\nabla$, enough that we need a memory aid to remember them. Thus I define mnemonics DoC, DoG, CoG, CoC and GoD as follows:

1. DoG: Divergence of the Gradient ($\nabla \cdot \nabla (\cdot) = \nabla^2 (\cdot)$), i.e., Laplacian
2. DoC*: Divergence of the Curl ($\nabla \cdot \nabla \times$),
3. CoG*: Curl of the Gradient ($\nabla \times \nabla$),
4. CoC: Curl of the curl ($\nabla \times \nabla \times$), and
5. GoD: Gradient of the Divergence (GoD) $\nabla \nabla \cdot$ and the vector Laplacian GoD* $\nabla^2$.

DoC*() and CoG*() are special, because they are always zero:

$$\nabla \times \nabla \phi = 0; \quad \nabla \cdot \nabla \times A = 0,$$

a property that makes them useful in proving the Fundamental Theorem of Vector Calculus (Helmholtz’ decomposition, Eq. 1.194, p. 171). A third key vector identity CoC may be expanded as

$$\nabla \times \nabla \times A = \nabla(\nabla \cdot A) - \nabla^2 A,$$  \hspace{1cm} (1.189)

defining the vector Laplacian (i.e., GoD*=GoD-CoC): $\nabla^2 (\cdot) = \nabla \nabla \cdot (\cdot) - \nabla \times \nabla \times (\cdot)$.
1.6. STREAM 3B: VECTOR CALCULUS (10 LECTURES)

When using second order differential operators one must be careful with the order of operations, which can be subtle in a few cases. Most of this is common sense. For example, don’t operate on a scalar field with $\nabla \times$, and don’t operate on a vector field with $\nabla$. The vector Laplacian $\nabla^2 \mathbf{A}$ must not be thought of as $\nabla(\nabla \cdot \mathbf{A})$, rather it acts as the Laplacian on each vector component $\nabla^2 \mathbf{A} = \nabla^2 A_x \mathbf{\hat{x}} + \nabla^2 A_y \mathbf{\hat{y}} + \nabla^2 A_z \mathbf{\hat{z}}$.

**Exercise:** Show that $\nabla \nabla \mathbf{A}$ and $\nabla \nabla^* \mathbf{A}$ differ. Sol: Use CoC on a $\mathbf{A}$ to explore this relationship.

**Helmholtz’s decomposition**

We may now restate everything defined above in terms of two types of vector fields that decompose every analytic vector field. The *irrotational fields* is defined as one that is “curl free.” An *incompressible field* is one that is “divergence free.” According to Helmholtz’s decomposition, every analytic vector field may be decomposed into independent rotational and a compressible components. Another name for *Helmholtz decomposition* is the *Fundamental theorem of vector calculus* (FTVC), Gauss’ and Stokes’ laws, along with Helmholtz’s decomposition, form the three key fundamental theorems of vector calculus. Images of Helmholtz and Kirchhoff are provided in Fig. 1.38, p. 172.

The term *magnetic solenoidal field* is one that is generated by a solenoidal coil, is an excellent approximation, uniform inside the coil. As a result, the curl of a solenoidal field is, to a good approximation, zero. I recommend you know this term (it is widely used), but suggest the preferred term *irrotational*. Strictly speaking the term “solenoidal field” only applies to a magnetic field produced by a solenoid, thus the term is specific to that case.

**The decomposition of differentiable vector fields:** This theorem is easily stated (and proved), but less easily appreciated (Heras, 2016). A physical description facilitates: *Every vector field may be split into two independent parts: dilation and rotation.* We have seen this same idea appear in vector algebra, where the scalar and cross products of two vectors are perpendicular (Fig. 1.16, p. 82). Also think of linear and angular momentum, which are independent in that they represent different ways of absorbing and delivering kinetic energy, leading to independent degrees of freedom. Thus the idea of the linear and rotational parts being independent, is a common theme, rooted in geometry. In the same sense, a vector field may be split into a dilation and rotational parts, which are independent (but can interact under certain conditions). An object with mass can be moving along a path, and independently be rotating. The two modes of motion define different types of kinetic energy, translational (compressional) and rotational. In some real sense, Helmholtz decomposition quantifies these degrees of freedom, one DoF for translation and three DoFs for rotation. Each eigen-mode of vibration can be viewed as a DoF. The role of the FTVC is especially powerful when applied to Maxwell’s Eqs.

**Solving Maxwell’s equations**

The application of FTVC is not as obvious as it might seem when applied to Maxwell’s equations, due to DoC and Cog. In Section 1.6.1 we defined the electric and magnetic field strengths in terms of the scalar and vector potentials $\phi(x, t)$ and $\mathbf{A}(x, t)$, repeated here for convenience. The electric field strength is (Eq. 1.111, p. 129)

$$E = -\nabla \phi - \frac{\partial}{\partial t} \mathbf{A} \quad (1.111)$$
while the magnetic field strength is (Eq. 1.112, p. 129)

\[ B = \nabla \times A \]  (1.112)

which are the solutions to Maxwell’s equations in terms of the two potentials (Sommerfeld, 1952, p. 146). Note these relations are invariant to the addition of a constant to each potential, and by the application of DoC and CoG, they are equivalent to Maxwell’s equations.

**Helmholtz theorem and the potential representation:** Taking the curl of Eq. 1.111, and using CoG=0, recovers Maxwell’s electric equation

\[
\nabla \times E = -\nabla \times \nabla \Phi + \nabla \times \frac{\partial A}{\partial t} = -\frac{\partial B}{\partial t},
\]  (1.190)

while taking the divergence along with DoC=0

\[
\nabla \cdot \nabla \times E = 0 - \frac{\partial \nabla \cdot B(x, t)}{\partial t} = 0,
\]  (1.191)

requires that \( \nabla \cdot B \) is independent of time, and therefore that

\[ \nabla \cdot B(x) = 0. \]  (1.192)

We would like to recover Maxwell’s magnetic equation \( \nabla \times H = C \) from the potential solution. Taking the curl of Eq. 1.112 gives

\[
\frac{1}{\mu_o} \nabla \times \mu_o H = C = \nabla \times \nabla \times A = \nabla \nabla \cdot A - \nabla^2 A.
\]

This says that the current \( C \) only depends on \( A \), which follows directly from Eq. 1.112 forward.

Since \( A \) must satisfy the wave equation

\[ \nabla^2 A = \frac{1}{c_o^2} A - C, \]

which requires that

\[ \nabla \nabla \cdot A = \frac{1}{c_o^2} A. \]

Taking the divergence of Eq. 1.111 gives an expression for \( \nabla \cdot A \):

\[
\frac{1}{\epsilon_o} \nabla \cdot D = \frac{\rho}{\epsilon_o} = -\nabla^2 \phi - \partial_t \nabla \cdot A.
\]

Here \( \epsilon_0 \) [Col/m] and \( \mu_o \) [H/m].
Exercise: Starting from the values of the speed of light $c_o = 3 \times 10^8$ [m/s] and the characteristic resistance of light waves $r_o = 377$ [Ohms], use the formula for $c_o = 1/\sqrt{\mu_o \epsilon_o}$ and $r_o = \sqrt{\epsilon_o/\mu_o}$ to find values for $\epsilon_o$ and $\mu_o$. Sol: Squaring $c_o^2 = 1/\mu_o \epsilon_o$ and $r_o^2 = \mu_o/\epsilon_o$ we may solve for the two unknowns: $c_o^2 r_o^2 = 1/\mu_o \epsilon_o = 1/\epsilon_o$, thus $\epsilon_o = 1/c_o r_o = (377/3) \times 10^{-8} \approx 125.67 \times 10^{-8}$. The value of $\mu_o$ is defined in the international SI standard as $4\pi 10^{-7} \approx 12.56610^{-7}$ [H/m].

In conclusion, Eq. 1.111, along with DoC=0 and CoG=0, give Maxwell’s Eq. 1.190 and Eq. 1.192 result. It would appear that Eq. 1.111 is the key. This equation defines the magnetic component of the field, expressed in terms of its vector potential, in the same way as Eq. 1.110 describes $E(x,t)$ in terms of the scalar potential (voltage). Does the same argument apply for Eq. 1.112?

Exercise: Take the divergence of Maxwell’s equation for the magnetic intensity
\[
\nabla \times H(x,t) = J(x,t) + \frac{\partial}{\partial t} D(x,t).
\]
and explain what results. Sol: The divergence of the curl is always zero (DoC=0), thus
\[
\nabla \cdot \nabla \times H(x,t) = \nabla \cdot J(x,t) + \frac{\partial}{\partial t} \rho(x,t) = 0,
\]
which is conservation of charge.

Helmholtz’s decomposition is expressed as the linear sum of a scalar potential $\phi(x, y, z)$ (think voltage) and a vector potential (think magnetic vector potential). Specifically
\[
E(x,s) = -\nabla \phi(x,s) + \nabla \times A(x,s),
\]
where $\phi$ is the scalar and $A$ is the vector potential, as a function of the Laplace frequency $s$. Of course this decomposition is general (not limited to the electro-magnetic case). It applies to linear fluid vector fields, which includes most liquids and air. When the rotational and dilation become coupled, this relation must break down.\(^{104}\)

To show how this relationship splits the vector fields $E$ into its two parts, we need DoC and CoG, the two key vector identities that are always zero for analytic fields: the curl of the gradient (CoG)
\[
\nabla \times \nabla \phi(x) = 0,
\]
and the divergence of the curl\(^ {105}\) (DoC)
\[
\nabla \cdot (\nabla \times A) = 0.
\]

The above identities are easily verified by working out a few specific examples, based on the definitions of the three operators, gradient, divergence and curl, or in terms of the operator’s integral definitions, defined in Sect. 1.6.11, p. 163. The identities have a physical meaning, as stated above: every vector field may be split into its translational and rotational parts. If $E$ is the electric field [V/m], $\phi$ is the voltage and $A$ is the induced rotational part, induced by a current. We shall explore this in our discussion of Maxwell’s equations in Sect. 1.6.13 and Chapter 5.

\(^{104}\)The nonlinear Navier–Stokes equations may be an example.

\(^{105}\)Helmholtz was the first person to apply mathematics in modeling the eye and the ear (Helmholtz, 1863a).
By applying these two identities to Helmholtz’s decomposition, we can better appreciate the theorems significance. It is a form of proof actually, once you have satisfied yourself that the vector identities are true. In fact one can work backward using a physical argument, that rotational momentum (rotational energy) is independent of the translational momentum. Once these forces are made clear, the meaning of the vector operations all take on a very well defined meaning, and the mathematical constructions, centered around Helmholtz’s theorem, begins to provide some common-sense meaning. One could conclude that the physics is simply related to the geometry via the scalar and vector product.

Specifically, if we take the divergence of Eq. 1.194, and use the DoG

$$\nabla \cdot E = \nabla \cdot \{ -\nabla \phi + \nabla \times A \}^0 = -\nabla \cdot \nabla \phi = -\nabla^2 \phi,$$

since the DoG zeros the vector potential $A(x, y, z)$. If instead we use the CoG

$$\nabla \times E = \nabla \times \{ -\nabla \phi^0 + \nabla \times A \} = \nabla \times \nabla \times A = \nabla (\nabla \cdot A) - \nabla^2 A,$$

since the CoG zeros the scalar field $\phi(x, y, z)$, and followed with the used of GoD.

**The four categories of linear fluid flow:** The following is a summary of the four cases for fluid flow, as summarized in Fig. 1.39:

1.1 Compressible and rotational fluid (general case): $\nabla \phi \neq 0, \nabla \times w \neq 0$. This is the case of wave propagation if a medium where viscosity can not be ignored, as in the case of acoustics close to the boundaries, where viscosity contributes losses (Batchelor, 1967).

1.2 Incompressible, but rotational fluid (Lubrication theory): $v = \nabla \times w \neq 0, \nabla \cdot v = 0, \nabla^2 \phi = 0$. In this case the flow is dominated by the walls, the viscosity and heat transfer introduce shear. This is typical of lubrication theory.

2.1 Fluid compressible irrotational flow (acoustics): $v = \nabla \phi, \nabla \times w = 0$. Here losses (viscosity and thermal diffusion) are small (assumed to be zero). One may define a velocity potential
1.6. STREAM 3B: VECTOR CALCULUS (10 LECTURES)

### Field:
- **Compressible**
  - \( \nabla \cdot \mathbf{v} \neq 0 \)
- **Incompressible**
  - \( \nabla \cdot \mathbf{v} = 0 \)

<table>
<thead>
<tr>
<th>Rotational</th>
<th>( \nabla \times \mathbf{v} \neq 0 )</th>
<th>( \mathbf{v} = \nabla \phi + \nabla \times \mathbf{\omega} )</th>
</tr>
</thead>
</table>
| Vector wave Eq. (EM) | \( \nabla^2 \mathbf{v} = \frac{1}{c^2} \ddot{\mathbf{v}} \) | \( \mathbf{v} = \nabla \times \mathbf{w} \)

<table>
<thead>
<tr>
<th>Irrotational</th>
<th>( \nabla \times \mathbf{v} = 0 )</th>
<th>( \mathbf{v} = -\nabla \phi )</th>
</tr>
</thead>
</table>
| Acoustics | \( \mathbf{v} = \nabla \psi \) | \( \nabla^2 \phi = 0 \)
| \( \nabla^2 \varrho(x, t) = \frac{1}{c^2} \ddot{\varrho}(x, t) \) | \( \mathbf{v} = \nabla \times \mathbf{w} \) |
| Statics | \( \nabla \times \mathbf{v} = 0 \) | \( \mathbf{v} = \nabla \times \mathbf{w} \)
| \( \nabla^2 \phi = 0 \) | \( \mathbf{v} = \nabla \times \mathbf{w} \)
| Boundary layers | \( \mathbf{v} = \nabla \times \mathbf{w} \) | \( \mathbf{v} = \nabla \times \mathbf{w} \)

Figure 1.39: Figure showing the four possible classifications of scalar and vector potential fields, rotational/irrotational, compressible/incompressible. Rotational fields are generated by the vector potential (e.g., \( \mathbf{A}(x, t) \)), while compressible fields are generated by the scalar potentials (e.g., voltage \( \phi(x, t) \), velocity \( \psi \), pressure \( \varrho(x, t) \), temperature \( T(x, t) \)).

\( \psi \), the gradient of which gives the air particle velocity. For an irrotational fluid (\( \nabla \times \mathbf{v} = 0 \)) (Greenberg, 1988, p. 826), thus \( \mathbf{v} = -\nabla \phi \). This is the case of the conservative field, where \( f \mathbf{v} \cdot \hat{n} \, dR \) only depends on the end points, and \( \oint \mathbf{B} \cdot \hat{n} \, dR = 0 \). When a fluid may be treated as having no viscosity, it is typically assumed to be irrotational, since it is the viscosity that introduces the shear (Greenberg, 1988, p. 814). A fluid’s angular velocity is \( \Omega = \frac{1}{2} \nabla \times \mathbf{v} = 0 \), thus irrotational fluids have zero angular velocity (\( \Omega = 0 \)).

2.2 Incompressible and irrotational fluid (statics): \( \nabla \cdot \mathbf{v} = 0 \) and \( \nabla \times \mathbf{v} = 0 \) thus \( \mathbf{v} = \nabla \phi \) and \( \nabla^2 \phi = 0 \). An example of such a case is water in a small space at low frequencies, where the wavelength is long compared to the size of the container, the fluid may be treated as incompressible. When \( \nabla \times \mathbf{v} = 0 \), the effects of viscosity may be ignored, as it is the viscosity that creates the shear leading to rotation. This is the case of modeling the cochlea, where losses are ignored and the quasi-static limit is justified.

In summary, each of the cases is some sort of approximation that best applies in the low frequency limit. This is why it is called quasi-static, meaning low, but not zero frequency, where the wavelength is large compared with the dimensions (e.g., diameter).

### 1.6.13 Lec 41 Maxwell’s Equations: The unification of electricity and magnetism

Once you have mastered the three basic vector operations, the gradient, divergence and curl, you are ready to appreciate Maxwell’s equations. Like the vector operations, these equations may be written in integral or vector form. A very important difference is, that with Maxwell’s equations, we are dealing with well defined physical quantities. The scalar and vector fields take on meaning, and units. Thus to understand these important equations, one must master the names of the four fields \( \mathbf{E}, \mathbf{H}, \mathbf{B}, \mathbf{D} \).

Figure 1.40: A solenoid is a uniform coil of wire. When a current is passed through the wire, a uniform magnetic field intensity \( \mathbf{H} \) is created. From a properties point of view, this coil is indistinguishable from a permanent bar magnet, having north and south poles. Depending on the direction of the current, one end of a finite solenoidal coil is the north pole of the magnet, and the other end is the south pole. The uniform field inside the coil is called solenoidal, a confusing synonym for irrotational. (Figure from Wikipedia.)
CHAPTER 1. INTRODUCTION

Field strengths $E, H$: As summarized in Fig. 1.6 there are two field strengths, the electric $E$ with units of [Volts/meter] and the magnetic $H$ having units of [Amps/meter]. Their the ratio $|E|/|H|$ is in [ohms].

To understand the meaning of $E$, if two conducting plates are place 1 [m] apart, with 1 [V] across them, the electric field is $E = 1$ [V/m]. If a charge (i.e., electron) is placed in an electric field, it feels a force $f = qE$, where $q$ is the magnitude of the charge [Col].

To understand the meaning of $H$, consider the solenoid made of wire, as shown in Fig. 1.40, which carries a current of 1 [Amp]. The magnetic field $H$ inside such a solenoid is uniform and is pointed along the long axis, with a direction that depends on the polarity of the applied voltage (i.e., direction of the current in the wire).

Table 1.6: The variables of Maxwell’s equations have names (e.g., EF, MI) and units (in square brackets [SI Units]). The units are required to gain a full understanding of each of the four variable and their corresponding equation. For example, Eq. EF has units [Volts/m]. When you integrate $E$ from $x = a, b$, you obtain the voltage difference between those two points. The speed of light in vacuo is $c = 3 \times 10^8 = 1/\sqrt{\mu_0 \varepsilon_0}$ [m/s], and the characteristic resistance of light $r_0 = 377 = \sqrt{\mu_0 / \varepsilon_0}$ [m/A] (i.e., ohms). The dot over a vector is shorthand for the partial with respect to time (i.e., $\dot{B} = \partial B/\partial t$).

<table>
<thead>
<tr>
<th>Symbol</th>
<th>Name</th>
<th>Units</th>
<th>Maxwell’s Eq.</th>
</tr>
</thead>
<tbody>
<tr>
<td>$E$</td>
<td>EF: Electric Field strength</td>
<td>[Volts/m]</td>
<td>$\nabla \times E = -\partial_t B$</td>
</tr>
<tr>
<td>$D = \varepsilon_0 E$</td>
<td>ED: Electric Displacement (flux density)</td>
<td>[Col/m$^2$]</td>
<td>$\nabla \cdot D = \rho$</td>
</tr>
<tr>
<td>$H$</td>
<td>MF: Magnetic Field strength</td>
<td>[Amps/m]</td>
<td>$\nabla \times H = \partial_t D$</td>
</tr>
<tr>
<td>$B = \mu_0 H$</td>
<td>MI: Magnetic Induction (flux density)</td>
<td>[Wb/m$^2$]</td>
<td>$\nabla \cdot B = 0$</td>
</tr>
</tbody>
</table>

Flux: Flux is a flow, such as the mass flux of water flowing in a pipe [kg/s], driven by a force (pressure drop) across the ends of the pipe, or the heat flux in a thermal conductor, having a temperature drop across it (i.e., a window or a wall). The flux is the same as the flow, be it current, mass or heat. In Maxwell’s equations there are also two fluxes, the electric flux $D$, and the magnetic flux $B$. The flux density units for $D$ [Amps/m$^2$] (flux in [Amps]) and the magnetic flux $B$ is measured in Webers [Wb] [A/m$^2$]) or Tesla (Henry-Amps/area) [HA/m$^2$].

Maxwell’s equations

As shown in Fig. 1.6, Maxwell’s equations consist of two curl equations, operating on the field strengths EF $E$ and MF $H$, and two divergence equations, operating of the field fluxes ED $D$ and MI $B$. Stokes’ law may be applied to the curl equations and Gauss’ law may be used on the divergence equations. This should be logically obvious.

Example: When a static current is flowing in a wire in the $\hat{z}$ direction, the magnetic flux is determined by Stokes theorem (Fig. 1.37). Thus just outside of the wire we have

$$I_{enc} = \iint_{S} (\nabla \times H) \cdot \hat{n} \, d|S| = \oint_{B} H \cdot dl \quad \text{[Amps]} \quad (1.197)$$

For this simple geometry, of the current in a wire is related to $H(x,t)$ by

$$I_{enc} = \oint_{B} H \cdot dl = H_0 2\pi r.$$
Here $H_\theta$ is perpendicular to both the radius $r$ and the direction of the current $\hat{z}$. Thus

$$H_\theta = \frac{I_{enc}}{2\pi r},$$

and we see that $H$, and thus $B = \mu_0 H$ drop off as the reciprocal of the radius $r$.

**Exercise:** Explain how Stokes’ law may be applied to $\nabla \times E = -\dot{B}$, and explain what it means. Hint: This is the identical argument given above for the current in a wire, but for the electric case.

**Sol:** Integrating the left side of equation EF over an open surface results in an voltage (emf) induced in the loop closing the boundary $\mathcal{B}$ of the surface

$$\phi_{\text{induced}} = \oint_{\mathcal{B}} (\nabla \times E) \cdot \hat{n} \, dl = \int_{\mathcal{B}} E \cdot dl \quad \text{[Volts]}.$$

The emf (electromagnetic force) is the same as the Thévenin source voltage induced by the rate of change of the flux. Integrating the Eq. 1.6.13 over the same open surface $\mathcal{S}$ results in the source of the induced voltage $\phi_{\text{induced}}$, which is proportional to the rate of change of the flux [Webers]

$$\phi_{\text{induced}} = -\frac{\partial}{\partial t} \int_{\mathcal{S}} B \cdot \hat{n} \, dA = L \dot{\psi}, \quad \text{[Webers/s]}.$$

where $L$ is the inductance of the wire. The area integral on the left is in [W/m$^2$] resulting in the total flux crossing normal to the surface $\psi$ [W].

If we apply Gauss’ law to the divergence equations, we find the total flux crossing the closed surface.

**Exercise:** Apply Gauss’ law to equation ED and explain what it means in physical terms. **Sol:** The area of the normal component of $D$ is equal to the volume integral over the charge density, thus Gauss’ law says that the total charge within the volume $Q_{\text{enc}}$, found by integrating the charge density $\rho(x)$ over the volume $V$ is equal to the normal component of the flux $D$ crossing the surface $\mathcal{S}$

$$Q_{\text{enc}} = \iint_{\mathcal{S}} \nabla \cdot D \, dV = \iint_{\mathcal{S}} D \cdot \hat{n} \, dA.$$

**Summary:** Maxwell’s four equations relate the field strengths to the flux densities. There are two types of variables, field strengths ($E, H$) and flux densities ($D, B$). There are two classes, electric ($E, D$) and magnetic $H, B$. One might naturally view this as a 2x2 matrix, with rows being electric and magnetic strengths, and columns being electric and magnetic and flux densities, defining a total of four variables:

<table>
<thead>
<tr>
<th>Strength</th>
<th>Flux</th>
</tr>
</thead>
<tbody>
<tr>
<td>Electric</td>
<td>$E$ [V/m]</td>
</tr>
<tr>
<td>Magnetic</td>
<td>$H$ [A/m]</td>
</tr>
</tbody>
</table>

Applying Stokes’ curl law to the forces induces a Thévenin voltage (emf) or Norton current source. Applying Gauss’ divergence law to the flows, gives the total charge enclosed. The magnetic charge always zero since $\nabla \cdot B = 0$, because magnetic mono-poles do not exist. However magnetic dipoles do exist, as in the example of the electron which contains a magnetic dipole.
CHAPTER 1. INTRODUCTION

The wave equation: When Maxwell’s equations are combined the vector wave equation results. There is the famous story that Maxwell introduce the displacement current \( \dot{D} \) to make the equations symmetric, allowing him to predict the formula for the speed of light. The displacement current was not verified by the experiments of Faraday and Ampere.

Taking the curl of Eq. EF [Maxwell’s Electric Field strength equation for the \( \nabla \times E \) (Fig. 1.6)], we obtain in the electric vector wave equation

\[
\nabla \times \nabla \times E = -\nabla \times \dot{B} = -\mu_0 \nabla \times \dot{H} = -\mu_0 \varepsilon_0 \ddot{E}.
\]

Using the GoD identity (Eq. 1.189) gives

\[
\nabla^2 E - \frac{1}{\varepsilon_0} \nabla(\nabla \cdot D) = \mu_0 \varepsilon_0 \ddot{E}.
\]

In a charge free region \( \nabla \cdot D = 0 \). Since \( 1/c^2 = \mu_0 \varepsilon_0 \), the vector wave equation results:

\[
\nabla^2 E(x, t) = \frac{1}{c^2} \ddot{E}(x, t) \leftrightarrow \frac{s^2}{c^2} E(x, \omega).
\]

It is important to distinguish the vector Laplacian from the scalar Laplacian. As was discussed in Table 1.3 (p. 130), Recall the d’Alembert solutions of the scalar wave equation (Eq. 1.95, p. 112)

\[
E(x, t) = f(x - ct) + g(x + ct),
\]

where \( f, g \) are arbitrary vector fields.

In a like manner one may derive the wave equation in terms of \( H \)

\[
\nabla^2 H(x, t) = \frac{1}{c^2} \ddot{H}(x, t) \leftrightarrow \frac{s^2}{c^2} H(x, \omega)
\]

This equation does not have the restriction that there is no free charge, because \( \nabla \cdot B = 0 \). Thus both \( E, H \) obey the wave equation (thus they are locked together in space-time if we assume no free charge) (Sommerfeld, 1952).

Poynting’s theorem: The energy flux density \( \mathcal{P} \) [W/m²] is perpendicular to \( E \) and \( B \), denoted as

\[
\mathcal{P} = \frac{1}{\mu_0} E \times B.
\]

Electrical impedance seen by an electron: Up to now we have only considered the Brune impedance which is a special case with no branch points or branch cuts. We can define impedance for the case of diffusion, as in the case of the diffusion of heat. There is also the diffusion of electrical and magnetic fields at the surface of a conductor, where the resistance of the conductor dominates the dielectric properties, which is called the electrical skin effect, where the conduction currents are dominated by the conductivity of the metal rather than the displacement currents. In such cases the impedance is proportional to \( \sqrt{s} \), implying that it has a branch cut. Still in this case the real part of the impedance must be positive in the right-half s plane, the required condition of all impedances, such that postulate P3 (p. 104) is satisfied.

Example: When we deal with Maxwell’s equations the force is defined by the Lorentz force

\[
f = qE + qv \nabla \times B = qE + C \times B,
\]
which is the force on a charge (e.g., electron) due to the electric $E$ and magnetic $B$ fields. The magnetic field plays a role when the charge has a velocity $\mathbf{v}$. When a charge is moving with velocity $\mathbf{v}$, it may be viewed as a current $C = q\mathbf{v}$.

In this case the impedance in a wire, where the current is constrained, the complex impedance density is

$$Z(s) = \sigma + s\epsilon_0, \quad [\Omega/m^2]$$

which when integrated over an area is the impedance in Ohms (Feynman, 1970c, p. 13-1). Here $\sigma$ is the electrical conductivity and $\epsilon_0$ is the electrical permittivity. Since $\sigma \gg \omega\epsilon_0$, this reduces to the resistance of the wire, per unit length.

1.6.14 Lec 42 The Quasi-static approximation

There are a number of assumptions and approximations that result in special cases, many of which are classic. These manipulations are all done at the differential equation level, by making assumptions that change the basic equations that are to be solved. These approximations distinct from assumptions made while solving a specific problem.

A few and important examples include

1. *In vacuo* waves (free-space scalar wave equation)

2. Expressing the vector wave equation in terms of scalar and vector potentials

3. Quasi-statics
   
   (a) scalar wave equation
   
   (b) Kirchhoff’s low-frequency lumped approximation (LRC networks)
   
   (c) Transmission line equations (Telephone and Telegraph equations)

One of the very first insights into wave propagation was due to Huygens (c1640) (Fig. 1.25). 

**Quasi-statics and its implications:** The term *quasi-statics* (Postulate P9, p. 105) is an approximation, used to reduce a partial differential equations to a scalar (one-dimensional) equation (Sommerfeld, 1952). Quasi-statics is a way of reducing a three-dimensional problem to a one-dimensional problem. So that it is not miss-applied, it is important to understand the nature of this approximation, which goes to the heart of transmission line theory. The quasi-static approximation states that the wavelength $\lambda$ is greater than the dimensions of the size of the object $\Delta$ (e.g., $\lambda \gg \Delta$). The best known example, Kirchhoff’s current and voltage laws, KCL and KVL, almost follow from Maxwell’s equations given the quasi-static approximation (Ramo *et al.*, 1965). These laws state that the sum of the currents at a node must be zero (KCL) and the some of the voltages around a loop must be zero (KCL).

These well known laws the analogue of Newton’s laws of mechanics. The sum of the forces at a point is the analogue of the sum of the voltages. Voltage $\phi$ is the force potential, since the electric field $E = -\nabla \phi$. The sum of the currents is the analogue of the vector sum of velocities at a point is zero.

The acoustic wave equation describes how the scalar field pressure $p(x, t)$, the vector force density potential $(f(x, t) = -\nabla p(x, t) [N/m^2])$, propagates in three dimensions. (The net force is the integral of the pressure gradient over an area.) If the wave propagation is restricted to a pipe
(e.g., organ pipe), or to a string (e.g., an guitar or lute), the transverse directions may be ignored, due to the quasi-static approximation. What needs to be modeled by the equations is the wave propagation along the pipe (string). Thus we may approximate the restricted three-dimensional wave by a one-dimensional wave.

However if we wish to be more precise about this reduction in geometry ($\mathbb{R}^2 \to \mathbb{R}$), we need to consider the quasi-static approximation, as it makes assumptions about what is happening in the other directions, and quantifies the effect ($\lambda \gg \Delta$). Taking the case of wave propagation in a tube, say the ear canal, there is the main wave direction, down the tube. But there is also wave propagation in the transverse direction, perpendicular to the direction of propagation. As shown in Table 3.2 (p. 229), the key statement of the quasi-static approximation is that the wavelength in the transverse direction is much larger that the radius of the pipe. This is equivalent to saying that the radial wave reaches the walls and is reflected back, in a time that is small compared to the distance propagated down the pipe. Clearly the speed of sound down the pipe and in the transverse direction is the same if the medium is homogeneous (i.e., air or water). Thus the sound reaches the walls and is returned to the center line in a time that the axial wave traveled about 1 diameter along the pipe. So if the distance traveled is several diameters, the radial parts of the wave have time to come to equilibrium. So the question one must ask is, what are the conditions of such an equilibrium. The most satisfying answer to this is to look at the internal forces on the air, due to the gradients in the pressure.

The pressure $p(x, t)$ is a potential, thus its gradient is a force density $f(x, t) = -\nabla p(x, t)$. What this equation tells us is that as the pressure wave approaches that of a plane wave, the radial (transverse) forces go to zero. If the tube has a curvature, or a change in area, then there will be local forces that create radial flow. But after traveling a few diameters, these forces will come to equilibrium and the wave will return to a plane wave. The internal stress caused by a change is area must settle out very quickly. There is a very important caveat however: it is only at low frequencies that the plane wave can dominate. At frequencies such that the wavelength is very small compared to the diameter, the distance traveled between reflections is much greater than a few diameters. Fortunately the frequencies where this happens are so high that they play no role in frequencies that we care about. This effect is referred to as cross-modes which imply some sort of radial standing waves. In fact such modes exist in the ear canal, but on the eardrum where the speed of sound is much slower that that of air. Because of the slower speed, the ear drum has cross-modes, and these may be seen in the ear canal pressure. Yet they seem to have a negligible effect on our ability to hear sound with good fidelity. The point here is that the cross modes are present, but we call upon the quasi-static approximation as a justification for ignoring them, to get closer to the first-order physics.

**Quasi-statics and Quantum Mechanics**

It is important to understand the meaning of Planck’s constant $\hbar$, which appears in the relations of both photons (light “particles”) and electrons (mass particles). If we could obtain a handle on what exactly Planck’s constant means, we might have a better understanding of quantum mechanics, and physics in general. By cataloging the dispersion relations (the relation between the wavelength $\lambda(\nu)$ and the frequency $\nu$), for between electrons and photons, this may be attainable.

Basic relations from quantum mechanics for photons and electrons include:

1. Photons (mass=0, velocity = $c$)

   (a) $c = \lambda \nu$: The speed of light $c$ is the product of its wavelengths $\lambda$ times its frequency $\nu$. 


This relationship is only for mono-chromatic (single frequency) light.

(b) The speed of light is

\[ c = \frac{1}{\sqrt{\mu_0\varepsilon_0}} = 0.3 \times 10^6 \text{ [m/s]} \]

(c) The characteristic resistance of light \( r_o = \sqrt{\mu_0/\varepsilon_0} = |E|/|H| = 377 \text{ [ohms]} \) is defined as the magnitude of the ratio of the electric \( E \) and magnetic \( H \) field, of a plane wave in-vacuo.

(d) \( E = h\nu \): the photon energy is given by Planck’s constant \( h \approx 6.623 \times 10^{-34} \text{ [m}^2\text{kgm/s]} \), times the frequency (or bandwidth) of the photon

2. Electrons (mass = \( m_e \), velocity \( V = 0 \)):

(a) \( E_e = m_c c^2 \approx 0.91 \cdot 10^{-30} \cdot 0.3^2 \cdot 10^{12} = 8.14 \times 10^{-20} \text{ [J]} \) is the electron rest energy (velocity \( V = 0 \)) of every electron, of mass \( m_e = 9.1 \times 10^{-31} \text{ [kgm]} \), where \( c \) is the speed of light.

(b) \( p = h/\lambda \): The momentum \( p \) of an electron is given by Planck’s constant \( h \) divided by the wavelength of an electron \( \lambda \). It follows that the bandwidth of the photon is given by

\[ \nu_e = \frac{E_e}{h} \]

and the wavelength of an electron is

\[ \lambda_e = \frac{h}{p_e} \]

One might reason that QM obeys the quasi-static (long wavelength) approximation. If we compare the velocity of the electron \( V \) to the speed of light \( c \), then we see that

\[ c = E/p \gg V = E/p = mV^2/V \]

**Conjecture on photon energy:**

Photons are seen as quantized because they are common generator by atoms, which produce light-particles having the difference in two energy (quantum, or eigen-states) levels. The relation \( E = h\nu \) does not inherently depend on \( \nu \) being a fixed frequency. Planck’s constant \( h \) is the EM energy density over frequency, and \( E(\nu_o) \) is the integral over frequency

\[ E(\nu_o) = h \int_{-\nu_o}^{\nu_o} d\nu = 2h\nu_o. \]

When the photon is generated by an atom \( \nu_o \) is quantized by the energy level difference that corresponds to the frequency (energy level difference) of the photon jump.
1.6.15 Lec 43: Review for the Final Exam

Summary

Mathematics began as a simple way of keeping track of how many things there were. But eventually physics and mathematics evolved together as tools to help us navigate our environment, not just physically around the globe, but how to solve daily problems such as food, water and waste management, understand the solar system and the stars, defend ourselves, use tools of war, etc.

Based on the historical record of the abacus, one can infer that people precisely understood the concept of counting, addition, subtraction and multiplication (recursive addition).

There is some evidence that the abacus, a simple counting tool, formalizing the addition of very large numbers, was introduced to the Chinese by the Romans, where it was used for trade. However this working knowledge of arithmetic did not show up in written number systems. The Roman numerals were not useful for doing calculations done on the abacus. The final answer would then be expressed in terms of the Roman number system.

According to the known written record, the number zero (null) had no written symbol until the time of Brahmagupta (628 CE). One should not assume the concept of zero was not understood simply because there was no symbol for it in the Roman Numeral system. Negative numbers and zero would be obvious when using the abacus. Numbers between the integers would be represented as rational numbers \(\mathbb{Q}\) since any number may be approximated with arbitrary accuracy with rationals.

Mathematics is the science of formalizing a repetitive method into a set of rules, and then generalizing it as much as possible. Generalizing the multiplication and division algorithm, to different types of numbers, becomes increasingly more complex as we move from integers to rational numbers, irrational numbers, real and complex numbers and ultimately, vectors and matrices. How do you multiply two vectors, or multiply and divide one matrix by another? Is it subtraction as in the case of two numbers? Multiplying and dividing polynomials (by long division) generalizes these operations even further. Linear algebra is further important generalization, fallout from the fundamental theorem of algebra, and essential for solving the generalizations of the number systems.

Many of the concepts about numbers naturally evolved from music, where the length of a string (along with its tension) determined the pitch (Stillwell, 2010, pp. 11, 16, 153, 261). Cutting the string’s length by half increased the frequency by a factor of 2. One forth of the length increases the frequency by a factor of 4. One octave is a factor of 2 and two octaves a factor of 4 while a half octave is \(\sqrt{2}\). The musical scale was soon factored into rational parts. This scale almost worked, but did not generalize (sometimes known as the Pythagorean comma\(^{106}\)), resulting in today’s well tempered scale, which is based on 12 equal geometric steps along one octave, or 1/12 octave (\(\frac{12}{\sqrt{2}} \approx 1.05946 \approx 18/17 = 1 + 1/17\)).

But the concept of a factor was clear. Every number may be written as either a sum, or a product (i.e., a repetitive sum). This led the early mathematicians to the concept of a prime number, which is based on a unique factoring of every integer. At this same time (c5000 BCE), the solution of a second degree polynomial was understood, which lead to a generalization of factoring, since the polynomial, a sum of terms, may be written in factored form. If you think about this a bit, it is sort of an amazing idea, that needed to be discovered (Stillwell, 2010, p. ). This concept lead to an important string of theorems on factoring polynomials, and how to numerically describe physical quantities. Newton was one of the first to master these tools with his proof that the orbits of the

\(^{106}\)https://en.wikipedia.org/wiki/Pythagorean_comma
planets are ellipses, not circles. This lead him to expanding functions in terms of their derivatives and power series. Could these sums be factored? The solution to this problem led to calculus.

So mathematics, a product of the human mind, is a highly successful attempt to explain the physical world. All aspects of our lives were impacted by these tools. Mathematical knowledge is power. It allows one to think about complex problems in increasingly sophisticated ways. An equation is a mathematical sentence, expressing deep knowledge. Witnessed $E = mc^2$ and $\nabla^2 \psi = \ddot{\psi}$.

Reading List: The above concepts come straight from mathematical physics, as developed in the 17th–19th centuries. Much of this was first developed in acoustics by Helmholtz, Stokes and Rayleigh, following in Green’s footsteps, as described by Lord Rayleigh (1896). When it comes to fully appreciating Green’s theorem and reciprocity, I have found Rayleigh (1896) to be a key reference. If you wish to repeat my reading experience, start with Brillouin (1953, 1960), followed by Sommerfeld (1952); Pipes (1958). Second tier reading contains many items Morse (1948); Sommerfeld (1949); Morse and Feshbach (1953); Ramo et al. (1965); Feynman (1970a); Boas (1987). A third tier might include Helmholtz (1863a); Fry (1928); Lamb (1932); Bode (1945); Montgomery et al. (1948); Beranek (1954); Fagen (1975); Lighthill (1978); Hunt (1952); Olson (1947). It would be a mistake to ignore other massive physics writings by stalwart authors, J.C. Slater and Landau and Lifshitz, and their impressive series of mathematical physics books.

You must enter at a level that allows you to understand. Successful reading of these books critically depends on what you already know. A rudimentary (high school) level of math comprehension must be mastered first. Read in the order that helps you best understand the material.

Without a proper math vocabulary, mastery is hopeless. I suspect that one semester of college math can bring you up to speed. This book is my attempt to present this level of understanding.

1.6.16 Lec 44: Fun speculations and loose ends: 5.4 p. 239

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Chapter 2

Number Systems: Stream 1

Chapters 2-5 contain advanced material that go beyond Chapter 1. You should start with Chapter 1, and once mastered, then move to the rest of the book.

This chapter (2) is devoted to Number Systems (Stream 1), starting with the counting numbers \( \mathbb{N} \). In this chapter we delve more deeply into the details of the topics of Chapter 1, Lectures 4-9.

2.1 Week 2

In Section 1.3.3 we explore in more detail the two fundamental theorems of prime numbers, working out a sieve example, and explore the logarithmic integral \( Li(N) \) which approximates the density of primes \( \rho_k(N) \) up to prime \( N \).

The topics of Section 1.3.4 consider the practical details of computing the greatest common divisor (GCD) of two integers \( m, n \) (Matlab’s routine \( 1=gcd(m,n) \)), with detailed examples and comparing the algebraic and matrix methods. Homework assignments will deal with these two methods. Finally we explore the relationship between coprimes and the GCD. In Section 1.3.5 we defined the Continued Fraction algorithm (CFA), a method for finding rational approximations to irrational numbers. The CFA and GCD are closely related, but the relation needs to be properly explained. In Section 1.3.6 we derive Euclid’s formula, the solution for the Pythagorean triplets (PT), based on Diophantus’s chord–tangent method. This method is used many times throughout the course notes, first for computing Euclid’s formula for the PTs, then for finding a related formula in Section 1.3.7 for the solutions to Pell’s equation, and finally for finding the mapping from the complex plane to the extended complex plane (the Riemann sphere).

Finally in Section 1.3.8 the general properties of the Fibonacci sequence is discussed. This equation is a special case of the second order digital resonator (well known in digital signal processing), so it has both historical and practical application for engineering. The general solution of the Fibonacci is found by taking the Z-transform and finding the roots, resulting in an eigenvalue expansion (Appendix E).

2.1.1 Lec 4 Two theorems on primes

Theorem 1: Fundamental Theorem of Arithmetic

Factoring integers: Every integer \( n \in \mathbb{N} \) has a unique factorization (Stillwell, 2010, p. 43) (Eq. 1.6, p. 42).
Cofactors: Integers 2312 and 2313 are said to be coprime, since they have no common factors. Coprimes may be identified via the greatest common divisor:
\[
gcd(a, b) = 1
\]
using the Euclidean algorithm (Stillwell, 2010, p. 41).

**Theorem 2: Prime Number Theorem**

The primes are a random field since there is no way to predict when the next prime will occur. Thus one needs to use statistical methods to characterize their density. Based on a sample of approximately 3 million primes, Gauss showed empirically that the average total number of primes less than \( N \) is

\[
\sum_{n=1}^{N} \delta_n \sim \frac{N}{\ln N}.
\] (2.1)

These primes were obtained by manual calculations “as a pastime” in 1792-3 (Goldstein, 1973).

Define \( \delta_n = 1 \) if \( n \) is a prime, and zero otherwise.\(^1\)

It follows that the average density of primes is \( \rho_\pi(N) \sim 1/\ln N \), thus

\[
\rho_\pi(N) \equiv \frac{1}{N} \sum_{n=1}^{N} \delta_n \approx \frac{1}{N} \int_2^{N} \frac{d\xi}{\ln \xi},
\] (2.2)

where \( Li(N) \) is the offset logarithmic integral (Stillwell, 2010, p. 585). The primes are distributed as \( 1/\ln(n) \) since the average total number of primes is proportional to the logarithmic integral \( Li(n) \) (Goldstein, 1973; Fine, 2007).

Here is a Matlab/Octave code that tests this formula:

```matlab
%Computes density of primes from average density
NP=1e6; % 10^6 primes
p=primes(NP); %compute primes
delta=zeros(1,NP); delta(p)=1; %put 1 at each prime
PI=cumsum(delta); %Number of primes vs N
rho=PI./(1:NP); %estimate of the density of primes= PI(N)./N

figure(1)
semilogy(rho); %plot of density vs number of primes
title('Density of primes vs. N');
ylabel('\rho(N)'); xlabel('N'); grid on
```

From the Prime Number Theorem it is clear that the density of primes is large (they are not scarce). As best I know there are no methods to find primes other than by the sieve method (Section 1.3.3, p. 40). If there is any good news it is that they only need to be computed once, and saved. In practical applications this may not help much, given their large number. But I suspect they could easily be saved on a modern disk (e.g., 1 TB).

Not surprisingly, playing with primes has been a popular pastime of mathematicians. Perhaps this is because those who have made inroads, providing improved understanding, have become famous.

\(^1\)You may view \( \delta_n \) for the first 100 numbers with the one-line Matlab/Octave command `stem(isprime(1:100))`
2.1.2 RSA public-private keys

Internet security depends on a public-private key system, called RSA\(^2\), which is built on the difficulty of factoring large primes. When a forward computation is easy, such as multiplying two primes, is easy, and the inverse problem (factoring a product into two primes) is hard, it is called a trap door.

What makes RSA work is a trap door. Internet security is not as important as global warming, but its pretty important. Global warming has no “end game.” Internet security does, redesign the internet.

Let’s consider the problem of factoring two primes, given the product \(N = \pi_k \pi_l\), \(k \neq l\). Let’s further assume we have list of all the primes \(\{\pi_1, ..., \pi_m\}\) including \(N < \pi_m\). To factor \(N\) we must divide \(N\) by each prime less than \(N\). With intelligent sampling, on average, we would find the prime factor in \(N/2\) tries. In theory, given primes \(\pi_n\) up to \(n = N\), the density \(\rho_\pi(N)\) could help one search for a particular prime of known size \(N\), by estimating how many primes there are in the neighborhood of \(N\). The cost of factoring any integer \(\in \mathbb{N}\) would be related to the magnitude of \(N = \pi_k \pi_l\), where \(N\) is the size of the private key, assuming a simple product key. Worst case an exhaustive search would take \(m - 1\) divides of the form \(N/\pi_k\) \(k = 1 \cdots m - 1\). If one knew approximately where the factor is \(\in \mathbb{P}\) (i.e., \(k\) of \(\pi_k\)), and knows one of the primes, the search could be done with an exhaustive search, by addition. The number of primes less that \(N\) may be computed using Gauss’ formula for the density of primes.

So the question remains: How does internet security actually work? Unfortunately at this time I can not give you a proper summary. But it is more complex than I have let on. The full answer requires a proper course in number theory and internet security, beyond what can be presented.\(^3\)

To save you the trouble of looking up the method, it is quoted here\(^4\).

Setting up an RSA system involves choosing large prime numbers \(p\) and \(q\), computing \(n = pq\) and \(k = \phi(n)\), and finding two numbers \(e\) and \(d\) such that \(ed \equiv 1 \pmod{k}\).

The numbers \(n\) and \(e\) (the “encryption key”) are released to the public, and \(d\) (the “decryption key”) is kept private.

A message, represented by an integer \(m\), where \(0 < m < n\), is encrypted by computing \(S = me \pmod{n}\). It is decrypted by computing \(t = Sd \pmod{n}\). Euler’s Theorem can be used to show that if \(0 < t < n\), then \(t = m\).

The security of an RSA system would be compromised if the number \(n\) could be factored or if \(\phi(n)\) could be computed without factoring \(n\).

The power of this security device is based on the relatively high density of primes \(\mathbb{P} \subset \mathbb{N}\), which is addressed by the Prime Number Theorem.

Example: The RSA public key is a string of characters representing a prime number \(\pi_{pub}^{}\):

\begin{verbatim}
AAAAB3NzaC1yc2EAAADAQABAAABAQCsyfp9uDXubR/ukXMpVqk2EtttQCeJEmg6mC7qjJa2hrjFCV1+rYSDgqKddkEfHL+
H13V4yTb09m4CmGFbPMKj3HyH1nQw6hUdhOhTQnoCp517WDpEeqIQCPe21JMs+3Y17b05hNMv9W65qzFCNmtJr11VXkcU
Xnrbb/bbIMdsAGGnViylup+p8rsNIq/i0h8WQcHGMn0h7P5Y/bJdgYWZz/9t5MoTNeuq7jg9nPOJ6aqUaFTXAg00SBKXyUBi
\end{verbatim}

\(^2\)https://en.wikipedia.org/wiki/RSA_(cryptosystem)
\(^3\)https://en.wikipedia.org/wiki/RSA_(cryptosystem)#Operation
\(^4\)https://en.wikipedia.org/wiki/Euler\%27s_totient_function#The_RSA_cryptosystem
The RSA private key is a second string of characters, representing the product of prime numbers \( \pi_{pri} \):

\[
\text{MIIEpAIBAAKCAQEArMn6fbg17m0f7pFzKVYCtnhLbUXChxCDIgpgu60Wtoa4RQ1dfg/KENanCnZJHxxy/hyNiQGMr4RpeJuAphhWzwPzMaVyc9Z20OMoosVHYRTrqwpw0de186BQriAe3tjptzYs3ptU2N90YTTL/VukgsqQjrs5a9vdV5HFF56zw2y2yDLOgBhplyIs7i7gfvK0J2a1tab/PcEHBxj9J1eK+WP2yXYGFs1/beTKEZxqru44Iz9Cemq1A0lwKtNEG5818IDootxk0jJ/J9YR85Ay
\]

This much longer string of character represents a second prime number \( \pi_{pri} \), the private key.

If I gave you the ASCII string that represents the product of the two prime numbers \( \pi_k \pi_l \), along with \( \pi_k \), can you find \( \pi_l \)? Yes, divide one by the other, with remainder zero. If I ask you to find \( \pi_l \) given \( \pi_k \pi_l \), you would need to call the NSA, or be some sort of expert on factoring primes. Given \( \pi_k \pi_l \), for some large values of \( k \) and \( l \), the GCD is not helpful, nor the CFA. While the factoring can be done (it is not impossible), it is a very expensive calculation.

Why not use Matlab’s \texttt{factor(N)} routine to find the factors? This is where cost comes in. If the primes are large enough, even the NSA cannot do it in a lifetime. The numbers used in RSA are too large for Matlab’s \texttt{factor(N)} to deliver an answer. In fact, even the largest computer in the world (such as the University of Illinois’ super computer (NCSA Water) cannot do this computation. But, for a carefully chose set of primes, not too large for the NCSA to factor them, they can decrypt a message, but you cannot. This is why they pay for these massive super-computers.

As best I know you need to try one prime at a time to see if the div is or gives you a zero remainder. There are no hints, such as \( l \approx k \). Nothing to help but to grind your way through all the known primes. And if the primes are too large to be on the list of all known primes, then you have the additional burden of generating a list of primes long enough to contain the ones you are looking for. You can be sure that the NSA (or whoever has the fastest computer) has a longer list than you do, and its getting longer every day.

What you are given is the public key (prime \( \pi_{pub} \)) (the shorter one), and you need to guess the private key \( \pi_{pri} \) and decrypt the message I send you with the product \( \pi_{pub} \cdot \pi_{pri} \).

### 2.1.3 Lec 5 Greatest common divisor (GCD)

#### Division with rounding method

This method implements \( \text{gcd}(a, b) \). In matrix form we have

\[
\begin{bmatrix}
  m_{k+1} \\
  n_{k+1}
\end{bmatrix} =
\begin{bmatrix}
  0 & 1 \\
  1 & -m/n
\end{bmatrix}
\begin{bmatrix}
  m_k \\
  n_k
\end{bmatrix}.
\]

(2.3)

This starts with \( k = 0, m_0 = a, n_0 = b \). With this method there is no need to test if \( n_k < m_n \), as it is built into the procedure. The method uses the floor function \( \lfloor x \rfloor \), which finds the integer part.
of $x$ ($\lfloor x \rfloor$ rounds toward $-\infty$). Following each step we will see that the value $n_{k+1} < m_{k+1}$. The method terminates when $n_{k+1} = 0$ with $\gcd(a, b) = m_{k+1}$.

Below is 1-line vectorized code that is much more efficient than the direct matrix method:

```matlab
k = gcd(m,n) %entry point: input m,n; output k=gcd(m,n)
M=[abs(m),abs(n)]; %init M
while M(2) ~= 0 % < n*eps for irrational inputs %
  M = [M(1); M(2)] with M(1)<M(2) %
  M = [M(2) - M(1)*floor(M(2)/M(1)); M(1)];
end
```

With a minor extension in the test for “end,” this code can be made to work with irrational inputs (e.g., $(n\pi,m\pi)$).

This method calculates the number of times $n < m$ must subtract from $m$ using the floor function. This operation is the same as the mod function.\footnote{https://en.wikipedia.org/wiki/Modulo_operation} Specifically

$$n_{k+1} = m_k - \text{floor}\left(\frac{m}{n}\right)n_k$$ (2.4)

so that the output is the definition of the remainder of modular arithmetic. This would have been obvious to anyone using an abacus, which explains why it was discovered so early.

Note that the next value of $m$ ($M(1)$) is always less than $n$ ($M(2)$), and must remain greater or equal to zero. This one-line vector operation is then repeated until the remainder ($M(1)$) is 0. The gcd is then $n$ ($M(2)$). When using irrational numbers, this still works except the error is never exactly zero, due to IEEE 754 rounding. Thus the criterion must be that the error is within some small factor times the smallest number (which in Matlab is the number $\text{eps} = 2.220446049250313 \times 10^{-16}$, as defined in the IEEE 754 standard.)

Thus without factoring the two numbers, Eq. 2.4 recursively simply finds the gcd. Perhaps this is best seen with some examples.

The GCD is an important and venerable method, useful in engineering and mathematics, but, as best I know, is not typically taught in the traditional engineering curriculum.

**Graphical meaning of the GCD:** The Euclidean algorithm is very simple when viewed graphically. In Fig. 2.1 we show what is happening as one approaches the threshold. After reaching the threshold, the two number must be swapped, which is addressed by upper row of Eq. 2.3.

Multiplication is simply recursive addition, and finding the gcd takes advantage of this fact. Lets take a trivial example, (9,6). Taking the difference of the larger from the smaller, and writing multiplication as sums, helps one see what is going on. Since $6 = 3 \times 2$, this difference may be written two different ways

$$9 - 6 = (3 + 3 + 3) - (3 + 3) = 0 + 0 + 3 = 3,$$

or

$$9 - 6 = (3 + 3 + 3) - (2 + 2 + 2) = 1 + 1 + 1 = 3.$$
CHAPTER 2. NUMBER SYSTEMS: STREAM 1

Figure 2.1: The Euclidean Algorithm recursively subtracts \( n \) from \( m \) until the remainder \( m - kn \) is either less than \( n \) or zero. Note that this is similar to \( \text{mod}(m, n) \). The GCD recursively computes \( \text{mod} \), then swaps \( m, n \) so that \( n < M \), and repeats, until it finds the GCD. For the case depicted here the value of \( k \) that renders the remainder less than \( n \) is \( k = 6 \). If one more step were taken \( (k = 7) \) the remainder would become negative. By linear interpolation we can find that \( m - kn = 0 \) when \( k = \text{floor}(m/n) \). For this example \( k = 6.5 \), thus \( 6 = \text{floor}(m/n) < n \). This is nonlinear arithmetic, which is the key to the GCD.

Coprimes: When the gcd of two integers is 1, the only common factor is 1. This is of key importance when trying to find common factors between the two integers. When \( 1 = \text{gcd}(m,n) \) they are said to be coprime, which can be written as \( m \perp n \). By definition, the largest common factor of coprimes is 1. But since 1 is not a prime (\( \pi_1 = 2 \)), they have no common primes. It can be shown (Stillwell, 2010, p. 41-42) that when \( a \perp b \), there exist \( m, n \in \mathbb{Z} \) such that

\[
am + bn = \text{gcd}(a,b) = 1.
\]

Generalizations of GCD: The GCD may be generalized in several significant ways. For example what is the GCD of two polynomials? To answer this question one must factor the two polynomials to identify common roots.

2.2 Week 3

2.2.1 Lec 6 Continued Fraction Expansion (CFA)

Continued Fractions and circuit theory: One of the most powerful generalizations of the CFA seems to be the expansion of a function of a complex variable, such as the expansion of an impedance \( Z(s) \), as a function of complex frequency \( s \), as described in Fig. 2.2 and Eq. 2.5. This is especially interesting in that it leads to a physical interpretation of the impedance in terms of a transmission line (horn), a structure well known in acoustics as having a variable area \( A(x) \) as function of the range variable \( x \).

The CFA expansion is of great importance in circuit theory, where it is equivalent to an infinitely long segment of transmission line, composed of series and shunt impedance elements. Thus such a cascade network composed of 1 ohm resistors, has an input impedance of \( (1 + \sqrt{5})/2 \approx 1.6180 \) [ohms] Eq. 1.8 (p. 47).

The CFA may be extended to monomials in \( s \). For example consider the input impedance of a cascade L-C transmission line as shown in Fig. 2.2. The input impedance of this transmission line
is given by a continued fraction expansion of the form

\[ Z_{in} = sL + \frac{1}{sC + \frac{1}{sL + \frac{1}{sC + \ldots}}} =: [sL, sC, sL, sC, \ldots]. \tag{2.5} \]

In some vague way, Eq. 2.5 is reminiscent of a Taylor series expansion about \( s = 0 \), yet very different. In the limit, as the frequency goes to zero (\( s \to 0 \)), the impedance of the inductors go to zero, and that of the capacitors go to \( \infty \). In physical terms, the inductors become short circuits, while the capacitors become open circuits. The precise relation may be quantified by the use of composition, described in Fig. 1.18 (p. 92). Specifically

\[
\begin{pmatrix} P_1 \\ U_1 \end{pmatrix} = \begin{pmatrix} 1 & sL \\ 0 & 1 \end{pmatrix} \begin{pmatrix} 1 & 0 \\ sC & 1 \end{pmatrix} \cdots \begin{pmatrix} 1 & sL \\ 0 & 1 \end{pmatrix} \begin{pmatrix} 1 & 0 \\ sC & 1 \end{pmatrix} \begin{pmatrix} P_2 \\ -U_2 \end{pmatrix}. \tag{2.6} \]

It seems possible that this is the CFA generalization of the Taylor series expansion, built on composition. If we were to do the algebra we would find that \( A(s), B(s), C(s), D(s) \) (i.e., Sections 1.4.11 (pp. 92, 208)) are ratios of polynomials having rational expansions as Taylor series. This seems like an important observation, that should have applications beyond the engineering literature (Campbell, 1903; Brillouin, 1953; Ramo et al., 1965). Note that Brillouin (1953) credits Campbell (1903).

Figure 2.2: This transmission line (TL) is known as a low-pass filter wave-filter (Campbell, 1922), composed of series inductors having an impedance \( Z_k = sL \) and shunt admittance \( Y_k = sC \). The characteristic resistance of the TL is \( r_0 = \sqrt{Z_k/Y_k} \) and the wave velocity is \( \kappa = \sqrt{Z_kY_k} = s/\sqrt{LC} = s/c \). Thus the wave velocity is \( c = 1/\sqrt{LC} \).

The length \( \Delta \) defines the physical size of each cell. For long wavelengths the wave-filter acts as a pure delay. But when the frequency increases above a cutoff frequency \( f_c > c/\Delta \), the wavelength \( \lambda = c/f \) is less than the size of a section \( \Delta \), and the system response becomes very high-order low-pass filter.

Figure 2.6 is a piece of wire having a delay determined by its velocity and the length, in units of the cells, each of length \( \Delta \). There are two basic parameters that characterize a transmission line, the characteristic resistance \( r_0 = \sqrt{sL/sC} = \sqrt{L/C} \), and wave number depends on the product of \( L \) and \( C \)

\[
\kappa = \sqrt{sL \cdot sC} = s\sqrt{LC} = s/c_0, \quad [\text{m}^{-1}] \tag{2.7}
\]

thus \( c_0 = 1/\sqrt{LC} \). The delay in each cell is

\[ \tau_o = \Delta/c_0 = \Delta\sqrt{LC} \quad [\text{s}]. \]

The characteristic wave resistance (aka surge resistance) is given by the ratio of \( L \) and \( C \)

\[ r_0 = \sqrt{L/C} \quad [\Omega]. \tag{2.8} \]
The total delay of the line is $T = \tau_0 \text{[s]}$, where $N$ is the number of sections. In that limit the waves travel as

$$f(t - x/c_0) = e^{-\kappa x} e^{-st},$$

as $\Delta \to 0$. When $L(x)$ and $C(x)$ depend on position, the transmission line is called a horn (Sect. ??, p.??, and described by the Webster Horn equation (Eq.1.127, p. 140).

Since the CFA has a physical representation as a transmission line, as shown in Fig. 2.2, it can be of high utility for the engineer. The theory behind this will be discussed in greater detail in Chapter 5. If you're ready to jump ahead, read the important book Brillouin (1960) and the collected works of Campbell (1937).

2.2.2 Lec 7 Derivation of Euclid’s formula for Pythagorean triplets

Pythagorean triplets (PTs) have many applications in architecture and scheduling, which explains why they are important. For example, if one wished to construct a triangle made from bricks having a perfect 90° angle, then the materials need to be squared off as shown in Fig. 1.9 (p. 49). The lengths of the sides need to satisfy PTs.

**Derivation of Euclid’s formula:** Euclid’s formula (Eq. 1.9, p. 48) provides integer solutions to the Pythagorean theorem (Eq. 1.1, p. 18). The solution method, said to be due to Diophantus, is called a chord/tangent method (Stillwell, 2010, p. 48). The method composes (Section 1.4.8, p. 80) a line and a circle, where the line defines a chord within the circle (its not clear where the tangent line might go). The slope of the line (i.e., denoted $t$) is then taken to be rational, allowing one to determine integer solutions of the intersections points. This solution for Pythagorean triplets $[a, b, c]$ is known as Euclid’s formula (Eq. 1.9, p. 48) (Stillwell, 2010, p. 4–9, 222).

The derivation methods of Diophantus have been lost, but Fermat and Newton figured out what Diophantus must have done (Stillwell, 2010, p. 7, 49, 218). Since Diophantus worked before algebra was invented, he described all the equations in prose (Stillwell, 2010, p. 93).

**Derivation of Euclid’s formula:** The derivation is outlined in Fig. 2.3. Starting from two integers $[p > q \in \mathbb{N}]$, composing a line having a rational slope $t = p/q > 0$, with a circle (Stillwell, 2010, p. 6), provides the formula for the Pythagorean triplets.

The construction starts with a circle and a line, which is terminated at the point $(-1, 0)$. The slope of the line is a free parameter $t$. By composing the circle and the line (i.e., solving for the intersection of the circle and line), the formula for the intersection point $(a, b)$ may be determined in terms of $t$, which will then be taken as the rational slope $t = p/q \in \mathbb{Q}$.

In Fig. 2.3 there are three panels, two labeled “Proofs.” The Euclidean Proof shows the angle relationships of two triangles, the first an isosceles triangle formed by the chord, having slope $t$ and two equal sides formed from the radius of the circle, and a second right triangle having its hypotenuse as the radius of the circle and its right angle vertex at $(a, 0)$. As shown, it is this smaller right triangle that must satisfy Eq. 1.1. The inner right triangle has its hypotenuse $c$ between the origin of the circle (O) to the point $(a, b)$. Side $a$ forms the $x$ axis and side $b$ forms the $y$ ordinate. Thus by construction Eq. 1.1 must be obeyed.

---

6Continued fraction expansions of functions are known in the circuit theory literature as a Cauer synthesis (Van Valkenburg, 1964b).
Euclid’s formula for Pythagorean triplets \([a, b, c]\)

**Euclidean Proof:**
1) \(t = \frac{p}{q} \in \mathbb{Q}\)
2) \(a = p^2 - q^2\)
3) \(b = 2pq\)
4) \(c = p^2 + q^2\)

Diophantus’s Proof:
1) \(b(a) = t(a + c)\)
2) \(\zeta(t) = a + jb = \frac{1-t^2+jt}{1+t^2}\)
3) \(\zeta = |c|e^{i\theta} = |c|\frac{1+it}{1-it}\)
4) \(\zeta(t) \equiv a + \frac{ct}{1 + t^2} = \frac{c}{1 + t^2}\)

Pythagorean triplets:
1) \(t = \frac{p}{q} \in \mathbb{Q}\)
2) \(a = p^2 - q^2\)
3) \(b = 2pq\)
4) \(c = p^2 + q^2\)

Figure 2.3: Derivation of Euclid’s formula for the Pythagorean triplets \([a, b, c]\), based on a composition of a line, having a rational slope \(t = \frac{p}{q} \in \mathbb{Q}\), and a circle \(c^2 = a^2 + b^2\), \([a, b, c] \in \mathbb{N}\). This analysis is attributed to Diophantus (Di-o-phant’-us) (250 CE), and today such equations are called Diophantine (Di-o-phant’-ine) equations. PTs have applications in architecture and scheduling, and many other practical problems.

The formula for the line is \(b(a) = t(a + c)\), which goes through the points \((-c, 0)\) and \((a, b)\). Composing the line with the circle gives

\[
a^2 + (t(a + c))^2 = c^2
\]

\[
a^2 + t^2(a^2 + 2ac + c^2) = c^2
\]

\[
(1 + t^2)a^2 + 2ct^2a + c^2(t^2 - 1) = 0
\]

This last equation is a quadratic equation in \(a\). In some sense it is not really a quadratic equation, since we know that \(a = -c\) is a root. Dividing by \(1 + t^2\)

\[
\frac{a^2 + 2ct^2}{1 + t^2}a + \frac{c^2(t^2 - 1)}{1 + t^2} = 0,
\]

makes it easier to complete the square, delivering the roots:

\[
\left(\frac{a}{1 + t^2} + \frac{ct^2}{1 + t^2}\right)^2 - \left(\frac{ct^2}{1 + t^2}\right)^2 + \frac{c^2(t^2 - 1)}{1 + t^2} = 0
\]

\[
\left(\frac{a}{1 + t^2} + \frac{ct^2}{1 + t^2}\right)^2 - \frac{c^2t^4}{(1 + t^2)^2} + \frac{c^2(t^2 - 1)(t^2 + 1)}{(1 + t^2)^2} = 0
\]

\[
\left(\frac{a}{1 + t^2} + \frac{ct^2}{1 + t^2}\right)^2 - \frac{c^2t^4 + c^2(t^2 - 1)}{(1 + t^2)^2} = 0
\]

\[
\left(\frac{a}{1 + t^2} + \frac{ct^2}{1 + t^2}\right)^2 = \left(\frac{c}{1 + t^2}\right)^2
\]

The second to last equation simplifies (magic happens) because the known root \(a = -c\) is embedded in the result.
Taking the square root gives the two roots
\[ a_\pm + \frac{ct^2}{1+t^2} = \pm \frac{c}{1+t^2} \]
\[ (1+t^2)a_\pm = -ct^2 \pm c = -c(t^2 \mp 1) \]
\[ a_\pm = -\frac{ct^2 \mp 1}{1+t^2}. \]

The known root is \( a_+ = -c \), because when the sign is +, the numerator and denominator terms cancel.

The root we have been looking for is \( a_- \)
\[ a_- = c \frac{1-t^2}{1+t^2}, \]
which allows us to solve for \( b_- \)
\[ b_- = \pm \sqrt{c^2 - a_-^2} \]
\[ = \pm c \sqrt{1 - \left( \frac{1-t^2}{1+t^2} \right)^2} \]
\[ = \pm c \sqrt{\frac{(1+t^2)^2 - (1-t^2)^2}{(t^2+1)^2}} \]
\[ = \pm \frac{2ct}{t^2+1}. \]

Therefore the coordinates \((a, b)\), the intersection point of the line and circle, are
\[ (a(t), b(t)) = c \frac{[1-t^2, 2t]}{1+t^2}. \]

To obtain the Pythagorean triplets, as given in Fig. 2.3 (p. 191) and Eq. 1.9 (p. 48), set \( t = p/q \), assuming \( p > q \in \mathbb{N} \), and simplify.

**Complex roots:** Defining the root as a complex number \( \zeta(\Theta) = a + bj \) forces \( a \perp b \) (i.e., forces the right triangle) and gives us polar coordinates, as defined by the figure as the Euclidean Proof
\[ \zeta(\Theta) = |c|e^{\Theta j} = |c| \left( \cos(\Theta) + j \sin(\Theta) \right). \]

This naturally follows since
\[ \zeta = |c|e^{\Theta(t)} = |c| \frac{1-t^2 + 2jt}{1+t^2} = |c| \frac{(1+j)^2(1+jt)}{(1+tj)(1-tj)} = (q+jp) \sqrt{\frac{q+jp}{q-pj}}. \]

Examples of PTs include \( a = 2^2 - 1^2 = 3, b = 2 \cdot 2 \cdot 1 = 4 \), and \( c = 2^2 + 1^2 = 5, 3^2 + 4^2 = 5^2 \).

Defining \( p = q+N \) \((p, q, N \in \mathbb{N})\) gives slightly better parametric representation of the answers, as the pair \((q, N)\) are a more systematic representation than \((p, q)\), because of the condition \( p > q \), so the general properties of the solutions are expressed more naturally. Note that \( b+c \) must always be a perfect square since \( b+c = (p+q)^2 = (2q+N)^2 \), as first summarized by Fermat (Stillwell, 2010, p. 212).
2.2. WEEK 3

2.2.3 Lec 8 Pell’s Equation

Calculation to show that Eqs. 1.10 and 1.11 are related. Starting from Eq. 1.10

\[ x_n^2 = (x_{n-1} + Ny_{n-1})^2 = x_{n-1}^2 + 2Nx_{n-1}y_{n-1} + N^2y_{n-1}^2 \]  \hspace{1cm} (2.9)

and

\[ Ny_{n}^2 = N(x_{n-1} + y_{n-1})^2 = x_{n-1}^2 + 2Nx_{n-1}y_{n-1} + N^2y_{n-1}^2 \]  \hspace{1cm} (2.10)

Subtracting removes the cross term leaving

\[ x_n^2 - Ny_n^2 = (1 - N)x_{n-1}^2 - N(1 - N)y_{n-1}^2 = (1 - N)(x_{n-1}^2 - Ny_{n-1}^2). \]

Using Eq. 1.12, the actual formula that follows from Eq. 1.11

\[ (-1)^n = (1 - N)(-1)^{n-1}, \]

thus (take \( n \) either even or odd) gives \( N = 2 \). Wallah.

**Eigen-value solution to Pell’s equation \((N = 2)\):** To provide a full understanding of what was known to the Pythagoreans about irrational numbers, it is helpful to provide the full solution to Eq. 1.11.

As shown in Fig. 1.11 (p. 51), \((x_n, y_n)\) may be written as a power series of the 2x2 matrix \(A\). The well known modern approach to find \(A^n\) is to diagonalize the matrix as detailed in Appendix D (p. 267). For the 2x2 matrix case, this is relatively simple. The final result written out in detail for the general solution \((x_n, y_n)\) (Appendix E, p. 271):

\[ \begin{bmatrix} x_n \\ y_n \end{bmatrix} = J^n \begin{bmatrix} 1 & 2 \\ 1 & 1 \end{bmatrix} \begin{bmatrix} 1 \\ 0 \end{bmatrix} = E \begin{bmatrix} \lambda_n^+ & 0 \\ 0 & \lambda_n^- \end{bmatrix} E^{-1} \begin{bmatrix} 1 \\ 0 \end{bmatrix}. \]  \hspace{1cm} (2.11)

The eigen-values are \( \lambda_{\pm} = j(1 \pm \sqrt{2}) \) while the eigen-matrix (Eq. D.1) and its inverse are

\[
\begin{bmatrix}
1 & 2 \\
1 & 1
\end{bmatrix}
\]

\[
\begin{bmatrix}
0.8165 & 0.8165 \\
0.5774 & -0.5774
\end{bmatrix},
\]

\[
\begin{bmatrix}
\sqrt{3} & \sqrt{2} \\
2\sqrt{2} & -\sqrt{2}
\end{bmatrix}
\]

\[
\begin{bmatrix}
0.6124 & 0.866 \\
0.6124 & -0.866
\end{bmatrix}
\]

The relative “weights” on the two eigen-solutions are equal, as determined by

\[
E^{-1} \begin{bmatrix} 1 \\ 0 \end{bmatrix} = \frac{\sqrt{3}}{2\sqrt{2}} \begin{bmatrix} 1 & \sqrt{2} \\ 1 & -\sqrt{2} \end{bmatrix} \begin{bmatrix} 1 \\ 0 \end{bmatrix} = \begin{bmatrix} 0.6124 & 0.866 \\ 0.6124 & -0.866 \end{bmatrix}
\]

We still need to prove that

\[
\frac{x_n}{y_n} \to \sqrt{N}.
\]

This follows intuitively from Pell’s equation, since as \((x_n, y_n) \to \infty\), the difference between \(x_n^2\) and \(2y_n^2\), the \((\pm 1)\) becomes negligible.

Given the development of linear algebra c19th century, this may be evaluated by eigen-vector diagonalization.\(^7\)

\(^7\)https://en.wikipedia.org/wiki/Transformation_matrix#Rotation
**Pell’s Equation and irrational numbers:** Since the eigen-values of Eq. 1.11 ($\lambda_{\pm} = 1 \mp \sqrt{N} \notin \mathbb{N}$), solutions to Pell’s equation raised the possibility that all numbers are not rational. This discovery of irrational numbers forced the jarring realization that the Pythagorean dogma “all is integer” was wrong. The significance of irrational numbers was far from understood.

**Derivation of Euclid’s formula for the Pell equation**

In this section we derive Euclid formula for the Pell’s equation, following the same reasoning as for the case of the Pythagorean triplets Eq. 1.9, following the logic described in Fig. 2.3.

**Proof I:**

1) $x^2 - Ny^2 = 1$
2) $y = t(x - 1)$
3) $(x, y) = \left(\frac{Nt^2 + 1, 2t}{Nt - 1}\right)$
4) $t = p/q$

Choose $q, n, N \in \mathbb{N}, p = q + n, r = Np^2 - q^2$

$x = \frac{(Np^2 + q^2)}{r}, y = \frac{2pq}{r}, \tan(\theta) = \frac{2p}{(Np^2 - q^2)}$

**Demo:** EvalPellEq.m

**Figure 2.4:** Derivation of general solution to Pell’s equation, using a chord/tangent composition.

The problem is to find integer solutions for $x, y \in \mathbb{N}$ to Pell’s equation (Eq. 1.10) for any $N \in \mathbb{N}$. The derivation is depicted in Fig. 2.4. Following the chord/tangent method due to Diophantus, we must compose a line (dashed) with a hyperbola, as shown in Fig. 2.4. The line must have a rational slope $t \in \mathbb{Q}$ and must pass through the trivial root $(1, 0)$. The insert “Proof I” gives (1) Pell’s equation, (2) the equation of the dashed line, the intersection point (3) $(x, y)$ and (4) the equation for the rational slope $t$ $(p, q \in \mathbb{Q})$. The step by step solution is given in Fig. 2.4.

The final solution, in terms of integers $[p, q], p > q$ is

$$x = \frac{Np^2 + q^2}{Np^2 - q^2}, \quad y = \frac{2pq}{Np^2 - q^2}.$$  \hfill (2.12)

Written as a complex numbers: $z = x + iy = (Np^2 + q^2 - j2qp)/(Np^2 - q^2)$
2.3 Week 4

2.3.1 Lec 9 Eigen-Analysis of Fibonacci sequence

The Fibonacci sequence is famous in number theory. It is said that the sequence commonly appears in physical systems. Fibonacci numbers are related to the “golden ratio” \((1 + \sqrt{5})/2\).

But from a mathematical point of view, the Fibonacci sequence does not seem special. It is generated by a linear recursion relationship, where the next number is the sum of the previous two (Eq. 1.18, p. 53)

\[
x_{n+1} = x_n + x_{n-1}.
\]

(2.13)

The term linear means that the principle of superposition holds [P1 (linear/nonlinear) of Section 1.4.17, p. 103) and 3.5.1, p. 224]. To understand the meaning of this we need to explore the z-transform, the discrete-time version of the Laplace transform. We will return to this in Chapter 4 (p. 231).

A related linear recurrence relation is to define the next output \(x_{n+1}\) be the average of the previous two

\[
x_{n+1} = \frac{x_n + x_{n-1}}{2}.
\]

In some ways this relationship, which has no famous name, is more useful than the famous Fibonacci recursion, since it perfectly removes oscillations of the form \((-1)^n\) (it is a 2-sample moving average, a trivial form of low-pass filter). Unlike the Fibonacci sequence, which diverges, it seems that biology prefers divergent sequences.

Equation 2.13 may be written as a 2x2 matrix relationship. If we define \(y_{n+1} = x_n\) then Eq. 2.13 is equivalent to (Eq. 1.19, p. 53)

\[
\begin{bmatrix}
x_{n+1} \\
y_{n+1}
\end{bmatrix}
= \begin{bmatrix}
1 & 1 \\
1 & 0
\end{bmatrix}
\begin{bmatrix}
x_n \\
y_n
\end{bmatrix}.
\]

(2.14)

The first equation is \(x_{n+1} = x_n + y_n\) while the second is \(y_{n+1} = x_n\), which is the same as \(y_n = x_{n-1}\). Note that the Pell 2x2 recursion is similar in form to the Fibonacci recursion. This removes mystique from both equations.

The most general 2d order recurrence relationships is

\[
x_{n+1} = -bx_n - cx_{n-1},
\]

with constants \(b, c \in \mathbb{R}\).

In the matrix diagonalization of the Pell equation we found that the eigen-values were \(\lambda_\pm = 1 \pm \sqrt{5}\), and the two solutions turned out to be powers of the eigen-values. The solution to the Fibonacci recursion may similarly be expressed in terms of a matrix. These two cases may thus be reduced by the same 2x2 eigen-value solution method.

The eigen-values of the Fibonacci matrix are

\[
det \begin{bmatrix}
1 - \lambda & 1 \\
1 & -\lambda
\end{bmatrix} = \lambda^2 - \lambda - 1 = (\lambda - 1/2)^2 - (1/2)^2 - 1 = (\lambda - 1/2)^2 - 5/4 = 0
\]

thus \(\lambda_\pm = \frac{1 \pm \sqrt{5}}{2} = [1.618, -0.618]\).
\[ x_n = x_{n-1} + x_{n-2} \]

- This is a 2-sample moving average difference equation with an unstable pole
- \( x_n = [0, 1, 1, 2, 3, 5, 8, 13, 21, 34, \cdots] \), assuming \( x_0 = 0, x_1 = 1 \):
- Analytic solution (Stillwell, 2010, p. 194):
  \[ \sqrt{5} x_n \equiv \left( \frac{1 + \sqrt{5}}{2} \right)^n - \left( \frac{1 - \sqrt{5}}{2} \right)^n \rightarrow \left( \frac{1 + \sqrt{5}}{2} \right)^\infty \]
  - \( \lim_{n \to \infty} \frac{x_{n+1}}{x_n} = \frac{1 + \sqrt{5}}{2} \)
  - Example: \( \frac{34}{21} = 1.6190 \approx \frac{1 + \sqrt{5}}{2} = 1.6180 \) 0.10\% error
- Matlab/Octave’s \texttt{rat}(1 + \sqrt{5}) = 3 + 1/(4 + 1/(4 + 1/(4 + 1/(4 + 1/4)))) =: [3; 4, 4, 4, \cdots]

\[ \text{https://en.wikipedia.org/wiki/Fibonacci_number} \]

Figure 2.5: Properties of the Fibonacci numbers (Stillwell, 2010, p. 28).

Note that this 2x2 equation is similar to Pell’s equation, suggesting that an eigen-function expansion of Eq. 1.19 may be used to analyze the sequence, as shown in Section 1.3.8 (p. 53) (Stillwell, 2010, 192). It is also related to a $10,000 prize, that was eventually solved.\(^8\)

2.3.2 Lec 10 Exam I

Chapter 3

Algebraic Equations: Stream 2

3.1 Week 4

3.1.1 Lec 11 Algebra and geometry as physics

Before Newton could work out his basic theories, algebra needed to be merged with Euclidean geometry. The key to putting geometry and algebra together is the Pythagorean theorem (Eq. 1.1), which is both geometry and algebra. To make the identification with geometry the sides of the triangle needed to be viewed as a length. This is done by recognizing that the area of a square is the square of a length. Thus a geometric proof requires one to show that the area of the square $A = a^2$ plus the area of square $B = b^2$ must equal the area of square $C = c^2$. There are many such constructions that show $A + B = C$ for the right triangle. It follows that in terms of coordinates of each vertex, the length of $c$ is given by

$$c = \sqrt{(x_2 - x_1)^2 + (y_2 - y_1)^2},$$

(3.1)

with $a = x_2 - x_1$ and $b = y_2 - y_1$ (Fig. 1.9, p. 49). Thus Eq. 1.1 is both an algebraic and a geometrical statement. This is not immediately obvious.

Analytic geometry is based on coordinates of points, where the length given by Eq. 3.1. Euclidean Geometry defines lengths without specifying the coordinates (Eq. 1.1). Algebra gave a totally new view to the quantification of lengths by introducing a coordinate system. This resulted in an entire new way to work with conic sections, which were now explained in terms of equations in coordinate systems. When viewed through the lens of algebra, Eq. 1.1 is a circle having radius $c$. Complex numbers provide an equivalent representations, since if $z = x + yj$, the unit circle is $z = e^{j\theta}$ and $|z|^2 = x^2 + y^2$. Here we explore the relationships between points, represented as coordinates, describing geometrical objects. We shall do this with simple examples from analytic geometry.

For example, in terms of the geometry, the intersection of two circles can occur at two points, and the intersection of two spheres gives a circle. These ideas may be verified using algebra, but in a very different, since the line can traverse through the circle, like a piece of thread going through the eye of a needle. In such cases the intersections are complex intersections.

For each of these problems, the lines and circles may intersect, or not, depending on how they are drawn. Yet we now know that even when they do not intersect on the sheet of paper, they still have an intersection, but the solution is complex. Finding such solutions require the use of algebra rather than geometry. These ideas were in the process of being understood, first by Fermat and Descartes, then by Newton, followed by the Bernoulli family and Euler.
Complex analytic functions: A very delicate point, that seems to have been ignored for centuries, is that the roots of $P_n(x)$ are, in general, complex, namely $x_k \in \mathbb{C}$. It seems a mystery that complex numbers were not accepted once the quadratic equation was discovered, but they were not. Newton called complex roots imaginary, presumably in a pejorative sense. The algebra of complex numbers is first attributed to Bombelli in 1575, more than 100 years before Newton. One can only begin to imagine what Bombelli learned from Diophantus, following his discovery of Diophantus’ Arithmetica in the Vatican library (Stillwell, 2010, p. 51).

It is interesting that Newton was using power series with fractional degree. These topics will be explored in Section 3.1.1.

When the argument is complex, analytic functions takes on an entirely new character. For example Euler's identity (1748) with $z = x + yj \in \mathbb{C}$ results in
\[
e^z = e^x (\cos(y) + j\sin(y)).\]

It should be clear that the complex analytic functions results in a new type of algebra, with no further assumptions beyond allowing the argument to be complex.

Prior to 1851 most of the analysis assumed that the roots of polynomials were real ($x_k \in \mathbb{R}$), even though there was massive evidence that they were complex ($x_n \in \mathbb{C}$). This is clearly evident in Newton’s work (c1687): When he found a non-real root, he ignored it (Stillwell, 2010, pp. 115-7). Euler (c1748) first derived the Zeta function as a function of real arguments $\zeta(x)$ with $\zeta, x \in \mathbb{R}$. Cauchy (c1814) broke this staid thinking with his analysis of complex analytic functions, but it was Riemann thesis (c1851), when working with Gauss (1777-1855), which had a several landmark breakthroughs. In this work Riemann introduced the extended complex plane, which explained the point at infinity. He also introduced Riemann sheets and Branch cuts, which finally allowed mathematics to better describe the physical world (Section 1.5.2).

The development of complex analytic functions led to many new fundamental theorems. Complex analytic functions have poles and zeros, branch cuts, Riemann sheets and can be analytic at the point at infinity. Many of these properties were first worked out by Augustin-Louis Cauchy (1789-1857), who drew heavily on the much earlier work of Euler, expanding Euler’s ideas into the complex plane (Chapter 4).

Systems of equations

We don’t need to restrict ourselves to polynomials in one variable. For example, we can work with the circle
\[y^2 + x^2 = r^2,\]
which is quadratic in $x, y$. Solving for roots $y(x_r) = 0$ ($y^2(x_r) = r^2 - x^2 = 0$) gives $(r - x_r)(r + x_r) = 0$, which simply says that when the circle crosses the $y = 0$ line at $x_r = \pm r$.

This equation may also be factored as
\[(y - xj)(y + xj) = r^2,\]
as is easily demonstrated by multiplying out the two monomials. This does not mean that a circle has complex roots. A root is defined by either $y(x_r) = 0$ or $x(y_r) = 0$.

Writing the conic as a 2d degree polynomial gives
\[P_2(x) = ax^2 + bx + c,\]
with \( y^2(x) = -P_2(x) \). Setting this equal to zero and completing the square (Eq. 1.29, p. 59), gives the equation for the roots

\[
\left( x \pm \frac{b}{2a} \right)^2 - \left( \frac{b}{2a} \right)^2 + \frac{c}{a} = 0,
\]
or

\[
x_\pm = -\frac{b}{2a} \pm \sqrt{\left( \frac{b}{2a} \right)^2 - \frac{c}{a}}.
\]
The polynomial in factored form is

\[
y^2 = -\left( x - \frac{b}{2a} \right)^2 + \left( \frac{b}{2a} \right)^2 - \frac{c}{a},
\]
is a conic section, and becomes a circle with \( a = 1, b = 0 \) and \( c = -r^2 \).

### 3.1.2 Lec 11a The physics of complex analytic expressions: linear vs. non-linear

The question we address here is “When do multi-variable complex analytic expressions appear in physics?” The most common example comes from the solution of the wave equation (Eq. 1.130) in three dimensions. Such cases arise in wave-guide problems, semiconductors, plasma waves, or for acoustic wave propagation in crystals (Brillouin, 1960) and the earth’s mantel (e.g., seismic waves, earthquakes, etc.). The solutions to these problems are based on the eigenfunction for the vector wave equation (see Chapter 5),

\[
P(s, \mathbf{x}) = e^{st} e^{-\kappa \cdot \mathbf{x}},
\]
where vector \( \mathbf{x} = [x \hat{x} + y \hat{y} + z \hat{z}] \in \mathbb{R}^3 \) points in the direction of the wave, unit vectors \([\hat{x}, \hat{y}, \hat{z}]\) are \( \in \mathbb{R}^3 \) and \( s = \sigma + \omega \jmath \in \mathbb{C} \) [rad] is the Laplace frequency. The vector function \( \kappa(s) \in \mathbb{R}^3 \) is the complex vector wave number, which describes the propagation of a plane wave of angular frequency \( \omega \), in the \( \kappa \) direction.

Just as the frequency \( s = \sigma + \omega \jmath \) must be complex, it is important to allow \( \kappa(s) \)\(^1\) to be a complex function of \( s \). The imaginary part accounts for the delay and the real part accounts for losses, as the wave propagates. While it is common to assume there are no losses (real part of \( \kappa = 0 \)), but in reality this assumption cannot be correct. In many cases zero loss is an excellent approximation, that gives realistic answers. But it is important to start the analysis with a notation that accounts for the most general situation, so that when losses are present, the notation doesn’t need to change. With this in mind, we take the vector wave number to be complex

\[
\kappa = k_r + k_j,
\]
where vector expression for the lattice vector is the imaginary part of \( \kappa \)

\[
\Im \kappa = k = \frac{2\pi}{\lambda_x} \hat{x} + \frac{2\pi}{\lambda_y} \hat{y} + \frac{2\pi}{\lambda_z} \hat{z},
\]
\(^1\)This function has many names in the literature, such as the wave number, propagation constant. However, its neither a number nor constant. In the case of Brillouin zones, it is the electron wave dispersion function (Brillouin, 1953, Ch. 1).
CHAPTER 3. ALGEBRAIC EQUATIONS: STREAM 2

is the vector wave number for three dimensional lossless plane-wave solutions ($\kappa_r = 0$).

Equation 3.3 is linear in $x$. If one takes the derivative of Eq. 3.2 with respect to either time or space,

$$\frac{\partial}{\partial t} e^{st} e^{-\kappa \cdot x} = se^{st} e^{-\kappa \cdot x}, \quad \frac{\partial}{\partial x} e^{st} e^{-k_x x} = \frac{2\pi}{\lambda_x} e^{st} e^{-k_x x}, \quad \nabla e^{st} e^{-\kappa \cdot x} = \kappa e^{st} e^{-\kappa \cdot x}$$

we find the eigenvalue of that derivative.

The units of $\kappa$ are reciprocal length [$m^{-1}$] since $\kappa \cdot x$ has units of radians. When there are losses $\kappa_r(s) = \Re \kappa(s)$ must be a function of frequency, due to the physics behind these losses. In many important cases, such as loss-less wave propagation in semiconductors, $\kappa \cdot x$ is a function of direction and position (Brillouin, 1960).

When the eigenfunction Eq. 3.2 is applied to the wave equation, a quadratic (degree 2) algebraic expression results, known as the dispersion relation. The three dimensional dispersion relation

$$\left(\frac{s}{c}\right)^2 = \kappa \cdot \kappa = \left(\frac{2\pi}{\lambda_x}\right)^2 + \left(\frac{2\pi}{\lambda_y}\right)^2 + \left(\frac{2\pi}{\lambda_z}\right)^2 = k_x^2 + k_y^2 + k_z^2$$

is a complex analytic algebraic relationship in four variables, frequency $s$ and the three complex lattice wave numbers. This represents a three-dimensional generalization of the well know relation between wavelength and frequency, i.e., $f \lambda = c$. For lossless plane waves propagating in free space, $|\kappa(s)| = \pm |s/c|$, where the sign accounts for the direction of the plane wave.

This scalar relation ($f \lambda = c$) was first deduced by Galileo in the 16th century and was then explored further by Mersenne a few years later. This relationship would have been important to Newton when formulating the wave equation, which he needed to estimate the speed of sound. We shall return to this in Chapters 4 and 5.

3.2 Week 5

3.2.1 Lec 12 Polynomial Factoring by deconvolution: degree > 3

Root classification for polynomials of Degree * = 1–4 (p.102); Quintic (* = 5) cannot be solved: Why?
Fundamental Thm of Algebra (d’Alembert, ≈1760)

**Inversion of polynomial convolution:** As we discussed in Chapter 1, given the roots, the construction of higher degree polynomials, is greatly assisted by the convolution method. This has physical meaning, and gives insight into the problem of factoring higher order polynomials. By this method we can obtain explicit relations for the coefficients of any polynomial in terms of its roots, as shown below for the cubic case.

Extending the example of Section 1.4.6, let’s find the relations between the coefficients and the roots for the cubic. To simplify the notation, assume that the polynomial has been normalized so that the lead $x^3$ term has coefficient 1. Then the cubic in terms of its roots $[a, b, c]$ is a convolution of three terms

$$[1, a] \ast [1, b] \ast [1, c] = [1, a + b, ab] \ast [1, c] = [1, a + b + c, ab + c(a + b), abc].$$
Thus in terms of its roots, the polynomial is
\[ x^3 + (a + b + c)x^2 + (ab + ca + cb)x + abc = 0. \]

As a second example the coefficients for a quartic gives

\[
[1, a + b + c, ab + c(a + b), abc] \star [1, d] =
[1, a + b + c + d, d(a + b + c) + c(a + b) + ab, d(ab + ac + bc) + abc, abcd].
\]

Thus
\[
\begin{align*}
x^4 & + (a + b + c + d)x^3 \\
+ [d(a + b + c) + c(a + b) + ab]x^2 \\
+ [d(ab + ac + bc) + abc]x \\
+ abcd x^0 &= 0.
\end{align*}
\]

It is clear what is going on here. The coefficient on \(x^4\) is 1 (by construction). The coefficient for \(x^3\) is the sum over the roots. The \(x^2\) term is the sum over all possible products of pairs of roots. The \(x\) term is the sum over all triple products of the four roots. The \(x^0 = 1\) term is the product of the four roots.

In fact this is a well known, a frequently quoted result from the mathematical literature, and trivial to show given an understand of convolution. If one wants the coefficients for the quintic, it is not even necessary to use convolution, as the pattern (rule) for all the coefficients is now clear.

You can experiment with this numerically using Matlab’s convolution routine \texttt{conv} (a, b). Once we start studying Laplace and Fourier transforms, convolution becomes critically important because
\[
f(t) \star g(t) \leftrightarrow F(s)G(s).
\]

So you didn’t need to learn how to take a Laplace transform, and then learn convolution. We have learned convolution first, independent of the Fourier and Laplace transforms.

For the case of the quadratic \((N = 2)\) we have the relations between the coefficients and the roots, found by completing the square. This required isolating \(x\) to a single term, and solving for it. We then proceeded to find the coefficients for the cubic \(N = 3\) and quartic \(N = 4\) case, after a few lines of calculation. For the quartic
\[
\begin{align*}
a_4 &= 1 \\
a_3 &= a + b + c + d \\
a_2 &= d(a + b + c) + c(a + b) + ba \\
a_1 &= d(ab + ac + bd) + abc \\
a_0 &= abcd
\end{align*}
\]

These relationships are algebraically nonlinear in the roots.

Based on the work of Galois, the system quartic \((N = 5)\), the equations are impossible to invert. Namely, given \(a_k\), one may not determine the five roots \([a, b, c, d, 3]\) analytically. However the roots are readily available by numerical methods, for example Newton’s method is a common method. based on gradient decent in the complex plane, for finding roots. Numerically this is considered to be a solved problem, subject only to numerical errors. Galois’ argument is based on the lack of
uniqueness of the roots. Specifically if the orders are permuted, there are more possible orders than there are roots. What seems to be missing are some constraints, based on symmetry conditions. For example, if the coefficients \(a_k \in \mathbb{R}\) are real, then the degrees of freedom are reduced. In such a case the roots come in conjugate pairs (i.e., \(s_k = s_k^*\)). If the coefficient vector are symmetric (i.e., \(a_0 = a_4, a_1 = a_3, \text{ etc.}\)), then the search space would be even further reduced. We might also place constraints on the locations of the roots, for example if a root \(s_k = \sigma_k + \omega j_k\) is in the left half plane (\(\sigma_k \leq 0\)), this would place a strong constraint on the roots. Such a constraint is physically very meaningful, as it indicates a stable response (or at least bounded output for bounded input (BIBO)). For a specific polynomial it is likely, given a physical situation, that the system is known to be positive real, as in the case of Brune impedance, which in practice have simple roots.

To gain some insight, let us look at the problem for \(N = 2\), which of course has a closed form solution:

\[
egin{align*}
a_2 & = 1 \\
a_1 & = a + b \\
a_0 & = ab
\end{align*}
\]

We must solve for \([a, b]\) given twice the mean, \(2(a + b)/2\), and the square of the geometric mean \((\sqrt{ab})^2\). Since we already know the answer (i.e., the quadratic formula). The solution was first worked out by the Babylonians (2000 BCE) Stillwell (2010, p. 92). It is important to recognize that for physical systems, the coefficients \(a_k\) are real. This requires that the roots come in conjugate pairs (\(b = a^*\)), thus \(ab = |a|^2\) and \(a + b = 2\Re a\), which makes the problem somewhat more difficult, due to the greater symmetry.

Once you have solved this problem, feel free to attempt the cubic case. Again, the answer is known, after thousands of years of searching. The solution to the cubic was first worked out by Cardano in 1545. According to Stillwell “The solution of the cubic was the first clear advance in mathematics since the time of the Greeks.” The ability to solve this problem required algebra, and the solutions were complex numbers. The denial of complex numbers was, in my view, the main stumbling block in the progress of these solutions. For example, how can two parallel lines have a solution? Equally mystifying, how can a circle and a line, that do not intersect, have intersections? From the algebra we know that they do. This was a basic problem that needed to be overcome. This story is still alive,\(^2\) because the cubic solution is so difficult.\(^3\)

One can only begin to imagine how much more difficult the quartic is, solved by Cardano’s student Ferrair, and published by Cardano in 1545. The impossibility of the quintic was finally resolved in 1826 by Abel (Stillwell, 2010, p. 102).

Finally with these challenges behind them, Analytic Geometry, relating algebra and geometry, via coordinate systems, was born.

### 3.2.2 Lec 13 Transmission line impedance

### 3.2.3 Lec 14: Development of Hilbert Space geometry

**Composition, Intersection and Gaussian elimination:**

When the impedance load, as a function of wave penetration depth \(x\) is uniform, then as the wave \(P^+\) travels, then there are no reflections. In this case \(\Gamma(s, x = 0) = 0\), and the impedance is equal

\(^2\)M. Kac, *How I became a mathematician.*” American Scientist (72), 498–499.

\(^3\)https://en.wikiquote.org/wiki/Mark_Kac
to \( r_o \). Given variations in the properties of the medium, say after a delay of \( \tau = x/c \), backward propagated waves are returned to the input. This results in a change in the load impedance, both in real and imaginary parts. As long as \(|\Gamma(s, x)| \leq 1\), the real part of \( Z(s, x = 0 \geq 0\). This is further discussed in Sect. ?? (p. ??).

As a counter example consider the well known problem in geometry: the intersection of a plane with a cone, which leads to the conic sections: the circle, hyperbola, ellipse and parabola, along with some degenerate cases, such as the intersection of two straight lines\(^4\). If we stick to such 3-dimensional objects, we can write equations in the three variables \([x, y, z]\), and be sure that they each represent some physical geometry. For example \(x^2 + y^2 + z^2 = r_o^2\) is a sphere of radius \(r_o\).

The geometry and the algebra do not always seem to agree. Which is correct? In general the geometry only looks at the real part of the solution, unless you know how to tease out the complex solutions. However the roots of every polynomial can be in \(\mathbb{C}\), so we may not ignore the imaginary roots, as Newton did.

### Composition of polynomials

When we multiply two polynomials, the resulting degree is the sum of the degrees of the individual polynomials (Section 1.4.7, p. 79). However when we compose two polynomials, of degree \(m\) and \(n\), the resulting degree is the \textit{product} of the degrees. For example given two polynomials of degree \(n\) and \(m\), \(f_n(x)\) and \(g_m(x)\), their composition is defined as \(f_n \circ g_m = f(g(x))\). For example, the composition of a line with a line, is a line. The composition of a line with a circle, is a circle, and the composition of a circle with a circle is a quartic. Composition of functions is also known as the \textit{construct of equations} (Stillwell, 2010, p. 118).

### Exercises:

1. **Two Circles**: The composition of 2 independent circles give 4 points:

\[
x^2 + y^2 = 1, \quad (x - 1)^2 + y^2 = 1
\]

Two circles intersect at 2 points: How does this work on using algebra?

Here we subtract the two equations, eliminating \(y\)

\[
x^2 - (x - 1)^2 = x^2 - (x^2 - 2x + 1) = 2x - 1 = 0
\]

Thus \(x = 1/2\). Verifying gives

\[
(1/2)^2 + y^2 = 1, \quad (1/2 - 1)^2 + y^2 = 1
\]

or

\[
y^2 = 1 - 1/4 = 3/4, \quad y^2 = 1 - 1/2 = 1/2
\]

or

\[
y = \pm \sqrt{3/4}, \quad y = \pm \sqrt{1/2}.
\]

In summary, there is only one value of \(x = 1/2\) but there are two values of \(y\), which is independent of the sign, since \(y\) appears in the equations as \(y^2\). Thus there are \textit{four} solutions: \((1/2, \pm \sqrt{3/4})\) and \((1/2, \pm \sqrt{1/2})\).

\(^4\)Such problems were first studied algebraically and Descartes (Stillwell, 2010, p. 118) and Fermat (c1637).
2. **Composition of a line and a circle:** clearly state the problem

Let the circle be \( x^2 + y^2 = 1 \) and the line be \( x + y = 1 \). Graphically it is easy to show that the solutions are \((0, 1)\) and \((1, 0)\). This problem has already been addressed with the search for Pythagorean triplets. In Fig. 2.3 we derive the real integer solutions for the line from \(-1, 0\) to \(x, y\) intersection the circle. The values of \(a, b, c\) are presented as Proof III in that figure.

But what can we say about the angles \(\Theta\)? The formula for the angles is

\[ e^{1_j\Theta} = \frac{1 + t_j}{1 - t_j} \]

with \(t = p/q \in \mathbb{Q}\).

If we solve this equation for \(t_j\) we find

\[ (1 - t_j)e^{1_j\Theta} = 1 + t_j \]
\[ (e^{1_j\Theta} - 1) = t_j(e^{1_j\Theta} + 1) \]
\[ t_j = \frac{e^{1_j\Theta} - 1}{e^{1_j\Theta} + 1} = \frac{e^{1_j\Theta/2} - e^{-1_j\Theta/2}}{e^{1_j\Theta/2} + e^{-1_j\Theta/2}} \]

3. **Problem 3:** Suppose we have two circles

\[ y^2 + x^2 = 1 \quad \text{and} \quad y^2 + x^2 = 1 \quad (3.6) \]

There are an infinite number of solutions since the circles lie on top of each other. All values of \(x, y\) are a solution. If

\[ y^2 + x^2 = 1 \quad \text{and} \quad y^2 + x^2 = 2 \quad (3.7) \]

there are no solutions, as the circles never touch. When

\[ y^2 + x^2 = 1 \quad \text{and} \quad y^2 + (x - 1)^2 = 1 \quad (3.8) \]

there are two intersection points. How can we find them?

We may remove \(y\) from these equations by subtracting them, which results in

\[ (x - 1)^2 - x^2 = 0 \]

which is easily solved since \(x^2 - 2x + 1 - x^2 = 0, x = 1/2\). We can then put this back into the first equation to find \(y^2 + 1/4 = 1\) or \(y = \pm \sqrt{3}/2\). This is easily verified.

What if the two circles are

\[ y^2 + x^2 = 1 \quad \text{and} \quad y^2 + (x - 1)^2 = 1 \quad (3.9) \]

Removing \(y\) as before gives \((x - 1)^2 - x^2 = 0\) or \(x^2 - 2x + 1 - x^2 = 0\). Thus \(x = 1j/2\).

Putting this back in the first (or second) equation gives \(y^2 - 1/2 = 1\), or \(y = \pm \sqrt{5}/2\). Thus the final solution is \((x_0, y_0) = 1j/2, \pm 5/2\).

What can we take away from this exercise? The geometry fails us, at least as far as we know how to view it, in real coordinates. However using algebra we can determine the complex solutions. It is no wonder that Newton, and many others, were fooled by such calculations, and called such results imaginary. But as we shall learn, only by understanding such equations, can we fully appreciate the physics that is built upon them.
4. **Problem 4:** This problem is similar to problem 3 of two circles, but is in 3 dimensions with the intersection of two spheres. A unit radius sphere is given by

\[ x^2 + y^2 + z^2 = 1 \]

If we take \([x, y, z] \in \mathbb{R}\) we can easily picture the intersection of two spheres. If they just touch, the intersection is a single point. If they overlap, the solution is a circle, and if one is inside the other, there are no (real) solutions.

Next is an example of a polynomial in three variables, with no linear terms. We can generalize it by a shift in space

\[(x - 1)^2 + (y + 2)^2 + (z - 3)^2 = 2^2.\]

Here we shift \(x\) by 1, \(y\) by -2 and \(z\) by 3, and scaled the radius from 1 to 2. Clearly this equation has the same geometry (its still a sphere), in factored form. Expanding it out gives

\[x^2 - 2x + y^2 + 2y + z^2 - 3z + 10 = 0.\]

Clearly the factored format is much easier to interpret. It should be clear how to find the factored form given the general quadratic polynomial in the three variables (use Eq. 1.29 three times). Another interesting question is “Why are there no \(ZY, ZY, xyz\) terms? What if \([x, y, z] \in \mathbb{C}\)? Does this make physical sense? The answer is yes, if the variables are not the spatial coordinates but something called the *complex wave number*. More on this later.

### 3.3 Week 6

#### 3.3.1 Lec 15 Gaussian Elimination of linear equations

**Example problems using Gaussian Elimination:** Gaussian Elimination (Appendix ??, p. ??) is valid for nonlinear systems of equations. Till now we have emphasized the reduction of linear systems of equations.

**Problem 1:** Two lines *in a plane* either intersect or are parallel, in which case they are said to meet at \(\infty\). Does this make sense? The two equations that describe this may be written in matrix form as \(Ax = b\), which written out as

\[
\begin{bmatrix}
  a_{11} & a_{12} \\
  a_{21} & a_{22}
\end{bmatrix}
\begin{bmatrix}
x_1 \\
x_2
\end{bmatrix}
= 
\begin{bmatrix}
b_1 \\
b_2
\end{bmatrix}
\]

(3.10)

The intersection point \(x_0, y_0\) is given by the solution of two equations:

\[
\begin{bmatrix}
x_1 \\
x_2
\end{bmatrix}
= 
\frac{1}{\Delta}
\begin{bmatrix}
a_{22} & -a_{12} \\
-a_{21} & a_{11}
\end{bmatrix}
\begin{bmatrix}
b_1 \\
b_2
\end{bmatrix},
\]

(3.11)

where \(\Delta = a_{11}a_{22} - a_{12}a_{21}\) is the determinant of matrix \(A\) (Matlab’s \texttt{det(A)} function).
It is useful to give an interpretation of these two equations. Each row of the 2x2 matrix defines a line in the \((x,y)\) plane. The top row is

\[ a_{11}x + a_{12}y = b_1. \]

Normally we would write this equation as \(y(x) = \alpha x + \beta\), where \(\alpha\) is the slope and \(\beta\) is the intercept (i.e., \(y(0) = \beta\)). In terms of the elements of matrix \(A\), the slope of the first equation is \(\alpha = -a_{11}/a_{12}\) while the slope of the second is \(\alpha = -a_{21}/a_{22}\). The two slopes are equal (the lines are parallel) when \(-a_{11}/a_{12} = -a_{21}/a_{22}\), or written out

\[ \Delta = a_{11}a_{22} - a_{12}a_{21} = 0. \]

Thus when the determinate is zero, the two lines are parallel and there is no solution to the equations.

This 2x2 matrix equation is equivalent to a 2\(^{d}\) degree polynomial. If we seek an eigenvector solution \([e_1, e_2]^T\) such that

\[
\begin{bmatrix}
  a_{11} & a_{12} \\
  a_{21} & a_{22}
\end{bmatrix}
\begin{bmatrix}
  e_1 \\
  e_2
\end{bmatrix} = \lambda
\begin{bmatrix}
  e_1 \\
  e_2
\end{bmatrix},
\]

the 2x2 equation becomes singular, and \(\lambda\) is one of the roots of the polynomial. One may proceed by merging the two terms to give

\[
\begin{bmatrix}
  a_{11} - \lambda & a_{12} \\
  a_{21} & a_{22} - \lambda
\end{bmatrix}
\begin{bmatrix}
  e_1 \\
  e_2
\end{bmatrix} = \begin{bmatrix}
  0 \\
  0
\end{bmatrix}.
\]

Clearly this new matrix has no solution, since if it did, \([e_1, e_2]^T\) would be zero, which is nonsense. If it has no solution, then the determinant of the matrix must be zero. Forming this determinate gives

\[ (a_{11} - \lambda)(a_{22} - \lambda) - a_{12}a_{21} = 0 \]

thus we obtain the following quadratic equation for the roots \(\lambda_{\pm}\) (eigenvalues)

\[ \lambda_{\pm}^2 - (a_{11} + a_{22})\lambda_{\pm} + \Delta = 0. \]

When \(\Delta = 0\), one eigenvalue is zero while the other is \(a_{11} + a_{22}\), which is known as the trace of the matrix.

**In summary:** Given a “linear” equation for the point of intersection of two lines, we see that there must be two points of intersection, as there are always two roots of the quadratic characteristic polynomial. However the two lines only intersect at one point. What is the meaning of this second root?

Some simple examples will help. The eigenvalues depend on the relative slopes of the lines, which in general can become complex. The intercepts are dependent on \(b\). Thus when the RHS is zero, the eigenvalues are irrelevant. This covers the very simple examples. When one eigenvalue is real and the other is imaginary, more interesting things are happening since the slope of one line is real and the slope of the other is pure imaginary. The lines can intersect in the real plane, and again in the complex plane.

**Needs work.**

Let us try an example of two lines, slopes of 1 and 2: \(y_1 = x + a\) and \(y_2 = 2x + b\). In matrix form Let

\[
\begin{bmatrix}
  1 & -1 \\
  1 & -2
\end{bmatrix}
\begin{bmatrix}
  y \\
  x
\end{bmatrix} = \begin{bmatrix}
  a \\
  b
\end{bmatrix}
\]

(3.14)
The determinate is \( \Delta = -1 \), thus the solution is
\[
\begin{bmatrix}
y_0 \\
x_0
\end{bmatrix}
= -1 \begin{bmatrix} -2 & 1 \\ -1 & 1 \end{bmatrix} \begin{bmatrix} a \\
b \end{bmatrix}
= \begin{bmatrix} 2 & -1 \\ 1 & -1 \end{bmatrix} \begin{bmatrix} a \\
b \end{bmatrix}
= \begin{bmatrix} 2a - b \\
a - b \end{bmatrix}.
\] (3.15)
Thus the two real lines having slopes of 1 and 2 having intercepts of \( a \) and \( b \), meet \((x_0, y_0) = (2a - b, a - b)\). We may verify by substituting \( x = a - b \) into the starting equations \( y_1 = (a - b) + a = 2a - b \) and \( y_2 = 2(a - b) + b = 2a - b \), which each \( 2a - b \).

While there is a unique solution, there are two eigenvalues, given by the roots of
\[
(1 - \lambda_\pm)(-2 - \lambda_\pm) + 1 = 0.
\]
If we transfer the sign from one monomial to the other
\[
(-1 + \lambda_\pm)(2 + \lambda_\pm) + 1 = 0
\]
and reorder for simplicity
\[
(\lambda_\pm - 1)(\lambda_\pm + 2) + 1 = 0
\]
we obtain the quadratic for the roots
\[
\lambda_\pm^2 + \lambda_\pm - 1 = 0.
\]
Completing the square gives
\[
(\lambda_\pm + 1/2)^2 = 3/4.
\]
or
\[
\lambda_\pm = -1/2 \pm \sqrt{3}/2.
\]
The question is, what is the relationship between the eigenvalues and the final solution, if any? Maybe none. The solution \((x_0, y_0)\) is reasonable, and its not clear that the eigenvalues play any useful role here, other than to predict there is a second solution. I’m confused.

**Two lines in 3-space:** In three dimensions
\[
\begin{bmatrix}
a_{11} & a_{12} & a_{13} \\
a_{21} & a_{22} & a_{23} \\
a_{31} & a_{32} & a_{33}
\end{bmatrix}
\begin{bmatrix}
x \\
y \\
z
\end{bmatrix}
= \begin{bmatrix} b_1 \\
b_2 \\
b_3
\end{bmatrix}
\] (3.16)
Each row of the matrix describes a plane, which is said to be linear in the unknowns \((x, y, z)\). Thus the system of linear equations represents three planes, which must intersect at one point. If two planes are parallel, there is no real solution. In this case the intersection by the third plane generates two parallel lines.

As in the 2x2 case, one may convert this linear equation into a cubic polynomial by setting the determinant of the matrix, with \(-\lambda\) subtracted from the diagonal, equal to zero. That is, \(det(A - \lambda I) = 0\). Here \( I \) is the matrix with 1 on the diagonal and zero off the diagonal.
**Simple example:** As a simple example, let the first plane be \( z = 0 \) (independent of \( x, y \)), the second parallel plane be \( z = 1 \) (independent of \( (x, y) \)) and the third plane be \( x = 0 \) (independent of \( y, z \)). This results in the system of equations

\[
\begin{bmatrix}
0 & 0 & a_{13} \\
0 & 0 & a_{23} \\
a_{31} & 0 & 0
\end{bmatrix}
\begin{bmatrix}
x \\ y \\ z
\end{bmatrix} =
\begin{bmatrix}
0 \\ 1 \\ 0
\end{bmatrix}
\]  
(3.17)

Writing out the three equations we find \( a_{13}z = 0 \), \( a_{23}z = 1 \), and \( a_{31}x = 0 \). Note that \( \det(A) = 0 \) (we need to learn how to compute the 3x3 determinant). This means the three planes never intersect at one point. Use Matlab to find the eigenvalues.

### 3.3.2 Lec 16 Matrix composition: Bilinear and ABCD transformations

**The Transmission matrix**

A transmission matrix is a 2x2 matrix that characterizes a 2-port circuit, one having an input and output voltage and current, as shown in Fig. 1.18. The input is the voltage and current \( V_1, I_1 \) and the output is the voltage and current \( V_2, -I_2 \), with the current always defined to flow into the port. For any such a linear network, the input-output relations may be written in a totally general way as

\[
\begin{bmatrix}
V_1 \\
I_1
\end{bmatrix} =
\begin{bmatrix}
A(s) & B(s) \\
C(s) & D(s)
\end{bmatrix}
\begin{bmatrix}
V_2 \\
-I_2
\end{bmatrix}.
\]

In Section 1.4.11 we showed that a cascade of such matrices is composition. We shall show below that the justification of this relationship is based on the composition of bilinear transformations.

Expanding Eq. 1.70 into its individual equations demonstrates the linear form of the relations

\[
V_1 = A(s)V_2 - B(s)I_2 \quad I_1 = C(s)V_2 - D(s)I_2,
\]

quantifying the relationship between the input voltage and current to its output voltage and current.

Define \( H(s) = V_2/V_1 \) as the **transfer function**, as the ratio of the output voltage \( V_2 \) over the input voltage \( V_1 \), under the constraint that the output current \( I_2 = 0 \). From this definition \( H(s) = 1/A(s) \).

In a similar fashion we may define the meaning of all four functions as

\[
A(s) \equiv \frac{V_1}{V_2} \bigg|_{I_2=0} \quad B(s) \equiv -\frac{V_1}{I_2} \bigg|_{V_2=0} \quad C(s) \equiv \frac{I_1}{V_2} \bigg|_{I_2=0} \quad D(s) \equiv -\frac{I_1}{I_2} \bigg|_{V_2=0}
\]  
(3.18)

From Eq. 1.70 one may compute any desired quantity, specifically those quantities defined in Eq. 3.19, the open circuit voltage transfer function \( 1/A(s) \), the short-circuit transfer current \( 1/D(s) \) and the two transfer impedances \( B(s) \) and \( 1/C(s) \).

In the engineering fields this matrix composition is called the **Transmission matrix**, also known as the ABCD method. It is a powerful method that is easy to learn and use, that gives important insights into transmission lines, and thus even the 1 dimensional wave equation. This method is exquisitely presented in the physics literature, with solid state applications, in an amazing book by Brillouin (1953).
Derivation of ABCD matrix for example of Fig. 1.18 (p. 92).

The derivation is straightforward by the application of Ohm’s law, as shown in Section 1.4.11.

The convenience of the ABCD matrix method is that the output of one is identically the input of the next. Cascading (composing) the results for the series inductor with the shunt compliance leads to the 2x2 matrix form that precisely corresponds to the transmission line CFA shown in Fig. 2.2,

\[
\begin{bmatrix}
V_n(s) \\
I_n(s)
\end{bmatrix} = \begin{bmatrix} 1 & sL_n \\ 0 & 1 \end{bmatrix} \begin{bmatrix} V_{n+1}(s) \\
-I_{n+1}(s)
\end{bmatrix}.
\] (3.20)

This matrix relation characterizes the series mass term \(sL_n\). A second equation maybe be used for the shunt capacitance term \(sY_n\)

\[
\begin{bmatrix}
V_n(s) \\
I_n(s)
\end{bmatrix} = \begin{bmatrix} 1 & 0 \\ sC_n & 1 \end{bmatrix} \begin{bmatrix} V_{n+1}(s) \\
-I_{n+1}(s)
\end{bmatrix}.
\] (3.21)

The positive constants \(L_n, C_n \in \mathbb{R}\) represent the series mass (inductance) and the shunt compliance (capacitance) of the mechanical (electrical) network. The integer \(n\) indexes the series and shunt sections, that are composed one following the next.

A more detailed analysis shows that for the case of no losses, the wave velocity is

\[
c = 1/\sqrt{LC}.
\] (3.22)

Also the wave number is

\[
\kappa = \sqrt{ZY} = \sqrt{sL \cdot sC} = s\sqrt{LC} = \frac{s}{c}.
\] (3.23)

Finally the characteristic resistance is

\[
r_0 = \sqrt{Z/Y} = \sqrt{sL/sC} = \sqrt{L/C}.
\] (3.24)

All of these based on a unit length \(\Delta_x\) as shown in Fig. 2.2 (p. 189).

Matrix composition and the bilinear transform: Now that we have defined the composition of two functions, we will use it to define the Möbius or bilinear transformation. Once you understand how this works, hopefully you will understand why it is the unifying element in many important engineering problems.

The bilinear transformation is given by

\[
w = \frac{a + bz}{c + dz}
\]

This takes one complex number \(z = x + iy\) and transforms it into another complex number \(w = u + iv\). This transformation is bilinear in the sense that its linear in both the input and output side of the equation. This may be seen when written as

\[(c + dz)w = a + bz,
\]

since this relation is linear in the coefficients \([a, b, c, d]\). An important example is the transformation between impedance \(Z(s)\) and reflectance \(\Gamma(s)\),

\[
\Gamma(s) = \frac{Z(s) - r_0}{Z(s) + r_0},
\]
which is widely used in transmission line problems. In this example \( w = \Gamma, z = Z(s), a = -r_0, b = 1, c = r_0, d = 1 \).

If we define a second bilinear transformation (this could be the transformation from reflectance back to impedance)

\[
    r = \frac{\alpha + \beta w}{\gamma + \delta w},
\]

and then compose the two something astray wrt arguments

\[
    w \circ r = \frac{a + b r}{c + d r} = \frac{a(\gamma + \delta w) + b(\alpha + \beta w)}{c(\gamma + \delta w) + d(\alpha + \beta w)} = \frac{a\gamma + b\alpha + (a\delta + b\beta)w}{c\gamma + d\alpha + (c\delta + d\beta)w},
\]

something surprising happens. The composition \( w \circ r \) may be written in matrix form, as the product of two matrices that represents each bilinear transform. This may be seen as true by inspecting the coefficients of the composition \( w \circ r \) (shown above) and the product of the two matrices

\[
    \begin{bmatrix}
        a & b \\
        c & d
    \end{bmatrix}
    \begin{bmatrix}
        \alpha & \beta \\
        \gamma & \delta
    \end{bmatrix}
    =
    \begin{bmatrix}
        (a\gamma + b\alpha) & (a\delta + b\beta) \\
        (c\gamma + d\alpha) & (c\delta + d\beta)
    \end{bmatrix}.
\]

The the power of this composition property of the bilinear transform may be put to work solving important engineering problems, using transmission matrices.

### 3.3.3 Lec 16a Transmission line example

In this problem, we will look at the transfer function of a two-port network, shown in Fig. 3.1. We wish to model the dynamics of a freight-train having \( N \) such cars. The model of the train consists of masses connected by springs.

**Transfer functions:** The velocity transfer function for this system is defined as the ratio of the output to the input velocity. Consider the engine on the left pulling the train at velocity \( V_1 \) and each car responding with a velocity of \( V_n \). Then

\[
    H(s) = \frac{V_N(s)}{V_1(s)}
\]

is the frequency domain ratio of the last car having velocity \( V_N \) to \( V_1 \), the velocity of the engine, at the left most spring (i.e., coupler).

![Figure 3.1: Depiction of a train consisting of cars, treated as a mass \( M \) and linkages, treated as springs of stiffness \( K \) or compliance \( C = 1/K \). Below it is the electrical equivalent circuit, for comparison. The mass is modeled as an inductor and the springs as capacitors to ground. The velocity is analogous to a current and the force \( f_n(t) \) to the voltage \( v_n(t) \).](image-url)
The transmission matrix  Consistent with the figure, break the model into cells each consisting of three elements: a series inductor representing half the mass \( L = M/2 \), a shunt capacitor representing the spring \( C = 1/K \), and another series inductor representing half the mass \( L = M/2 \). Each cell is reversible, making the model a cascade of identical cells.

Exercise:  Use the ABCD method to find the matrix representation of a train, shown in Fig. 3.1. At each node define the force \( f_n(t) \leftrightarrow F_n(\omega) \) and the velocity \( v_n(t) \leftrightarrow V_n(\omega) \) at junction \( n \).

1. Write the ABCD matrix \( T \) for a single cell, composed of series mass \( M/2 \), shunt compliance \( C \) and series mass \( M/2 \), that relates the input node 1 to node 2 where

\[
\begin{bmatrix} F_1 \\ V_1 \end{bmatrix} = T \begin{bmatrix} F_2(\omega) \\ -V_2(\omega) \end{bmatrix}
\]

Note that here the mechanical force \( F \) is analogous to electrical voltage, and the mechanical velocity \( V \) is analogous to electrical current.

Sol:  

\[
T = \begin{bmatrix} 1 & sM/2 \\ 0 & 1 \end{bmatrix} \begin{bmatrix} 1 & 0 \\ sC & 1 \end{bmatrix} \begin{bmatrix} 1 & sM/2 \\ 0 & 1 \end{bmatrix} = \begin{bmatrix} 1 + s^2MC/2 & (sM/2)(2 + s^2MC/2) \\ sC & 1 + s^2MC/2 \end{bmatrix} \quad (3.25)
\]

\[
= \begin{bmatrix} 1 + \frac{1}{2}s^2/c_o^2 & sM(1 + \frac{1}{2}s^2/c_o^2) \\ sC & 1 + \frac{1}{2}s^2/c_o^2 \end{bmatrix} \quad (3.26)
\]

\[
\approx \begin{bmatrix} 1 & sM \\ sC & 1 \end{bmatrix}, \quad (3.27)
\]

the final expression being valid when ignoring terms in \( s^2/c_o^2 \) compared to 1. Here \( c_o = 1/\sqrt{MC} \) is the phase velocity of waves traveling along the line.

Since \( c_o = \lambda f, \lambda = f/c_0 \) is the wave length of waves traveling on the line. When \( \lambda > \Delta \), the cell size, the lumped line is a good approximation to a continuous transmission line. The condition \( \lambda > 2\Delta \) is called the spatial Nyquist sampling condition.

2. Assuming that \( N = 2 \) and that \( F_2 = 0 \) (two mass problem), find the transfer function \( H(s) \equiv V_2/V_1 \). From the results of the \( T \) matrix you determined above, find

\[
H_{21}(s) = \frac{V_2}{V_1} \bigg|_{F_2=0}
\]

Sol: From the lower equation we see that \( V_1 = sCF_2 - (s^2MC/2 + 1)V_2 \). Recall that \( F_2 = 0 \), thus

\[
\frac{V_2}{V_1} = \frac{-1}{s^2MC/2 + 1} = \left( \frac{c_+}{s-s_+} + \frac{c_-}{s-s_-} \right).
\]

with \( s_\pm = \pm j\sqrt{\frac{2}{MC}} \) and \( c_\pm = \pm j/\sqrt{2MC} \).
3. Find \( h_{21}(t) \), the inverse Laplace transform of \( H_{21}(s) \). Sol:

\[
h(t) = \oint_{\sigma-j\infty}^{\sigma+j\infty} \frac{e^{st}}{s^2MC/2 + 1} \frac{ds}{2\pi j} = c_+ e^{-s^2t}u(t) + c_- e^{-s^2t}u(t).
\]

The integral follows from the CRT. The poles are at \( s_{\pm} = \pm j\sqrt{\frac{2}{MC}} \) and the residues are \( c_{\pm} = \pm j/\sqrt{2MC} \).

4. What is the input impedance \( Z_{in} = F_1/V_1 \) if \( F_2 = -r_0V_2 \)? Sol:

Starting from \( T \) calculated above, find \( Z_{in} \)

\[
Z_{in}(s) = \frac{F_1}{V_2} = T \left[ \begin{array}{c} F_3 \\ -V_2 \end{array} \right] = \frac{-V_2r_o(1 + \frac{1}{2}s^2/c_0^2) + r_0sM(1 + \frac{1}{2}s^2/c_0^2) + V_2sCr_o}{V_2sCr_o + V_2(1 + \frac{1}{2}s^2/c_0^2)}
\]

Constant \( c_0 = \sqrt{L/C} \) is called the TLs phase velocity.

5. Simplify the expression for \( Z_{in} \), assuming that:

(a) \( s^2/c_0^2 \ll 1 \):

(b) the line is terminated in its characteristic resistance \( r_o = \sqrt{M/C} \), \( F_2 = -r_0V_2 \)

(c) \( N \to \infty \)

Sol: Using assumption (a)

\[
Z_{in}(s) \approx \frac{r_o(1 + \frac{1}{2}s^2/c_0^2) + r_0sM(1 + \frac{1}{2}s^2/c_0^2)}{sCr_0 + (1 + \frac{1}{2}s^2/c_0^2)} \approx \frac{r_o + sM}{1 + sCr_0} = \frac{1 + sM/r_o}{1 + sCr_0}
\]

Using assumption (b), followed (a second time) by (a), gives

\[
Z_{in} = \frac{r_o(1 + s\sqrt{C/M})}{s\sqrt{M/C} + 1} = \frac{r_o(1 + s\sqrt{C/M})}{1 + s\sqrt{C/M}} = \frac{r_o(1 + s/c_0)}{1 + s/c_0} \to r_o
\]

The above shows that below the Nyquist spatial cutoff frequency (approximation (a)), the system approximates a transmission line, in that when terminated in its own characteristic resistance \( r_o \), the input equal \( r_o \). It is expected that as \( N \to \infty \) the slope becomes steeper at the cutoff frequency, and this last approximation sharpens. Two ways of studying this effect are numerical simulations, and an eigen-function analysis (see below).

6. State the ABCD matrix relationship between the first and \( N \)th node, in terms of of the cell matrix. Sol:

\[
\begin{bmatrix} F_1 \\ V_1 \end{bmatrix} = T^N \begin{bmatrix} F_N(\omega) \\ -V_N(\omega) \end{bmatrix}
\]

7. Given a \( T \) (ABCD) transmission matrix, the eigen-values are and vectors are given in Appendix C of the Notes (p. 143), repeated here.
3.3. WEEK 6

**Eigen-values:**

\[
\begin{bmatrix}
\lambda_+ \\
\lambda_-
\end{bmatrix}
= \frac{1}{2} \left[ (A + D) - \sqrt{A^2 + D^2 - 4\Delta T} \right]
\]

Due to a reversible symmetry \((A = D)\) this simplifies to \(\lambda_\pm = A \mp \sqrt{BC}\) so that the eigen matrix is

\[
\Lambda = \begin{bmatrix}
A - \sqrt{BC} & 0 \\
0 & A + \sqrt{BC}
\end{bmatrix}
\]

In the lossless case described in Fig. 3.1, \(A = D = 0\) and \(BC = s^2 MC = s^2/c_o^2\), simplifying \(\Lambda\) even further

\[
\Lambda = \begin{bmatrix}
-s/c_o & 0 \\
0 & +s/c_o
\end{bmatrix}.
\]

This is then the complex propagation matrix since \(\kappa = s/c_o\) is the complex propagation function.

**Eigen-vectors:** The eigenvectors are

\[
\begin{bmatrix}
E_+ \\
E_-
\end{bmatrix} = \frac{1}{2C} \left[ (A - D) \mp \sqrt{A^2 + D^2 - 4\Delta T} \right].
\]

The determinant of \(E\) is

\[
\Delta_E = -\frac{1}{2C} \sqrt{A^2 + D^2 - 4\Delta T}.
\]

In the reversible case (when \(A = D\))

\[
E = \begin{bmatrix}
-\sqrt{\frac{B}{C}} & +\sqrt{\frac{B}{C}} \\
\frac{1}{1} & \frac{1}{1}
\end{bmatrix}, \quad E^{-1} = \frac{1}{2} \begin{bmatrix}
-\sqrt{\frac{C}{B}} & 1 \\
+\sqrt{\frac{C}{B}} & 1
\end{bmatrix}.
\]

and

\[
\Delta_E = -2 \sqrt{\frac{B}{C}} = -2r_o.
\]

Thus there are two important invariants for the transmission lines, the complex propagation function \(\kappa = \sqrt{BC} = s/c_o\), and the characteristic resistance \(r_o = \sqrt{B/C}\).

**Transfer functions of the transmission line:** The velocity transfer function \(H_{N,1} = \frac{V_N}{V_i}\) is given by Sol:

\[
\begin{bmatrix}
F_i \\
V_i
\end{bmatrix} = T_N \begin{bmatrix}
F_N(\omega) \\
-V_N(\omega)
\end{bmatrix}
\]

along with the eigen-value expansion

\[
T^N = EA^N E^{-1} = E \begin{bmatrix}
\lambda^N_+ & 0 \\
0 & \lambda^N_-
\end{bmatrix} E^{-1}.
\]
3.3.4 Lec 17 Introduction to the Branch cut and Riemann sheets

Branch cuts are required to preserve the single-valued nature of complex analytic functions. When an analytic function is multi-valued, some method needs to be devised to allow the multi-valued complex analytic function to be expanded as a Taylor series, which is necessarily single-valued. It follows that each single-valued sheet must have a different expansion, valid out to the nearest pole (or singularity). We shall explain these ideas with the simplest case, the double-valued square root function \( w(z) = \pm \sqrt{z} \), as shown in Fig. 1.27 (p. 118).

3.4 Week 7

3.4.1 Lec 18 Complex analytic mappings (domain coloring)

When one uses complex analytic functions it is helpful to understand their properties in the complex plane. In this section we explore several well-known functions using domain coloring, first discussed in Section 1.4.14, p. 98. For the following figures the coordinate systems are defined by \( s = \sigma + \omega j = \sqrt{x^2 + y^2} e^{\phi j} \) and \( w = u + v j = \sqrt{u^2 + v^2} e^{\psi j} \).

For the first example (Section 1.4.14), \( w(s) = s^2 \) (Fig. 1.28, p. 118, left) and its inverse \( s(w) = \sqrt{w} \) (right) are shown. On the left the red region, corresponding to \( 0^\circ \) [degrees], appears at both 0 and 180 \((u = \pm 1)\) in the \( w \) plane. This is because in polar coordinates \( s^2 = |s|^2 e^{2\theta j} \) where \( \theta \) is the angle of \( s = |s| e^{\theta j} \). The square causes the phase to rotate twice around for once around the \( s \) plane. Namely the angle is doubled, and the magnitude squared. Due to the faster changing phase in \( w \), there are two red regions, one when \( \theta = 0 \) and the second at \( \theta = \pi \) \((\angle w(s) = 2\theta)\). The black spot is dilated due to the squaring of the radius (expanding it).

On the right the \( \sqrt{w} = \sqrt{|w|} e^{\psi j/2} \). Because the angle of \( w \) is divided by two, it takes twice as much phase (in \( w \)) to cover the same angle. Thus the red region (\( 0^\circ \)) is expanded. We barely see the violet \( 90^\circ \) and yellow \(-90^\circ\) angles. There is a branch cut running from \( w = 0 \) to \( w = \infty \). As the branch cut is crossed, the function switches Riemann sheets, going from the top sheet (shown here) to the bottom sheet (not shown). Figure 1.27 in Section 3.3.4 depicts what is going on with these two sheets, and show the branch cut from the origin (point O) to \( \infty \). In this depiction the first sheet \((+\sqrt{z})\) is on the bottom, while the second sheet \((-\sqrt{z})\) is on top. For every value of \( z \) there are two possible outcomes, \( \pm \sqrt{z} \), represented by the two sheets.

Two more examples are given in Fig. 3.2 to interpret the two complex colorized mappings \( w = \cos(\pi s) \) (left) and the Bessel function \( J_0(\pi z) \). Note how the white and black contour lines are always perpendicular where they cross, just as in the calibration plots for the \( x \) and \( y \) axes, shown in Fig. 1.21 in Section 1.4.14 (p. 98).

Along the \( \sigma \) axis the \( \cos(\pi x) \) is the periodic with a period of \( \pi \). The dark spots are at the zeros. at \( \pm \pi/2, \pm 3\pi/2, \ldots \). When we stray off the \( \omega j = 0 \) axis, the function either goes to zero (black) or \( \infty \) (white). This behavior carries the same \( \pi \) periodicity as it has along the \( \omega = 0 \) line. On the right is the Bessel function \( J_0(\pi z) \), which is similar to \( \cos(\pi z) \), except the zeros are distorted away from the origin. These figure are worthy of careful study to develop an intuition for complex functions of complex variables. In Section 1.4.14 we shall explore more complex mappings, and in greater detail.

In the third example (Fig. 3.3) we show \( w = \tan(z) \) and its inverse \( z = \tan^{-1}(w) \). The tangent function has zeros where \( \sin(z) \) has zeros (e.g., at \( z = 0 \)) and poles where \( \cos(z) \) is zero (e.g., at \( \pm \pi/2 \)). The inverse function \( s = \text{atan}(w) \) has a zero at \( w = 0 \) and branch cuts eliminating from \( z = \pm \pi \).
Figure 3.2: On the left is the Bessel function \( J_0(\pi z) \), which is similar to \( \cos(\pi z) \), except the zeros are distorted away from \( s = 0 \) by a small amount, due to the cylindrical geometry. On the right is the related Hankel function \( H_0^{(1)}(u^2) \). The Bessel and Hankel functions are solution to the wave equation in cylindrical coordinates with different boundary conditions. The zeros in the function are the places where the pinned boundary condition is satisfied (where the string is restricted, by the boundary, from moving). The Hankel function \( H_0^{(1)}(\pi z/2) \) has a branch cut, and a zero \( z_{0,1} \) at \( \pi + z_{0,1}/2 = -1.5 - 0.1j \).

The command \texttt{zviz besselj(0,pi*Z) besseli(0,j*pi*Z)} gives exactly the same plot, demonstrating that \( I_0(z) = J_0(zj) \).

It is fun, easy and interesting to study polynomials, say of degree 5 and 4 (i.e., with one zero removed), to demonstrate the Fundamental Theorem of Algebra.

3.4.2 Lec 20: Laplace transform and the Riemann zeta function (I)

3.4.3 Filter classification

Let \( z \equiv e^{sT} \) where \( T \) is the “sample period” at which data is taken (every \( T \) seconds). For example if \( T = 22.7 = 1/44100 \) seconds then the data is sampled at 44100 kHz. This is how a CD player works with high quality music. Thus the unit-time delay operator \( z^{-1} \) as

\[
\delta(t - T) \leftrightarrow e^{-sT}
\]

Filter properties

Given the function

\[
F(s) = \frac{(s + 1)(s - 1)}{(s + 2)},
\]

1. find the minimum phase \( M(s) \) and all-pass \( A(s) \) parts. The minimum phase part has all of its poles and zeros in the left half plane (LHP), while the all-pass part has its poles in the LHP and mirrored zeros in the RHP. Thus we place a removable pole zero pair symmetrically across from the RHP zero, and then write the expression as the product, that is \( F(s) = M(s) \cdot A(s) \):

\[
F(s) = \frac{(s + 1)(s - 1)}{(s + 2)} \cdot \frac{s + 1}{s + 1} = \frac{(s + 1)^2}{s + 2} \cdot \frac{s - 1}{s + 1}
\]
Table 3.1: The following table provides an extended table of Laplace Transforms. \( J_0, K_1 \) are Bessel functions of the first and second kind.

<table>
<thead>
<tr>
<th>( f(t) ) ↔ ( F(s) )</th>
<th>Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>( \delta(at) \ ↔ \frac{1}{a} )</td>
<td>( a \neq 0; ) Time-scaled Dirac</td>
</tr>
<tr>
<td>( \delta(t + T_o) \ ↔ e^{sT_o} )</td>
<td>negative delay</td>
</tr>
<tr>
<td>( u(at) \ ↔ \frac{a}{s} )</td>
<td>( a \neq 0; ) dilate</td>
</tr>
<tr>
<td>( u(-t) \ ↔ -\frac{1}{s} )</td>
<td>anti-causal step</td>
</tr>
<tr>
<td>( e^{at}u(-t) \ ↔ \frac{1}{-s + a} )</td>
<td>anticausal damped step</td>
</tr>
</tbody>
</table>

\[
\frac{d^{1/2}}{dt^{1/2}} f(t)u(t) \leftrightarrow \sqrt{s} F(s)
\]

\[
\frac{d^{1/2}}{dt^{1/2}} u(t) \leftrightarrow \sqrt{s}
\]

\[
\frac{1}{\sqrt{\pi t}} u(t) \leftrightarrow \frac{1}{\sqrt{s}}
\]

\[
erfc(\alpha\sqrt{t}) \leftrightarrow \frac{1}{s} e^{-2\alpha\sqrt{s}} \quad (\text{Morse-Feshbach-II, p. 1582})
\]

\[
a > 0; \ erfc
\]

| \( J_0(at) u(t) \ ↔ \frac{1}{s^2 + a^2} \) | Bessel |
| \( J_1(t)u(t)/t \ ↔ \sqrt{s^2 + 1} - s \) |
| \( J_1(t)u(t)/t + 2u(t) \ ↔ \sqrt{s^2 + 1} + s = e^{\sinh^{-1}(s)} \) |
| \( \delta(t) + J_1(t)u(t)/t \ ↔ \sqrt{s^2 + 1} \) |
| \( I_0(t)u(t) \ ↔ 1/\sqrt{s^2 - 1} \) |
| \( u(t)/\sqrt{t + 1} \ ↔ e^{s \sqrt{\frac{\pi}{s}} \text{erfc}(\sqrt{s})} \) |
| \( \sqrt{tu(t)} \ast \sqrt{1 + tu(t)} \ ↔ e^{\sqrt{s/2}} K_1(s/2)/2s \) |
Thus \( M(s) \equiv \frac{(s+1)^2}{s+2} \) and \( A(s) \equiv \frac{s-1}{s+1} \)

2. Find the magnitude of \( M(s) \) Take the real part of the log of \( M \) and then the anti-log. Thus
\[
|M| = e^{\Re \ln M(s)}
\]

3. Find the phase of \( M(s) \) In this case we use the imaginary part:
\[
\angle M = \Im \ln M(s)
\]

4. Find the magnitude of \( A(s) \) 1, by definition.

5. Find the phase of \( A(s) \angle A = \Im \ln (A) \)

**More questions**

There are a number of question I would like to address in this report. Some of these are

1. Can we interpret the zeta function as a frequency domain quantity, and then inverse transform it into the time domain?

   The answer to this is yes, and the results are quite interesting.

2. Make a histogram of the entropy for the first million integers.

   This is a 5 minute job in Matlab/Octave. It goes something line this:

   ```matlab
   K=1e5; N=1:K; F=zeros(K,10);  
   for n=1:K; 
     f=factor(n); 
     F(n,1:length(f))=f; 
   end; 
   hist(F); ```
Entropy analysis

According to the Fundamental theorem of arithmetic, every integer may be uniquely written as the product of primes. Thus \( n = \prod_k p_i^{\alpha_k} \) (e.g., \( 12 = 2^2 \cdot 3 \) with \( \alpha_2 = 2 \) and \( \alpha_3 = 1 \)). If one thinks of \( K_n \) as a random variable on \( n \), then the constant \( K_n \) may be characterized by the concept of entropy. Thus each integer is associated with an entropy, defined by \( H_n = \sum K_n \log_2(K_n) \).

The zeta function The zeta function depends explicitly on the primes, which makes it very important. In 1737 Euler proposed the real-valued function \( \zeta(x) \in \mathbb{R} \) and \( x \in \mathbb{R} \), to prove that the number of primes is infinite (Goldstein, 1973). Euler’s definition of \( \zeta(x) \in \mathbb{R} \) is given by the power series

\[
\zeta(x) = \sum_{n=1}^{\infty} \frac{1}{n^x} \quad \text{for} \ x > 0 \in \mathbb{R}. \tag{3.29}
\]

This series converges for \( x > 0 \), since \( R = n^{-x} < 1, \, n > 1 \in \mathbb{N} \).

In 1860 Riemann extended the zeta function into the complex plane, resulting in \( \zeta(s) \), defined by the complex analytic power series, identical to the Euler formula, except \( x \in \mathbb{R} \) has been replaced by \( s \in \mathbb{C} \)

\[
\zeta(s) \equiv \frac{1}{1^s} + \frac{1}{2^s} + \frac{1}{3^s} + \frac{1}{4^s} + \cdots = \sum_{n=1}^{\infty} \frac{1}{n^s} = \sum_{n=1}^{\infty} n^{-s} \quad \text{for} \ \Re\{s\} = \sigma > 0. \tag{3.30}
\]

This formula converges for \( \Re\{s\} > 1 \) (Goldstein, 1973). To determine the formula in other regions of the \( s \) plane one, must extend the series via analytic continuation. As it turns out, Euler’s formulation provided detailed information about the structure of primes, going far beyond his original goal.

Euler product formula

As was first published by Euler in 1737, one may recursively factor out the leading prime term, resulting in Euler’s product formula. Euler’s procedure is an algebraic implementation of the sieve of Eratosthenes (Section 1.3.3, p. 40 and Section 1.6.1, page 128).

Multiplying \( \zeta(s) \) by the factor \( 1/2^s \), and subtracting from \( \zeta(s) \), removes all the even terms \( \propto 1/(2n)^s \) (e.g., \( 1/2^s + 1/4^s + 1/6^s + 1/8^s + \cdots \))

\[
\left(1 - \frac{1}{2^s}\right) \zeta(s) = 1 + \frac{1}{2^s} + \frac{1}{3^s} + \frac{1}{4^s} + \frac{1}{5^s} + \cdots - \left(\frac{1}{2^s} + \frac{1}{4^s} + \frac{1}{6^s} + \frac{1}{8^s} + \frac{1}{10^s} + \cdots\right), \tag{3.31}
\]

which results in

\[
\left(1 - \frac{1}{2^s}\right) \zeta(s) = 1 + \frac{1}{3^s} + \frac{1}{5^s} + \frac{1}{7^s} + \frac{1}{9^s} + \frac{1}{11^s} + \frac{1}{13^s} + \cdots. \tag{3.32}
\]

Repeating this with a lead factor \( 1/3^s \) applied to Eq. 3.32 gives \( 6 \)

\[
\frac{1}{3^s} \left(1 - \frac{1}{2^s}\right) \zeta(s) = \frac{1}{3^s} + \frac{1}{9^s} + \frac{1}{15^s} + \frac{1}{21^s} + \frac{1}{27^s} + \frac{1}{33^s} + \cdots. \tag{3.33}
\]

\(^{3}\)Sanity check: For example let \( n = 2 \) and \( x > 0 \). Then \( R = 2^{-x} < 1 \), where \( \epsilon \equiv \lim x \to 0^+ \). Taking the log gives \( \log_2 R = -\epsilon \log_2 2 = -\epsilon < 0 \). Since \( \log R < 0, R < 1 \).

\(^{6}\)This is known as Euler’s sieve, as distinguish from the Eratosthenes sieve.
Subtracting Eq. 3.33 from Eq. 3.32 cancels the RHS terms of Eq. 3.32

\[
\left(1 - \frac{1}{3s}\right) \left(1 - \frac{1}{2s}\right) \zeta(s) = 1 + \frac{1}{5s} + \frac{1}{7s} + \frac{1}{11s} + \frac{1}{13s} + \frac{1}{17s} + \frac{1}{19s} \cdots.
\]

Further repeating this process, with prime scale factors, (i.e., \(1/5^s, 1/7^s, \cdots, 1/\pi_k^s, \cdots\)), removes all the terms on the RHS but 1, results in Euler’s analytic product formula \((s = x \in \mathbb{R})\) and Riemann’s complex analytic product formula

\[
\zeta(s) = \prod_{k} \frac{1}{1 - \pi_k^{-s}} = \frac{1}{1 - 2^{-s}} \cdot \frac{1}{1 - 3^{-s}} \cdot \frac{1}{1 - 5^{-s}} \cdot \frac{1}{1 - 7^{-s}} \cdots \frac{1}{1 - \pi_n^{-s}} \cdots
\]

(3.34)

where \(\pi_k\) represents the \(k^{th}\) prime. The above defines each prime factor

\[
\zeta_k(s) = \frac{1}{1 - \pi_k^{-s}}
\]

(3.35)

as the \(k^{th}\) term of the complex analytic Riemann product formula, having an RoC of \(|\pi_k^{-s}| < 1\), but valid for \(|\pi_k^{-s}| \neq 1\). Each recursive step in this construction assures that the lead term, along with all of its multiplicative factors, are subtracted out, just as with the cancellations with the sieve of Eratosthenes. It is instructive to compare each iteration with that of the sieve (Fig. 1.6, p. 41).

**Exercise:** Work out the RoC for \(k = 2\). Sol: Todo

**Exercise:** Show how to construct \(\text{Zeta}_2(t) \leftrightarrow \zeta_2(s)\) by working in the time domain. The basic rules of building a sieve is to sum the integers

\[
S_1 = \sum_{n=1}^{\infty} n2^{n-1} = 1 \cdot 2^0 + 2 \cdot 2^1 + 3 \cdot 2^2 \cdots,
\]

while the sive for the \(k^{th}\) prime \(\pi_k\) is

\[
S_k = \sum_{n=1}^{\infty} n\pi_k^{n-1} = 1 \cdot \pi_k^0 + \pi_k \cdot 2^1 + \pi_k \cdot 2^2 \cdots.
\]

This sum may be written in terms of the convolution with the step function \(u_k\) since

\[
u_k \star u_k = n u_k = 0 \cdot u_0 + 1 \cdot u_1 + \cdots k u_k + \cdots.
\]

Here \(u_k = 1 = \sum_{n=0}^{\infty} \delta_k\) for \(k \geq 0\) and zero for \(k < 0\), and \(\delta_0 = 1\), and 0 for \(k \neq 0\).
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u+jv = 1./(1-exp(-s*pi))

Figure 3.4: Plot of \( w(s) = \frac{1}{1-e^{-s*\pi}} \) (note \( \pi_1 = 2 \)), which is factor \( \zeta_1(s) \) of Eq. 3.34. Here \( w(s) \) has poles where \( e^{s_n\ln 2} = 1 \), namely where \( \omega_n\ln 2 = n2\pi \), as seen in the colorized-map (\( s = \sigma + \omega j \) is the Laplace frequency [rad]).

Poles of \( \zeta_k(s) \)

Discuss RoC for each term in the product representation, and the anti-causal nature of \( \zeta_k(1-s) \).

Riemann proposed that Euler’s zeta function \( \zeta(s) \in \mathbb{C} \) have a complex argument (first actually explored by Chebyshev in 1850, (Bombieri, 2000)), extending \( \zeta(s) \) into the complex plane, (\( s \in \mathbb{C} \)), making it a complex analytic function.

Given that \( \zeta(s) \) is a complex analytic function, one might naturally wonder if \( \zeta(s) \) has an inverse Laplace transform. There seems to be very little written on this topic (Hill, 2007). We shall explore this question further here.

One may now identify the poles of \( \zeta_k(s) \) (\( p \in \mathbb{N} \)), which are required to determining the RoC. For example, the \( k^{th} \) factor of Eq. 3.34, expressed as an exponential, is

\[
\zeta_k(s) \equiv \frac{1}{1-\pi_k^{-s}} = \frac{1}{1-e^{-sT_k}} = \sum_{k=0}^{\infty} e^{-skT_k},
\]

(3.36)

where \( T_k \equiv \ln \pi_k \). Thus \( \zeta_p(s) \) has poles at \( -s_nT_p = 2\pi n j \) (when \( e^{-sT_p} = 1 \)), thus

\[
\omega_n = \frac{2\pi n}{T_k},
\]

with \(-\infty < n \in \mathbb{Z} < \infty \). These poles are the eigen-modes of the zeta function. A domain-colorized plot of this function is provided in Fig. 3.4. It is clear that the RoC of \( \zeta_k \) is \( > 0 \). It would be helpful to determine why \( \zeta(s) \) as such a more restrictive RoC than each of its factors.

Inverse Laplace transform

The inverse Laplace transform of Eq. 3.36 is an infinite series of delays of delay \( T_p \) (Table 3.1, p. 216)\(^7\)

\[
Z^\text{eta}_{\alpha}(t) = \delta(t)T_p \equiv \sum_{k=0}^{\infty} \delta(t-kT_p) \leftrightarrow \frac{1}{1-e^{-sT_p}}.
\]

(3.37)

\(^7\)Here we use a shorthand double-parentheses notation \( f(t)T \equiv \sum_{k=0}^{\infty} f(t-kT) \) to define the one-sided infinite sum.
Inverse transform of Product of factors

The time domain version of Eq. 3.34 may be written as the convolution of all the $Z_{eta}^e(t)$ factors

$$Z_{eta}^e(t) = Z_2^e * Z_3^e(t) * Z_5^e(t) * Z_7^e(t) * \cdots * Z_p^e(t) * \cdots,$$  \hspace{1cm} (3.38)

where $*$ represents time convolution (Table F.3, p. 278).

Physical interpretation

Such functions may be generated in the time domain as shown in Fig. 3.5, using a feedback delay of $T_p [s]$ as described by the equation in the figure, with a unity feedback gain $\alpha = 1$.

$$Z_{eta}^e(t) = Z_{eta}^e(t - T_p) + \delta(t).$$

Taking the Laplace transform of the system equation we see that the transfer function between the state variable $q(t)$ and the input $x(t)$ is given by $Z_{eta}^e(t)$. Taking the $\mathcal{LT}$, we see that $\zeta(s)$ is an all-pole function

$$\zeta_p(s) = e^{-sT_p} \zeta_p(s) + 1(t), \text{ or } \zeta_p(s) = \frac{1}{1 - e^{-sT_p}}.$$  \hspace{1cm} (3.39)

Discussion: In terms of the physics, these transmission line equations are telling us that $\zeta(s)$ may be decomposed into an infinite cascade of transmission lines (Eq. 3.38), each having a unique delay given by $T_k = \ln \pi_k$, $\pi_k \in \mathcal{P}$, the log of the primes. The input admittance of this cascade may be interpreted as an analytic continuation of $\zeta(s)$ which defines the eigen-modes of that cascaded impedance function.

Working in the time domain provides a key insight, as it allows us to determine the analytic continuation of the infinity of possible continuations, which are not obvious in the frequency domain. Transforming to the time domain is a form of analytic continuation of $\zeta(s)$, that depends on the assumption that $Z_{eta}^e(t)$ is one-sided in time (causal).

Additional relations: Some important relations provided by both Euler and Riemann (1859) are needed when studying $\zeta(s)$.

With the goal of generalizing his result, Euler extended the definition with the functional equation

$$\zeta(s) = 2^s \pi^{s-1} \sin \left( \frac{\pi s}{2} \right) \Gamma(1-s) \zeta(1-s).$$  \hspace{1cm} (3.40)
This seems closely related to Riemann’s *time reversal symmetry* properties (Bombieri, 2000)

\[ \pi^{-s/2}\Gamma\left(\frac{s}{2}\right)\zeta(s) = \pi^{-(1-s)/2}\Gamma\left(\frac{1-s}{2}\right)\zeta(1-s). \]

This equation is of the form \( F\left(\frac{s}{2}\right)\zeta(s) = F\left(\frac{1-s}{2}\right)\zeta(1-s) \) where \( F(s) = \Gamma(s)/\pi^s \).

As shown in Table F.3 the \( \mathcal{L}^{-1}(f(t)) \leftrightarrow F(-s) \) is simply a time-reversal. This leads to a causal and anti-causal function that are symmetric about \( \Re\{s\} = 1/2 \) (Riemann, 1859). It seems likely this is an important insight into the Euler’s functional equation.

Riemann (1859, page 2) provides an alternate integral definition of \( \zeta(s) \), based on the complex contour integration

\[ 2\sin(\pi s)\Gamma(s-1)\zeta(s) = \int_{x=-\infty}^{x=\infty} \frac{(-x)^{s-1}}{e^x - 1} dx. \]

Given the \( \zeta_k(s) \) it seems important to look at the inverse \( \mathcal{L} \) of \( \zeta_k(1-s) \), to gain insight into the analytically extended \( \zeta(s) \)

**Integral definition of the complex Gamma function** \( \Gamma(s) \)  

The definition of the complex analytic Gamma function (p. 278)

\[ \Gamma(s + 1) = s\Gamma(s) \equiv \int_0^{\infty} \xi^s e^{-\xi} d\xi, \]

which is a generalization of the real integer factorial function \( n! \).

**Zeros of \( \zeta(s) \)**  

We are still left with the most important question of all “Where are the zeros of \( \zeta(s) \)?” Equation 3.39 has no zeros, it is an all-pole system. The cascade of many such systems is also all-pole. As I see it, the issue is what is the actual formula for \( \zeta(s) \)?

### 3.4.4 Lec 20b: \( \mathcal{L} \) of the Riemann zeta function

This analytic function is a strong argument for the need for a deeper understanding of the analytic function. Just like the Pythagorean theorem is important to all mathematics, the zeta function is important to analysis, with many streams of analysis emanating from this form. For example the analytic Gamma function \( \Gamma(s) \) is a generalization of the factorial by the relationship

\[ n! = n(n-1)! \Rightarrow \Gamma(s) = s\Gamma(s-1). \]

Another useful relationship is

\[ \sum_{k=n}^{\infty} k = nu_n = u_n \star u_n \]

where the \( \star \) represents convolution. If this is treated in the frequency domain we obtain \( z \)-transforms of a very simple second-order pole

\[ nu_n \leftrightarrow \frac{2}{(z-1)^2}. \]

---

8Verify Riemann’s use of \( x \), which is taken to be real rather than complex. This could be more natural (i.e., modern Laplace transformation notation) if \( -x \rightarrow yj \rightarrow z \).

9Need to verify the exact form of these relationships, not work from my memory.
This follows from the geometric series

\[
\frac{1}{1-z} = \sum_n z^n
\]

with \( z = e^s \), and the definition of convolution.

![Figure 3.6: Angle of Riemann Zeta function \( \angle \zeta(z) \) as a function of complex \( z \). Red \( \Rightarrow \angle \zeta(z) < \pm \pi/2 \)](image)

The Laplace transform does not require that the series converge, rather that the series have a region of convergence that is properly specified. Thus the non-convergent series \( n u_n \) is perfectly well defined, just like

\[
t u(t) = u(t) \ast u(t) \leftrightarrow \frac{1!}{s^2}
\]

is well defined, in the Laplace transform sense. More generally

\[
t^n u(t) \leftrightarrow \frac{n!}{s^{n+1}}.
\]

From this easily understood relationship we can begin to understand \( \Gamma(p) \), as the analytic extension of the factorial. Its definition is simply related to the inverse Laplace transform, which is an integral. But to go there we must be able to think in the complex frequency domain. In fact we have the very simple definition for \( \Gamma(p) \) with \( p \in \mathbb{C} \)

\[
t^{p-1} u(t) \leftrightarrow \frac{\Gamma(p)}{s^p}
\]

which totally explains \( \Gamma(p) \). Thinking in the time domain is crucial for understanding. Since

\[
\sum_{k=0}^{\infty} \delta(t - k T_p) \leftrightarrow \zeta_p(s) = \frac{1}{1 - e^{-sT_p}} = \sum_0^{\infty} e^{-s n T_p}, \quad (3.41)
\]

by changing the sign of \( s \) we have

\[
\sum_{k=0}^{\infty} \delta(t + k T_p) \leftrightarrow \zeta_p(-s) = \frac{1}{1 - e^{sT_p}} = \sum_0^{\infty} e^{s n T_p}, \quad (3.42)
\]
Alternatively we may “shift” $s \rightarrow s + 1$, resulting in
\[
\sum_{k=0}^{\infty} \delta(t - kT_p) \leftrightarrow \zeta_p(s - 1) = \frac{1}{1 - e^{-(s-1)T_p}} = \sum_{0}^{\infty} e^{nT_p} e^{-snT_p},
\] (3.43)

### 3.5 Week 8

#### 3.5.1 Lec 21 The ten postulates of System of algebraic Networks

Physical systems obey very important rules, that follow from the physics. It is helpful to summarize these physical restrictions in terms of postulates, presented in terms of a taxonomy, or categorization method, of the fundamental properties of physical systems. Nine of these are listed below. These nine come from a recently published paper (Kim and Allen, 2013). It is possible that given time, others could be added.

A taxonomy of physical systems comes from a systematic summary of the laws of physics, which includes at least the nine basic network postulates, described in Section 1.4.17. To describe each of the network postulates it is helpful to begin with the 2-port transmission (aka ABCD, chain) matrix representation, discussed in Section 3.3.2 (p. 208).

![Figure 3.7: The schematic representation of an algebraic network, defined by its 2-port ABCD transmission, having three elements (i.e., Hunt parameters): $Z_e(s)$, the electrical impedance, $z_m(s)$, the mechanical impedances, and $T(s)$, the transduction coefficient (Hunt, 1952; Kim and Allen, 2013). matrix of an electro-mechanic transducer network. The port variables are $\Phi(f)$, $I(f)$: the frequency domain voltage and current, and $F(f)$, and $U(f)$: the force, and velocity. The matrix ‘factors’ the 2-port model into three 2x2 matrices, separating the three physical elements as matrix algebra. It is a standard impedance convention that the flows $I(f), U(f)$ are always defined into each port. Thus it is necessary to apply a negative sign on the velocity $-U(f)$ so that it has an outward flow, as required to match the next cell with its inward flow.](figures/as/3.7)

As shown in Fig. 3.7, the 2-port transmission matrix for an acoustic transducer (loudspeaker) is characterized by the equation
\[
\begin{bmatrix}
\Phi \\
\bar{I}
\end{bmatrix} = 
\begin{bmatrix}
A(s) & B(s) \\
C(s) & D(s)
\end{bmatrix}
\begin{bmatrix}
F_l \\
-U_l
\end{bmatrix} = 
\frac{1}{T} 
\begin{bmatrix}
z_m(s) & z_e(s)z_m(s) + T^2 \\
z_e(s) & 1
\end{bmatrix}
\begin{bmatrix}
F_l \\
-U_l
\end{bmatrix},
\] (3.44)

shown as a product of three 2x2 matrices in the figure, with each factor representing one of the three physical elements (i.e., Hunt parameters) of the loudspeaker.

This figure represents the electromechanical motor of the loudspeaker, consists of three elements, the electrical input impedance $Z_e(s)$, a gyrator, which is similar to a transformer, but relates current to force, and an output mechanical impedance $Z_m(s)$. This circuit describes what is needed to fully characterize it operation, from electrical input to mechanical (acoustical) output.

The input is electrical (voltage and current) $[\Phi_i, I_i]$ and the output (load) are the mechanical (force and velocity) $[F_l, U_l]$. The first matrix is the general case, expressed in terms of four unspecified functions $A(s), B(s), C(s), D(s)$, while the second matrix is for the specific example of Fig. 3.7. The four entries are the electrical driving point impedance $Z_e(s)$, the mechanical impedance $z_m(s)$ and the transduction $T = B_0l$ where $B_0$ is the magnetic flux strength and $l$ is the
length of the wire crossing the flux. Since the transmission matrix is anti-reciprocal, its determinate $\Delta_T = -1$, as is easily verified.

Other common transduction examples of cross-modality transduction include current–thermal (thermoelectric effect) and force–voltage (piezoelectric effect). These systems are all reciprocal, thus the transduction has the same sign.

**Impedance matrix**

These nine postulates describe the properties of a system having an input and an output. For the case of an electromagnetic transducer (Loudspeaker) the system is described by the 2-port, as shown in Fig. 3.7. The electrical input impedance of a loudspeaker is $Z_e(s)$, defined by

$$Z_e(s) = \left. \frac{V(\omega)}{I(\omega)} \right|_{U=0}.$$  

Note that this *driving-point impedance* must be causal, thus it has a Laplace transform and therefore is a function of the complex frequency $s = \sigma + j\omega$, whereas the Fourier transforms of the voltage $V(\omega)$ and current $I(\omega)$ are functions of the real radian frequency $\omega$, since the time-domain voltage $v(t) \leftrightarrow V(\omega)$ and the current $i(t) \leftrightarrow I(\omega)$ are signals that may start and stop at any time (they are not typically causal).

The corresponding 2-port *impedance matrix* for Fig. 3.7 is

$$\begin{bmatrix} \Phi_i \\ F_i \end{bmatrix} = \begin{bmatrix} z_{11}(s) & z_{12}(s) \\ z_{21}(s) & z_{22}(s) \end{bmatrix} \begin{bmatrix} I_i \\ U_i \end{bmatrix} = \begin{bmatrix} Z_e(s) & -T(s) \\ T(s) & z_m(s) \end{bmatrix} \begin{bmatrix} I_i \\ U_i \end{bmatrix}. \quad (3.45)$$

Such a description allows one to define Thévenin parameters, a very useful concept used widely in circuit analysis and other network models from other modalities.

The impedance matrix is an alternative description of the system, but with generalized forces $[\Phi_i, F_i]$ on the left and generalized flows $[I_i, U_i]$ on the right. A rearrangement of the equations allows one to go from the ABCD to impedance set of parameters (Van Valkenburg, 1964b). The electromagnetic transducer is anti-reciprocal (P6), $z_{12} = -z_{21} = T = B_0 l$.

**Taxonomy of algebraic networks**

The postulates must go beyond postulates P1-P6 defined by Carlin and Giordano (Section 1.4.17, p. 103), when there are interaction of waves and a structured medium, along with other properties not covered by classic network theory. Assuming QS, the wavelength must be large relative to the medium’s lattice constants. Thus the QS property must be extended to three dimensions, and possibly to the cases of an-isotropic and random media.

**Causality: P1**  
As stated above, due to causality the negative properties (e.g., negative refractive index) must be limited in bandwidth, as a result of the Cauchy-Riemann conditions. However even causality needs to be extended to include the delay, as quantified by the d’Alembert solution to the wave equation, which means that the delay is proportional to the distance. Thus we generalize P1 to include the space dependent delay. When we wish to discuss this property we denote it *Einstein causality*, which says that the delay must be proportional to the distance $x$, with impulse response $\delta(t - x/c)$. 

Linearity: P2  The wave properties of may be non-linear. This is not restrictive as most physical systems are naturally nonlinear. For example, a capacitor is inherently nonlinear: as the charge builds up on the plates of the capacitor, a stress is applied to the intermediate dielectric due to the electrostatic force $F = qE$. In a similar manner, an inductor is nonlinear. Two wires carrying a current are attracted or repelled, due to the force created by the flux. The net force is the product of the two fluxes due to each current.

In summary, most physical systems are naturally nonlinear, it’s simply a matter of degree. An important counter example is a amplifier with negative feedback, with very large open-loop gain. There are, therefore, many types of non-linear, instantaneous and those with memory (e.g., hysteresis). Given the nature of P1, even an instantaneous non-linearity may be ruled out. The linear model is so critical for our analysis, providing fundamental understanding, that we frequently take P1 and P2 for granted.

Passive/Active: P3  This postulate is about conservation of energy and Otto Brune’s positive Real (PR aka physically realizable) condition, that every passive impedance must obey. Following up on the earlier work of his primary PhD thesis advisor Wilhelm Cauer (1900-1945) and Ernst Guillemin, along with Norbert Weiner and Vannevar Bush at MIT, Otto Brune mathematically characterized the properties of every PR 1-port driving point impedance (Brune, 1931b).

When the input resistance of the impedance is real, the system is said to be passive, which means the system obeys conservation of energy. The real part of $Z(s)$ is positive if and only if the corresponding reflectance is less than 1 in magnitude. The definition of the reflectance of $Z(s)$ is defined as a bilinear transformation of the impedance, normalized by its surge resistance $r_0$ (Campbell, 1903)

$$\Gamma(s) = \frac{Z(s) - r_0}{Z(s) + r_0} = \frac{\hat{Z} - 1}{\hat{Z} + 1}$$

where $\hat{Z} = Z/r_0$. The surge resistance is defined in terms of the inverse Laplace transform of $Z(s) \leftrightarrow z(t)$, which must have the form

$$z(t) = r_0 \delta(t) + \rho(t),$$

where $\rho(t) = 0$ for $t < 0$. It naturally follows that $\gamma(t) \Leftrightarrow \Gamma(s)$ is zero for negative and zero time, namely $\gamma(0) = 0, t \leq 0$, at

Given any linear PR impedance $Z(s) = R(\sigma, \omega) + jX(\sigma, \omega)$, having real part $R(\sigma, \omega)$ and imaginary part $X(\sigma, \omega)$, the impedance is defined as being PR (Brune, 1931b) if and only if

$$\Re Z(s) = R(\sigma \geq 0, \omega) \geq 0. \quad (3.46)$$

That is, the real part of any PR impedance is non-negative everywhere in the right half $s$ plane ($\sigma \geq 0$). This is a very strong condition on the complex analytic function $Z(s)$ of a complex variable $s$. This condition is equivalent to any of the following statements (Van Valkenburg, 1964a):

1. There are no poles or zeros in the right half plane ($Z(s)$ may have poles and zeros on the $\sigma = 0$ axis).

2. If $Z(s)$ is PR then its reciprocal $Y(s) = 1/Z(s)$ is PR (the poles and zeros swap).

3. If the impedance may be written as the ratio of two polynomials (a limited case, related to the quasi-statics approximation, P9) having degrees $N$ and $L$, then $|N - L| \leq 1$. 
4. The angle of the impedance \( \theta \equiv \angle Z \) lies between \([-\pi \leq \theta \leq \pi]\).

5. The impedance and its reciprocal are complex analytic in the right half plane, thus they each obey the Cauchy Riemann conditions there.

**Energy and Power:** The PR (positive real or Physically realizable) condition assures that every impedance is positive-definite (PD), thus guaranteeing conservation of energy is obeyed (Schwinger and Saxon, 1968, p.17). This means that the total energy absorbed by any PR impedance must remain positive for all time, namely

\[
E(t) = \int_{-\infty}^{t} v(t) i(t) \, dt = \int_{-\infty}^{t} i(t) \ast z(t) i(t) \, dt > 0,
\]

where \( i(t) \) is any current, \( v(t) = z(t) \ast i(t) \) is the corresponding voltage and \( z(t) \) is the real causal impulse response of the impedance, e.g., \( z(t) \leftrightarrow Z(s) \) are a Laplace Transform pair. In summary, if \( Z(s) \) is PR, \( E(t) \) is PD.

As discussed in detail by Van Valkenburg, any rational PR impedance can be represented as a partial fraction expansion, which can be expanded into first-order poles as

\[
Z(s) = K \prod_{i=1}^{L} \frac{1}{s - n_i} \prod_{k=1}^{N} \frac{1}{s - d_k} = \sum_{n} \frac{\rho_n}{s - s_n} e^{j(\theta_n - \theta_d)},
\]

(3.47)

where \( \rho_n \) is a complex scale factor (residue). Every pole in a PR function has only simple poles and zeros, requiring that \(|L - N| \leq 1\) (Van Valkenburg, 1964b).

Whereas the PD property clearly follows P3 (conservation of energy), the physics is not so clear. Specifically what is the physical meaning of the specific constraints on \( Z(s) \)? In many ways, the impedance concept is highly artificial, as expressed by P1-P7.

When the impedance is not rational, special care must be taken. An example of this is the semi-inductor \( M \sqrt{s} \) and semi-capacitor \( K/\sqrt{s} \) due, for example, to the skin effect in EM theory and viscous and thermal losses in acoustics, both of which are frequency dependent boundary-layer diffusion losses (Vanderkooy, 1989). They remain positive-real but have a branch cut, thus are double valued in frequency.

**Real time response:** P4  The impulse response of every physical system is real, vs. complex. This requires that the Laplace Transform have conjugate-symmetric symmetry \( H(s) = H^\ast(s^\ast) \), where the \( \ast \) indicates conjugation (e.g., \( R(\sigma, \omega) + X(\sigma, \omega) = R(\sigma, \omega) - X(\sigma, -\omega) \)).

**Time invariant:** P5  The meaning of time-invariant requires that the impulse response of a system does not change over time. This requires that the system coefficients of the differential equation describing the system are constant (independent of time).

**Rayleigh Reciprocity:** P6  Reciprocity is defined in terms of the unloaded output voltage that results from an input current. Specifically

\[
\begin{bmatrix}
  z_{11}(s) & z_{12}(s) \\
  z_{21}(s) & z_{22}(s)
\end{bmatrix}
= \frac{1}{C(s)} \begin{bmatrix}
  A(s) & \Delta_T \\
  1 & D(s)
\end{bmatrix},
\]

(3.48)
where $\Delta_T = A(s)D(s) - B(s)C(s) = \pm 1$ for the reciprocal and anti-reciprocal systems respectively. This is best understood in term of Eq. 3.45. The off-diagonal coefficients $z_{12}(s)$ and $z_{21}(s)$ are defined as

$$z_{12}(s) = \frac{\Phi_i}{U_l} \bigg|_{I_i=0} \quad z_{21}(s) = \frac{F_l}{I_i} \bigg|_{U_l=0}.$$  

When these off-diagonal elements are equal [$z_{12}(s) = z_{21}(s)$] the system is said to obey Rayleigh reciprocity. If they are opposite in sign [$z_{12}(s) = -z_{21}(s)$], the system is said to be anti-reciprocal. If a network has neither of the reciprocal or anti-reciprocal characteristics, then we denote it as non-reciprocal (McMillan, 1946). The most comprehensive discussion of reciprocity, even to this day, is that of Rayleigh (1896, Vol. I). The reciprocal case may be modeled as an ideal transformer (Van Valkenburg, 1964a) while for the anti-reciprocal case the generalized force and flow are swapped across the 2-port. An electromagnetic transducer (e.g., a moving coil loudspeaker or electrical motor) is anti-reciprocal (Kim and Allen, 2013; Beranek and Mellow, 2012), requiring a gyrator rather than a transformer, as shown in Fig. 3.7.

**Reversibility: P7** A second 2-port property is the reversible/non-reversible postulate. A reversible system is invariant to the input and output impedances being swapped. This property is defined by the input and output impedances being equal.

Referring to Eq. 3.48, when the system is reversible $z_{11}(s) = z_{22}(s)$ or in terms of the transmission matrix variables $\frac{A(s)}{C(s)} = \frac{D(s)}{C(s)}$ or simply $A(s) = D(s)$ assuming $C(s) \neq 0$.

An example of a non-reversible system is a transformer where the turns ratio is not one. Also an ideal operational amplifier (when the power is turned on) is non-reversible due to the large impedance difference between the input and output. Furthermore it is active (it has a power gain, due to the current gain at constant voltage) (Van Valkenburg, 1964b).

Generalizations of this lead to group theory, and Noether’s theorem. These generalizations apply to systems with many modes whereas quasi-statics holds when operate below a cutoff frequency (Table 3.2), meaning that like the case of the transmission line, there are no propagating transverse modes. While this assumption is never exact, it leads to highly accurate results because the non-propagating evanescent transverse modes are attenuated over a short distance, and thus, in practice, may be ignored (Montgomery et al., 1948; Schwinger and Saxon, 1968, Chap. 9-11).

We extend the Carlin and Giordano postulate set to include (P7) Reversibility, which was refined by Van Valkenburg (1964a). To satisfy the reversibility condition, the diagonal components in a system’s impedance matrix must be equal. In other words, the input force and the flow are proportional to the output force and flow, respectively (i.e., $Z_e = z_m$).

**Spatial invariant: P8** The characteristic impedance and wave number $\kappa(s, x)$ may be strongly frequency and/or spatially dependent, or even be negative over some limited frequency ranges. Due to causality, the concept of a negative group velocity must be restricted to a limited bandwidth (Brillouin, 1960). As is made clear by Einstein’s theory of relativity, all materials must be strictly causal (P1), a view that must therefore apply to acoustics, but at a very different time scale. We first discuss generalized postulates, expanding on those of Carlin and Giordano.

**The QS constraint: P9** When a system is described by the wave equation, delay is introduced between two points in space, which depends on the wave speed. When the wavelength is large compared to the delay, one may successfully apply the quasi-static approximation. This method has wide-spread application, and is frequency used without mention of the assumption. This can
lead to confusion, since the limitations of the approximation may not be appreciated. An example is the use of QS in Quantum Mechanics. The QS approximation has wide spread use when the signals may be accurately approximated by a band-limited signal. Examples include KCL, KVL, wave guides, transmission lines, and most importantly, impedance. The QS property is not mentioned in the six postulates of Carlin and Giordano (1964), thus they need to be extended in some fundamental ways.

When the dimensions of a cellular structure in the material are much less than the wavelength, can the QS approximation be valid. This effect can be viewed as a mode filter that suppresses unwanted (or conversely enhances the desired) modes (Ramo et al., 1965). QSs may be applied to a 3 dimensional specification, as in a semiconductor lattice. But such applications fall outside the scope of this text (Schwinger and Saxon, 1968).

Although I have never seen the point discussed in the literature, the QS approximation is applied when defining Green’s theorem. For example, Gauss’s Law is not true when the volume of the container violates QS, since changes in the distribution of the charge have not reached the boundary, when doing the integral. Thus such integral relationships assume that the system is in quasi steady-state (i.e., that QS holds).

Table 3.2: There are several ways of indicating the quasi-static (QS) approximation. For network theory there is only one lattice constant \( a \), which must be much less than the wavelength (wavelength constraint). These three constraints are not equivalent when the object may be a larger structured medium, spanning many wavelengths, but with a cell structure size much less than the wavelength. For example, each cell could be a Helmholtz resonator, or an electromagnetic transducer (i.e., an earphone).

<table>
<thead>
<tr>
<th>Measure</th>
<th>Domain</th>
</tr>
</thead>
<tbody>
<tr>
<td>( ka &lt; 1 )</td>
<td>Wavenumber constraint</td>
</tr>
<tr>
<td>( \lambda &gt; 2\pi a )</td>
<td>Wavelength constraint</td>
</tr>
<tr>
<td>( f_c &lt; c/2\pi a )</td>
<td>Bandwidth constraint</td>
</tr>
</tbody>
</table>

Formally, QS is defined as \( ka < 1 \) where \( k = 2\pi/\lambda = \omega/c \) and \( a \) is the cellular dimension or the size of the object. Other ways of expressing this include \( \lambda/4 > a \), \( \lambda/2\pi > a \), \( \lambda > 4a \) or \( \lambda > 2\pi a \). It is not clear if it is better to normalize \( \lambda \) by 4 (quarter-wavelength constraint) or \( \lambda/4 \approx 6.28 > 4 \), which is more conservative by a factor of \( \pi/2 \approx 1.6 \). Also \( k \) and \( a \) can be vectors, e.g., Eq. 1.24, p. 57, Section 1.5.1.

Schelkunoff may have been the first to formalize this concept (Schelkunoff, 1943) (but not the first to use it, as exemplified by the Helmholtz resonator). George Ashley Campbell was the first to use the concept in the important application of a wave-filter, some 30 years before Schelkunoff (Campbell, 1903). These two men were 40 years apart, and both worked for the telephone company (after 1929, called AT&T Bell Labs) (Fagen, 1975).

There are alternative definitions of the QS approximation, depending on the geometrical cell structure. The alternatives are outlined in Table 3.2.

Summary

A transducer converts between modalities. We propose the general definition of the nine system postulates, that include all transduction modalities, such as electrical, mechanical, and acoustical. It is necessary to generalize the concept of the QS approximation (P9) to allow for guided waves.

Given the combination of the important QS approximation, along with these space-time, linearity, and reciprocity properties, a rigorous definition and characterization a system can thus be
established. It is based on a taxonomy of such materials, formulated in terms of material and physical properties and in terms of extended network postulates.

3.5.2 Lec 22 Exam II (Evening)
Chapter 4

Scalar Differential Equations: Stream 3a

4.1  Week 8-continued

4.1.1  Lec 23  Integration in the complex plane

The period of analytic discovery:  Coming out of the dark ages, from algebra, to analytic geometry, to calculus.

Complex analytic functions: Cauchy I, II, III

Starting from real analytic functions by Euler, Cauchy moved to integration of complex analytic functions.
4.2  Week 9

4.2.1  Lec 24 Cauchy Riemann conditions
4.2.2  Lec 25a Complex analytic functions and Brune impedance (I)
4.2.3  Lec 25b Complex analytic functions and Brune impedance (II)
4.2.4  Lec 26: Summary: Branch points, cuts, Sheets & Transmission Lines

4.3  Week 10

4.3.1  Lec 27 Cauchy Theorem I
4.3.2  Lec 28 Cauchy Theorems: II, III
4.3.3  Lec 29 Inverse Laplace Transform: Causality $t < 0$

4.4  Week 11

4.4.1  Lec 30 Inverse Laplace Transforms II: $t > 0$
4.4.2  Lec 31 Laplace Transform properties
4.4.3  Lec 32: Review for Exam III
Chapter 5

Vector Calculus: Stream 3b

5.1 Stream 3b

5.1.1 Lec 33: Grad, Div, Curl and Laplacian

5.1.2 Lec 34: Scalar wave equation (I)

5.1.3 Lec 35: Scalar wave equation (II)

Equation 1.127 (p. 140) is the traditional Webster horn equation in terms of the average pressure. There is an alternative form of the Webster equation (Webster, 1919; Morse, 1948; Morse and Feshbach, 1953; Pierce, 1981), derived from Eq. 1.160 (p. 156, by taking the partial derivative (abbreviated \( \partial_z \)) of the average pressure, giving

\[
P_{xx} + \nabla_x^2 V + \nabla(s)V_x = 0.
\]

Next, using Newton’s and Hooke’s equations a second time, one may remove the velocity, obtaining the horn pressure wave equation

\[
P_{rr} - \left( \frac{Z_r}{Z} \right) P_r = \frac{s^2}{c_o^2} P \leftrightarrow \frac{1}{c_o} \varrho u.
\]

in terms of the Laplace Transform pair \( \varrho(r, t) \leftrightarrow P(r, \omega) \) for pressure, where

\[
\kappa(s) = \pm \sqrt{\mathcal{Z}(r, s) \mathcal{Y}(r, s)} = \pm \frac{s}{c_o}.
\]

The coefficient on \( P_r \) may be further analyzed, as it represents the damping term in the equation

\[
- \frac{\mathcal{Z}_r}{\mathcal{Z}} = -\partial_r \ln \mathcal{Z}_s(r, s)
\]

\[
= -\partial_r \left( \ln s \frac{\rho_o}{A(r)} \right)
\]

\[
= \partial_r \ln A(r)
\]

\[
= \frac{1}{A(r)} \frac{\partial}{\partial r} A(r)
\]

Equation 1.172 in fact shows that by integration by parts, this “loss” term in Eq. 5.1 is related to the area function in an interesting way, but further analysis is required to study this. How does the formula for the characteristic impedance (Eqs. 1.166 & 1.154, p. 157) come about? This is an important question that needs to be addressed.
Examples:
The primitive solutions for the four horns are summarized in Table 1.4. In each case we may start from Eq. 5.3 with the area $A(r)$ given for a particular case to find the differential equation for that geometry, and then find the primitive solutions.

Exercise: By direct differentiation show that $\delta(t \pm x/c_o) \leftrightarrow e^{\mp \kappa x}e^{st}$ are solutions of the 1-dimensional wave equation. First show this for the time-domain solution and then again for the frequency domain solution. Third shown that this pair of functions are Laplace transforms of each other.

Conical horn: For the spherical geometry $A(r) = A_o r^2$ (See Table 1.4), resulting in the spherical horn wave equation

$$\frac{1}{A_o r^2} \frac{\partial}{\partial r} \left( A_o r^2 \frac{\partial \varphi}{\partial r} \right) = \frac{\partial^2 \varphi}{c^2 \partial t^2}, \quad (5.3)$$

having primitive solutions

$$P^\pm = e^{\mp \kappa r} / r.$$

Here the $+$ is the “outbound” wave and the $-$ is the inbound wave. As we shall see, when the area is variable, each of the primitive solutions has local reflections due to this variation. This gives rise to a reactive mass-component in the radiation impedance.

Surge admittance: Every radiation admittance may be written as $y_{rad}(t) \leftrightarrow Y_{rad}(s)$ which may be further split into a real surge admittance\(^2^ \) $y_o \delta(t)$ (Campbell, 1922), and a causal remainder admittance $y_r(t)$, as\(^3^)

$$y_{rad}(t) = y_o \delta(t) + y_r(t).$$

Alternatively this may also be written as the sum of an impedance surge and remainder components $z_{rad}(t) = z_o \delta(t) + z_r(t)$. These functions are inverses of each other in the convolution sense, namely $y_{rad}(t) \ast z_{rad}(t) = \delta(t)$, which follows from the definition $Z_{rad}(s)Y_{rad}(s) = 1$. Any function having a causal inverse is said to be minimum phase thus every impedance and admittance must be minimum phase. The remainder characterizes the dispersive component of the impedance, thus when it is zero, the impedance is purely real (e.g., the reactance is zero).

Wave admittance $Y(x, s)$: The driving-point wave admittance, defined as the ratio of the volume velocity $V(r, s)$ to the average pressure $P(r, s)$ at any point $r$ along the range axis, may be interpreted as follows: If the horn were split at any point $x$ (i.e., a node is defined), the pressure at the two throats are the same ($P^+ = P^- \text{.}$). The velocities are defined as into the port, thus $V^+ = -V^- \text{.}$ Due to this definition of the flow into the port, the total velocity is the difference of these two driving point node velocities ($V = V^+ - V^-$). Consistent with Kirchhoff’s laws, the wave admittance as the sum of the two radiation admittances

$$Y(x, s) \equiv Y_{rad}^+(x, s) + Y_{rad}^-(x, s). \quad (5.4)$$

\(^1^)\text{For spherical case we typically take the radius } r \text{ as the range variable.}

\(^2^)\text{Since it is real it would best be called a surge conductance.}

\(^3^)\text{It seem obvious that } y_o \equiv Y? \text{?}
Impedance Matrix: For a finite section of horn, the \(2 \times 2\) impedance matrix (a generalization of Ohm’s law) may be expressed in terms of the ABCD matrix elements (Van Valkenburg, 1964b) as

\[
\begin{bmatrix}
\mathcal{P}_o \\
\mathcal{P}_L
\end{bmatrix}
= \frac{1}{\mathcal{C}(s)} \begin{bmatrix}
\mathcal{A}(s) & \mathcal{D}(s) \\
1 & -\mathcal{D}(s)
\end{bmatrix}
\begin{bmatrix}
\mathcal{V}_o \\
\mathcal{V}_L
\end{bmatrix}.
\] (5.5)

Note that \(\Delta_T = 1\) since the horn must be reciprocal (Morse and Feshbach, 1953; Hunt, 1982; Pierce, 1981).

While the Transmission (ABCD) matrix is convenient when modeling, the impedance matrix (and its inverse, the admittance matrix) are useful when one makes experimental measurements. For example

\[
\mathcal{Y}_o \bigg|_{\mathcal{V}_L = 0} \equiv \frac{\mathcal{C}(s)}{\mathcal{A}(s)} \quad \text{and} \quad \mathcal{Y}_L \bigg|_{\mathcal{V}_o = 0} \equiv \frac{\mathcal{C}(s)}{\mathcal{D}(s)}
\]

are the unloaded input admittances of the horn looking into the two ports (Eq. 1.155). These admittances are typically easily measured experimentally, given access to the endpoints.

In section 1.6.10 we work out these relationships for the trivial case of the 1D horn (Goldsmith and Minton, 1924; Olson, 1947). Then the cases for 2D, 3D and exponential horns are derived. The parameters for each of the cases is provided in Table 1.4.

Transmission (ABCD) matrix: The transmission matrix (Eq. 1.70, p. 92) is useful for computing the cascade of several systems, such as a horn driven by a Thévenin system and loaded by the radiation impedance or a cascade of several horns. The solution of a horn having finite length may be expressed in terms of a 2-port ABCD matrix, that relates the pressure and volume velocity at the input and output ports (the two ends) of the horn \((x = 0\) and \(x = L)\)

\[
\begin{bmatrix}
\mathcal{P}_o \\
\mathcal{V}_o
\end{bmatrix}
= \begin{bmatrix}
\mathcal{A}(s) & \mathcal{B}(s) \\
\mathcal{C}(s) & \mathcal{D}(s)
\end{bmatrix}
\begin{bmatrix}
\mathcal{P}_L \\
-\mathcal{V}_L
\end{bmatrix}. 
\] (5.6)

Note that \(\mathcal{A}(s) \equiv \frac{\mathcal{P}_o}{\mathcal{P}_L} \bigg|_{\mathcal{V}_L = 0}\) must not be confused with the horn area \(A(x)\) (note the different font).

By definition, the output velocity \(\mathcal{V}_L\), of an ABCD matrix is out of the port, hence the negative sign, since \(\mathcal{V}_o, \mathcal{V}_L\) are defined into their respective ports (Orfanidis, 2009). When the system is reversible, \(\mathcal{A} = \mathcal{D}\), reciprocal when \(\Delta_T \equiv \mathcal{A}\mathcal{D} - \mathcal{B}\mathcal{C} = 1\), and anti-reciprocal when \(\Delta_T = -1\). With the trivial exception of the uniform horn, all horns are non-reversible and reciprocal.

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2D parabolic Horn

For 2D cylindrical waves the area function is \(A(r) = A_o r\), (horn radius \(\propto \sqrt{r}\)) thus the Webster horn equation reduces to the cylindrical wave equation (Appendix 1.6.7.)

\[
\mathcal{P}_{rr}(r, \omega) + \frac{1}{r} \mathcal{P}_r(r, \omega) = \kappa^2(s) \mathcal{P}(r, \omega)
\]

having Bessel function primitive solutions

\[
\mathcal{P}^+(r, \omega) = J_0(\omega r/c) - iY_0(\omega r/c) = H^+_o(-jkr)
\]

and

\[
\mathcal{P}^-(r, \omega) = J_0(\omega r/c) + iY_0(\omega r/c) = H^-_o(-jkr),
\]
where $J_o$ and $Y_o$ are the standard two types Bessel functions (like $\cos()$ and $\sin()$), and $H_o^{\pm}(r)$ are the two kinds of Hankel function, of the first (-) and second (+) kind (similar to $e^{\pm i\kappa r}$), all of order zero (indicated by the subscript) (Salmon, 1946a; Olson, 1947; Morse and Feshbach, 1953).

2D input admittance: As shown in Fig. 5.1, the half-infinite section of parabolic horn has an input admittance given by Eq. 1.168

$$Y_{in}^\pm(r, s) = \mp \frac{\gamma}{\kappa} \frac{\partial}{\partial r} \ln H_o^\pm(-j\kappa r) = \mp j\gamma \frac{H^{\pm}_{1}}{H_o^{\pm}}.$$  \hspace{1cm} \text{(5.7)}

Figure 5.1: Real and imaginary parts of the normalized radiation admittance $Y_{in}/\gamma$ (upper) and impedance $Z_{in}/\gamma$ (lower), as a function of frequency, for a infinite section of horn, as computed from Eq. 5.7. The real and imaginary parts of the admittance are much simpler to characterize: the imaginary part is a mass (i.e., is $\propto 1/s$) while the real part $\to 1$ above the critical frequency (where the real and imaginary parts are equal) at $\approx 0.3$ [kHz].

2D ABCD Transmission matrix: Based on Eq. 1.152 (p. 152)

$$\begin{bmatrix} P_o \\ V_o \end{bmatrix} = \frac{-1}{\Delta L} \begin{bmatrix} Y_{in}^+ & P_{-L}^- \\ Y_{in}^- & P_{L}^+ \end{bmatrix} \begin{bmatrix} Y_{in}^- & -1 \\ Y_{in}^+ P_{L}^- & P_{L}^+ \end{bmatrix} \begin{bmatrix} P_L \\ V_L \end{bmatrix}. \hspace{1cm} \text{(5.8)}$$

Verify the following

$$\begin{bmatrix} P_o \\ V_o \end{bmatrix} = \frac{-1}{\Delta L} \begin{bmatrix} 1 \\ Y_{in}^+ H_o^-(r-L) \\ Y_{in}^- H_o^+(r-L) \end{bmatrix} \begin{bmatrix} Y_{in}^- H_o^+(r) & -1 \\ Y_{in}^- H_o^+(r) & Y_{in}^- H_o^+(r) \end{bmatrix} \begin{bmatrix} P_L \\ V_L \end{bmatrix}, \hspace{1cm} \text{(5.9)}$$

where the subscript on each matrix indicates the value of $r$ at which it is evaluated.$^4$

$^4$Note that

$$\frac{d}{dr} \ln H_o^\pm(kr) = -\frac{k H^{\pm}_{1}(kr)}{H_o^{\pm}(kr)}.$$
Impedance matrix:

\[
\begin{bmatrix}
P_o \\
P_L
\end{bmatrix} = \frac{1}{C(s)} \begin{bmatrix}
A(s) & 18 \\
D(s) & \end{bmatrix} \begin{bmatrix}
V_o \\
V_L
\end{bmatrix}.
\] (5.10)

5.2 Week 14

5.2.1 Lec 39 Geometry of gradient, divergence and curl vector operators

Geometry of gradient, divergence and curl vector operators

Laplacian operator in \( N \) dimensions

To show that the Webster equation is in agreement with the wave equation in 2 and three dimensions, we need to express the Laplacian, and then determine \( F(n) \) of Table 1.4. In general it may be shown that, in \( N \) dimensions (Sommerfeld, 1949, p. 227)\(^5\)

\[
\nabla_r^2 P \equiv \frac{1}{r^{N-1}} \frac{\partial}{\partial r} \left( r^{N-1} \frac{\partial P}{\partial r} \right)
\] (5.11)

This may be expanded as

\[
\frac{\partial^2 P}{\partial r^2} + F(r) \frac{\partial P}{\partial r},
\] (5.12)

where \( F(r) = (N - 1)/r \) is the same as in Table 1.4. In terms of the Webster Horn equation, \( F(x) \equiv -\partial \ln Z(x, s)/\partial x \). Thus we see there is a fundamental relation between \( Z \) and the dimensionality of the horn. For each \( N \), \( r^{N-1} \) is proportional to the area function \( A(x) \), thus is a special case of the Webster Horn equation Eq. 1.160.

When \( N = 3 \) (the conical Horn) there is an important special relationship

\[
\nabla_r^2 P = \frac{1}{r^2} \partial_r r^2 \partial_r P
\] (5.13)

\[
= \frac{1}{r} \partial_{rr} P
\] (5.14)

resulting in the general d’Alembert solutions

\[
P^{\pm}(r, s) = \frac{1}{r} e^{\pm \kappa(s)r}.
\]

Exercise: Prove this last result by expanding Eq. 5.13, 5.14 using the chain rule.

Sol: To show these are equal, expanded them using the chain rule.

- Expanding Eq. 5.13:

\[
\frac{1}{r^2} \partial_r r^2 \partial_r P = \frac{1}{r^2} \left( 2r + r^2 \partial_r \right) \partial_r P
\]

\[
= \frac{2}{r} P_r + P_{rr}.
\]

\(^5\)http://en.wikipedia.org/wiki/Laplacian#Three_dimensions
• Expanding Eq. 5.14:

\[
\frac{1}{r} \partial_r r \mathbf{P} = \frac{1}{r} \partial_r (\mathbf{P} + r \mathbf{P}_r) = \frac{1}{r} (\mathbf{P}_r + \mathbf{P} + r \mathbf{P}_{rr}) = \frac{2}{r} \mathbf{P}_r + \mathbf{P}_{rr}.
\]

### 5.2.2 Lec 41 Derivation of Maxwell’s vector wave equation

Here we do a full derivation of the vector wave equation for Maxwell’s equations, similar to the derivation of the Webster Horn equation.

Starting from Maxwell’s equations in free space, and using a matrix format

\[
\nabla \times \begin{bmatrix} E(x, t) \\ H(x, t) \end{bmatrix} = \partial_t \begin{bmatrix} -B(x, t) \\ +D(x, t) \end{bmatrix} = \begin{bmatrix} 0 & -\mu_o \\ \epsilon_o & 0 \end{bmatrix} \begin{bmatrix} E(x, t) \\ H(x, t) \end{bmatrix} \leftrightarrow \begin{bmatrix} 0 & -s\mu_o \\ s\epsilon_o & 0 \end{bmatrix} \begin{bmatrix} E(x, \omega) \\ H(x, \omega) \end{bmatrix}
\]

Working in the frequency domain, and taking the curl of both sides gives

\[
\nabla \times \nabla \times \begin{bmatrix} E \\ H \end{bmatrix} = \begin{bmatrix} 0 & -s\mu_o \\ s\epsilon_o & 0 \end{bmatrix} \nabla \times \begin{bmatrix} E \\ H \end{bmatrix} = \begin{bmatrix} 0 & -s\mu_o \\ s\epsilon_o & 0 \end{bmatrix} \begin{bmatrix} E \\ H \end{bmatrix} = -\frac{s^2}{c_o^2} \begin{bmatrix} E \\ H \end{bmatrix}
\]

Using the identity \( \nabla \times \nabla \times () = \nabla \nabla \cdot () - \nabla^2 () \) (Eq. 1.189, p. 168) we obtain

\[
\nabla^2 \begin{bmatrix} E \\ H \end{bmatrix} - \nabla \nabla \cdot \begin{bmatrix} E \\ H \end{bmatrix} = \frac{s^2}{c_o^2} \begin{bmatrix} E \\ H \end{bmatrix}
\]

or

\[
\nabla^2 \begin{bmatrix} E \\ H \end{bmatrix} - \frac{s^2}{c_o^2} \begin{bmatrix} E \\ H \end{bmatrix} = \nabla \left( \frac{1}{\epsilon_o} \nabla \cdot \mathbf{D} \right) = \frac{1}{\epsilon_o} \left( \nabla \rho(x, s) \right).
\]

This is the vector wave equation, with an electric excitation term \( \nabla \rho(x, s) \).

### 5.3 Week 15

#### 5.3.1 Lec 42: The Quasi-static approximation and more applications

The quasi-static approximation failure at high frequencies: At high frequencies the quasi-static approximation must break down. Thus at higher frequencies we need to consider other significant physics of the system, known as higher order modes. A further complication is that at higher frequencies, damping becomes an issue.

In acoustics viscosity and thermal effects are typically ignored, by assuming that wave propagation is irrotational, thus is described by the scalar wave equation. It turns out that these two loss mechanisms are related (Rayleigh, 1896). But to understand why is complex. Helmholtz, with help from Kirchhoff, explained this interaction, and independently published them between 1863 (Helmholtz, 1863b) and 1868 (Kirchhoff, 1868). Their collective theory is summarized by Lord
Rayleigh (Rayleigh, 1896), and experimentally verified by Warren P. Mason (Mason, 1928). The nature of the correction is that the wave number is extended to be of form

$$\kappa(s) = \frac{s + \beta_0 \sqrt{s}}{c_0},$$  \hspace{1cm} (5.15)

where the forwarded \(P_-\) and backward \(P_+\) pressure waves propagate as

$$P_{\pm}(s, x) = e^{-\kappa(s)x} e^{-\pi(s)x}$$  \hspace{1cm} (5.16)

with \(\pi(s)\) the complex conjugate of \(\kappa(s)\), and \(\Re \kappa(s) > 0\). The term \(\beta_0 \sqrt{s}\) affects both the real and imaginary parts of \(\kappa(s)\). The real part is a frequency dependent loss and the imaginary part introduces a frequency dependent velocity (Mason, 1928).

The frequency where the loss-less part equals the lossy part is an important parameter of the system. This frequency is

$$s_0 + \beta_0 \sqrt{s_0} = 0, \text{ or } \sqrt{s_0} = \beta_0 \text{ or } f_0 = \beta_0^2 / 2\pi.$$

Assuming air at 23.5°C, \(c_0 = \sqrt{\eta_0 P_0 / \rho_0} \approx 344 \text{ [m/s]}\) is the speed of sound, \(\eta_0 = c_p / c_v = 1.4\) is the ratio of specific heats, \(\mu_0 = 18.5 \times 10^{-6} \text{ [Pa-s]}\) is the viscosity, \(\rho_0 \approx 1.2 \text{ [kgm/m}^2\text{]}\) is the density, \(P_0 = 10^5 \text{ [Pa]}\) (1 atm).\footnote{\text{\textcopyright Add Figure.}}

The constant \(\beta_0 = P \eta' / 2S \sqrt{\rho_0} - 1 / 2 \eta_0 - 1/2)\]

is a thermodynamic constant, \(P\) is the perimeter of the tube and \(S\) the area (Mason, 1928).

For a cylindrical tube having radius \(R = 2S / P\), \(\beta_0 = \eta'_0 / R \sqrt{\rho_0} \). To get a feeling for the magnitude of \(\beta_0\) consider a 7.5 [mm] tube (i.e., the average diameter of the adult ear canal). Then \(\eta'_0 = 6.6180 \times 10^{-3}\) and \(\beta_0 = 1.6110\). Using these conditions the wave-number cutoff frequency is \(1.6110^2 / 2\pi = 0.4131\) [Hz]. At 1 kHz the ratio of the loss over the propagation is \(\beta_0 / \sqrt{|s|} = 1.6011 / \sqrt{2\pi 10^2} \approx 2\%\). At 100 [Hz] this is a 6.4% effect.\footnote{\text{\textcopyright Add WKB derivation (NeelyAllen08-MOH)?}}

Mason shows that the wave speed drops from 344 [m/s] at 2.6 [kHz] to 339 [m/s] at 0.4 [kHz], which is a 1.5% reduction in the wave speed. In terms of the losses, this is much larger effect. At 1 [kHz] the loss is 1 [dB/m] for a 7.5 [mm] tube. Note that the loss and the speed of sound vary inversely with the radius. As the radius approaches the boundary layer thickness (the radial distance such that the loss is \(e^{-1}\)), the effect of loss dominates.

In Section 5.3.1 we shall look at some simple problems where we use the quasi-static effect and derive the Kirchhoff voltage and current equations, starting from Maxwell’s equations.

5.3.2 Lec 43 Summary review of Fund Thms of Mathematics

5.4 Speculations on “all that” (Needs lots of work)

5.4.1 WKB method

This method is use in approximating the solution to the horn equation under the assumption that the reflections are minimal (no strong change in the area function with the range variable. Briefly, if

$$\frac{d^2}{dx^2} \Phi(x, s) = F(x) \Phi(x, s)$$

\footnote{\text{\textcopyright Add WKB derivation (NeelyAllm08-MOH)?}}
where the effective wave number is
\[ F(x) = \frac{2m}{h^2} \left[ V(x) - E \right]. \]

The WKB approximation gives the solution
\[ \Phi(x, s) \approx C_o e^{i \int F(x) dx}. \]

Picking the right branch of the square root is essential in the application of the WKB method, as is unwrapping the phase of each branch of the wave number.

This method is integrating the phase delay of the wave number as a function of the range variable. Thus its complex-analytic properties are essential to obtaining the correct answer (the wave number is not a single valued function of \( x \) and \( s \)).

This method is especially relevant to the case of the Webster Horn equation Eq. 1.127 (p. 140), which has been integrated once by the use of the integration factor method.

For example, integrating Eq. 1.127 (p. 140) over the surface of constant gradient of \( \Phi(x, s) \)

Some careful attention to the details is needed here:
\[ \int \frac{\partial}{\partial x} \left[ A(x) \frac{\partial}{\partial x} \Phi(x, s) \right] dx = \frac{s^2}{c^2} \int A(x) \Phi(x, s) dx \]
\[ \int A(x) \frac{\partial \phi}{\partial x} dx = \frac{s^2}{c^2} \int A(x) \Phi(x, s) dx \]

NOTE: The material from this point on is speculative ramblings of Allen, and should be ignored for now.

### 5.4.2 Inverse problem: Solving for \( A(x) \) given \( z_{\text{rad}}(t) \)

It was shown by Sondhi and Gopinath (1971) [henceforth SG71] that given the horns driving-point impedance \( z_{\text{rad}}(t) \) at the input \( x = x_o \), one may determine the area function \( A(x) \). A portion of this derivation is summarized here.

For each area function \( A(x) \) there is a unique driving point input impedance \( z_{\text{rad}}(t) \leftrightarrow Z_{\text{rad}}(s) \). We define \( z_r(t) \) as the impedance remainder after normalizing by the surge impedance \( z_o \) and then subtracting the initial impulse (delta function)
\[ z_r(t) \equiv z_{\text{rad}}(t)/z_o - \delta(t). \] (5.17)

In the frequency domain \( Z_r(s) = Z_{\text{rad}}(s)/z_o - 1 \). The remainder is related to the reactive portion of \( Z(s) \).

For each remainder there is a corresponding unique \( A(x) \). This relation is
\[ f(a, \xi_o) + \frac{1}{2c} \int_{-a}^{a} z_r(|\xi_o/c - \xi/c|) f(\xi_o, \xi) d\xi = 1. \] (5.18)

Once this integral equation has been solved for \( f(x, x) \), the area function is \( A(x) = f^2(x, x) \) (Sondhi and Gopinath, 1971; Sondhi and Resnick, 1983).
The argument of SG71 begins with the double integration of *Hooke’s Law* (conservation of mass) with respect to time and space. Defining the *per-unit-length compliance* as \( C(x) = A(x)/\eta_o P_o \) (Eq. 1.162), the double integral of Hook’s Law is

\[
\int_{\tau_o}^{\tau} dt \int_{0}^{\xi} dx \ [C(x)p_t(x,t) + \nu_s(x,t)] = 0
\]

\[
\int_{0}^{\xi} dx \ C(x) p(x,t) \bigg|_{\tau_o}^{\tau} + \int_{\tau_o}^{\tau} dt \nu(x,t) \bigg|_{0}^{\xi} = 0
\]

\[
\int_{0}^{\xi} C(x) \left[ p(x,\tau) - p(x,\tau_o) \right] dx + \int_{\tau_o}^{\tau} \nu(\xi,t) - \nu(0,t) \ dt = 0.
\]

Assuming the system starts out from rest, for \( t < 0 \) the quiescent pressure \( p(x,t) \) and volume velocity \( \nu(\xi,t) \) are zero. For \( t = \tau > 0 \) and \( x > c\tau \), the pressure and the outer boundary velocity are also zero [i.e., for \( x > c\tau \), \( p(x,\tau) = \nu(x,\tau) = 0 \)]. This is because there is no disturbance beyond the wave’s characteristics] (SG71, page 1686).

Thus Hooke’s Law reduces to (SG71, Eq. 3)

\[
\int_{\tau_o}^{\tau} \nu(0,t) \ dt = \frac{1}{\eta_o P_o} \int_{0}^{c\tau} A(x)p(x,\tau) \ dx.
\]  

(5.19)

This equation relates the horn volume velocity density, at any location \( x \) (right term), to the input \((x = 0)\) volume velocity, as a function of time (left term).

More specifically, at time \( \tau \) the horn pressure times the shunt compliance \( C(x) \equiv A(x)/\eta_o P_o \) (i.e., the shunt volume velocity density), integrated over the excited region \( 0 \leq x \leq \xi \), is equal to the integral from \( \tau_o \leq t \leq \tau \) of the volume velocity at the input \( x = 0 \).

The idea here is that, due to conservation of mass, at any time \( t = \tau \) there is a relationship between the change in area at the outer rim of the wave at \( \xi = c\tau \), to the change of the input velocity at \( x = 0 \). Because of the finite velocity of the wave, there is a round-trip delay between these two points. It is this delay that makes the relationship much less obvious.

To better understand this relationship, we seek a unique input velocity \( v(t) \) such that \( p(x,\tau) = 1 \). Assuming such a velocity exists,\(^7\)

\[
\int_{\tau_o}^{\tau} v(t) \ dt = \frac{1}{\eta_o P_o} \int_{0}^{c\tau} A(x) \ dx.
\]  

(5.20)

The integral on the right is simply the volume of fluid in the horn at time \( \tau \), between \( 0 \leq x \leq c\tau \). The integral on the right is the total flow into the horn. This relation allows us to find \( A(x) \) given \( v(t) \). The problem is to find \( v(t) \) given \( A(x) \). This is trivial since

\[
v(\tau) = \frac{c}{\eta_o P_o} [A(\tau) - A(0)].
\]  

(5.21)

\( v(0) = 0. \)

\(^7\)This violates causality. The delay of the wave must be accounted for, else the anti-causal input must be used, as in SG71.
5.4.3 Inverse-Solution

We seek \( A(x) \) given the input radiation impedance \( z_{\text{rad}}(t) \) via Eq. 5.19. The derivation of the SG71 results is neither transparent nor self-contained. In the present presentation we hope to rectify this, providing both a complete and intuitive plausible explanation for Eq. ?? (Sondhi and Gopinath, 1971, Eq. 8).

In a few words, here is what is going on: When the input velocity at \( x = 0 \), denoted \( v(t) \) is causally turned on, a wave propagates in the tube at the fixed speed of sound \( c = \sqrt{\eta_0 P_0 / \rho_o} \). There are two waves, a forward and retrograde traveling wave. At the input the current is specified for a zero-pressure boundary condition, while at the wave-front, at \( \tau_o \equiv ct \), there is a moving free boundary, having an impedance boundary condition that must satisfy conservation of mass and momentum. This condition is that the pressure and current are partially reflected due to the change in the impedance (due the change in area). If we knew the pressure at the wave front, we could control the input volume velocity to hold the pressure constant as the wave propagates. But to do this requires knowledge of the future, since there is a delay between the wave front at \( x = ct \).

It is not obvious that there exists an input volume velocity that will keep the pressure constant, and another principle must be called into play to make this feasible.

Let’s try the following exercise: At time 0, a wave enter from the input a distance of \( \delta \equiv cT \) in time \( T \), where it is reflected due to the change in \( A(x) \) and thus the local characteristic admittance \( \mathcal{Y}(x) \). In order that the total pressure remain constant, the forward and retrograde wave velocities must satisfy \( \nu^-(x,t) = \nu^+(x,t) \). For this to occur the retrograde going wave \( \nu^1(x,t) \), reflected from the moving boundary at \( x = ct \) must be equal to the time reversed forward going wave \( \nu^+(x,t) \) launched from the input. If this condition can be made to hold, then the sum of the forward and retrograde waves will be constant. For this to happen we must pre-load the line with an input prior to \( t = 0 \).

We set this condition up by starting with \( v(-t) \) coming from both the left, forward traveling, and from the right, retrograde traveling. Since the boundary is traveling forward, and the retrograde wave is traveling back, and the conditions are determined at the wave-front, the waves are uniquely determined. In the region \( 0 \leq x \leq ct \) the pressure is constant and the velocity is zero. To the right of \( x = ct \) we have a left bound wave, that we wish to specify, and to the left of \( x = 0 \) we have a right bound wave which determines the input admittance.

Thus \( p^+(ct,t) = p^- (ct,t) = 1 \) and \( \nu^-(ct,t) = \nu^+(ct,t) \). Thus just to the left of \( x = ct \), the total pressure is 2 and the total velocity is 0. Based on the boundary condition at \( x = ct \) we have

\[
\nu^-(ct,t) = \nu^+(ct,t) = \mathcal{Y}(ct,t) \equiv \frac{A(ct)}{\rho_0 c}.
\]  

(5.22)

This is sufficient to determine \( \nu^-(ct,t) = \nu^+(ct,t) = A(x)/\rho_0 c \) (since \( p^\pm(ct,t) = 1 \)).

Since the sum of the pressures at the input is also 2, by our constraint (i.e., since the total pressure is 2 everywhere), we may find the input velocity and pressure, and thus the input admittance. This may seem like magic, but as proved by SG71, and other before them (apparently starting with Cauchy), this must work.

Since there can be no wave in front of the wave-front, the moving reflection coefficient must necessarily be 1, as follows from Eq. 5.22

\[
\frac{p^-(ct,t)}{p^+(ct,t)} = \frac{\nu^-(ct,t)}{\nu^+(ct,t)} = 1.
\]  

(5.23)
We may next determine the reflection coefficient at $x = 0$ by delaying the wave front value by the round trip travel time, namely
\[ p^-(0, ct) = p^+(0, 2ct) = 1 \] (5.24)
and
\[ \nu^-(0, ct) = \nu^+(0, 2ct). \] (5.25)

The pressure equation requires that the pressure be symmetric in time ($p^-(t) = pk(-t)$) while the second equation requires that $\nu^-(0, t) = -\nu^+(0, -t)$. Thus the input signals cannot be causal, yet they are related by the causal input admittance, according to

\[ p(0, t) \equiv z_{rad}(t) \star \nu(t) \equiv \int_{-\infty}^{t} z_{rad}(t - \tau)\nu(0, \tau)d\tau. \] (5.26)

Here we assume that $\nu(t) = 0$ for $t < -a$, the pressure is symmetric in time and the velocity is anti-symmetric in time.

Thus the lower limit on the integral $t = -a$ depends on the time when $p(0, \tau)$ is nonzero. If the input velocity is causal (i.e., zero for $t < 0$) then $a = 0$.

At time $t = 0$ the velocity starts out as $(\nu)(0) = \delta(t)$, which is the input initial condition. The input map admittance is therefore entirely determined by the area function.

In practice it is better to work with a step function than with a delta function input. We start with the velocity and pressure zero at $t=0$, and then for positive time, the pressure is $p(0, t) = 2$ while the velocity is $v(t) = 2\nu(0, t)$. At each time step the pressure must be symmetric and the velocity anti-symmetric. The forward and retrograde velocity are related by the variable delay $ct$, and
\[ v^-(t - ct) = v^+(t + ct) = A(ct). \]

At each time step the delay from the wave front is increasing as the boundary moves away. At the wave front, the reflection coefficient is necessarily 1, since the wave may not penetrate beyond the wave front boundary, according to the requirements of the wave equation causality.

Toward this end we recognize that the input pressure $p(x, t)$ (at $x = 0$) may be determined from the input (radiation) impedance $z_{rad}(t)$ by convolving with the input velocity $\nu(t)$.

The $z_{rad}(t)$ is the pressure impulse response, defined as is the inverse Laplace transform of the radiation impedance expressed in the frequency domain. Since it may be measured, or computed given $A(x)$, it is assume here to be known.

Substitution of the above convolution into Eq. 5.19 relates the pressure to the radiation impedance in a linear manner (namely it is similar to a matrix equation.)

What remains is to transform the integral on the right into a convolution, rendering a Fredholm integral equation, which is known to have a well formed solution (Sondhi and Gopinath, 1971).

Solution method In the following $\delta(x)$ is the Dirac delta function and $1(x) = \int_{-\infty}^{x} \delta(x)$ represents the Heaviside unit step function, unity for $x > 0$, zero for $x < 0$ and undefined at $x = 0$.

As the wave front at $x = t/c$ moves through the tube, the change in area causes continuous reflections, which propagate back to the input. It is the accumulation of these continuous reflections, that determines the solution of Eq. 5.3.

As discussed by SG71, we desire the velocity at the input $\nu(0, t)$ such that $P(x, t) = 2$ (see Eq. 5.28 below), then Eq. 5.19 represents the step response of the radiation impedance being equal to the volume of the horn between $0 \leq x \leq ct$.

Given the forward solution to Eq. 5.3 [i.e., $p^+(x, t), \nu^+(x, t)$], it is trivial to show by direct substitution that a time-reversal of the forward solution [i.e., $p^+(x, -t), -\nu^+(x, -t)$] is also a solution (Sondhi and Gopinath, 1971). By superposition of the sum of the forward and time reversed
solutions, i.e.,
\[
\tilde{U}(x, t) = \nu^+(x, t) - \nu^+(x, -t) = 0 \tag{5.27}
\]
and
\[
\tilde{P}(x, t) = p^+(x, t) + p^+(x, -t) = 2. \tag{5.28}
\]
We shall see that such a constructed pressure is constant in the excited region (Sondhi and Gopinath, 1971). In this case \(\tilde{P}(x, 0) = 1\) for \(0 \leq x \leq a\), and is zero for \(x > \xi = c\tau\).

Cauchy problem: Boundary conditions and solution

Outside R:
\[
P(x, t) = U(x, t) = 0
\]
Region R:
\[
U(x, t) = 0, \quad P(x, t) = 2
\]

We seek a solution having a velocity boundary condition \(\nu(0, t) = \delta(t)\), having pressure \(p(0, t) = z_{\text{rad}}(t) \ast \nu(0, t)\) representing the driving-point impedance impulse response.

Given Eq. 30 we may rewrite Eq. 5.28 on the boundary \(x = 0\), as
\[
z_{\text{rad}}(t) \ast \nu(0, t) + z_{\text{rad}}(-t) \ast \nu(0, -t) = 2 \tag{5.29}
\]
thus
\[
\int_{-\infty}^{t} z_{\text{rad}}(t - \tau) \nu(0, \tau) d\tau + \int_{-\infty}^{-t} z_{\text{rad}}(-t - \tau) \nu(0, \tau) d\tau = 2 \tag{5.30}
\]
Since \(z_{\text{rad}}(t) \equiv \delta(t) + h(t)\) we may pull out the surge impedance delta function, to obtain (SG71 Eq. 8)
\[
\nu(a, t) + \frac{1}{2} \int_{-a}^{t} h(t - \tau) \nu(0, \tau) d\tau + \frac{1}{2} \int_{-a}^{-t} h(-t - \tau) \nu(0, \tau) d\tau = 2 \tag{5.31}
\]

5.4.4 WHEN WKB approximation

The final question is what happens when we approximate the true solution with the WKB solution?\(^8\) We know from previous work that when \(|\Gamma_L| = 1\) that the WKB solution fails. Thus it is interesting to consider this case, for the exponential horn, since we know the exact solution, we may do this comparison exactly.

The WKB solution\(^9\) is found by integrating
\[
\frac{dP^+}{dx} = - [\kappa(x, s) + \partial_x \ln \mathcal{Z}'(x, s)] P^+
\]

---

\(^8\)http://en.wikipedia.org/wiki/WKB_approximation

where $\kappa = \mathcal{V} / \mathcal{Y} = sA^2(x)/\eta_o \rho_o$ and $\mathcal{Z}' = sp/A(x)$, thus

$$P^+(x, s) \approx P^+(0, s) e^{\int_0^x [s + \kappa \ln z] dx}$$

Other points that need to be made

1. The primitive waves are one-sided in $x \mp ct$ and therefore must be analytic in $\kappa$. Since $\kappa$ is analytic in $s$, this connects wave-number space to frequency space analytically. This is a major point, and behind splitting waves into primitives. I suspect that what is going on here, is that the two primitives live on different ROCs. For example, $P^+$ must be analytic in on half plane, and $P^-$ is analytic in the other half plane. Thus to compute them, one must close the contour at $\infty$ differently, making them “one-sided” in opposite directions (sides). A complete theory would take advantage of this fact.

2. There must be a simple relationship between the exp horn, and it cutoff region, and Anderson localization. This effect, as I see it, is due to the curvature in the wave front, induced somewhere between the conical and exponential horn’s rate of area change vs. $x$. For example: http://lpmmc.grenoble.cnrs.fr/spip.php?article408

3. I need to work in normalized variables so that $x = 0$ has an area of 1 cm$^2$ and $x = 1$ corresponds to 1 meter down-stream. This would be useful when comparing to Olson’s figure (Fig. 1.35). To do this, I must define $x = (\xi - \xi_o)/L$, $A_o \equiv A(x_o)$ and $A_L = A(1)$. To match Olson, $A_o = 1e - 4$ (1 cm$^2$) and $A_L = 0.01$ (100 cm$^2$).

5.4.5 Rydberg atom model

Fundamental the quantum mechanics is the Rydberg series, which describes the quantized energy levels of atoms

$$\nu_{n,m} = cRZ_n^2 \left( \frac{1}{n^2} - \frac{1}{m^2} \right)$$

where $\nu_{n,m}$ are the possible eigen-frequencies, $c$ is the speed of light, $R \approx 1.097 \times 10^7$ is the Rydberg constant, $Z_n$ is the atomic number, along with positive integers $n$ and $m$ ($m > n$) which represent two principal quantum numbers that label all possible allowed atomic eigen-states. Integer $n$ indicates the lowest (initial) atomic eigen-state while $m$ is the higher state.$^{10}$ When $n = 1$ the series is the Lyman series corresponding to Hydrogen ($Z_1 = 1$).

Given observed frequencies $\nu_{n,m}$ it is possible to find the area function that traps the photons into the Rydberg eigen-states.

One way to think of eigen-modes is to make an analogy to a piano string, or an organ pipe. In these much simpler systems, there is an almost constant delay, say $\tau$ due to a characteristic length, say $L = \tau c$, such that the eigen-modes are given by integer multiples of a half wavelength $\nu_n = nc/2L$. This assumes the endpoint boundary conditions are pinned displacement (i.e., zero velocity). For an organ pipe, closed at one end and open at the other the corresponding formula is multiples of a quarter wavelength $\nu_n = nc/4L$. In each case $\nu = n/\tau$ where $\tau = 2L/c$ is the round trip delay, thus $\nu = nc/2L$. We suggest looking at the Rydberg series in the say way, but with the very different eigen frequencies (Eq. 5.32). Sommerfeld (1949, p. 201) makes a very interesting comment regarding Eq. 5.32:

This equation reduces to a simple mathematical the enigma of the spectral lines, with their finite cummulation point, the behavior of which differs so fundamentally from that of all mechanical systems.

Abstract:

While it is clear that both Schrödinger’s equation and Dirac’s equations are highly accurate, after about 100 years, it is not clear why. Both of these theories seem to violate classical electromagnetics (EM) (i.e., Ohm’s law), since they are built on energy principles rather than electric and magnetic fields. Here we delve into this question, by providing a classical (i.e., EM based) derivation for the Hydrogen atom, one of the most important and obvious successes of quantum mechanics (QM). The problem with QM is not that it fails, rather that it succeeds, without obvious basis. The problem is that one cannot understand the basic principles, and it seems to be in contradiction with any any principles of a physical theory. Based on the Rydberg series, we determine the reflection coefficient, and thus the radiation impedance seen by the electron, in a radial coordinate centered on the proton. Since the electron and proton both have spin 1/2, their magnetic fields must attractively align, accounting for the near field vector potential, and complementing the far field attraction due to their opposite signs. As the electron and proton approach each other, due to their far-field potential attraction, the magnetic near field would further be attractive at close range, due to the magnetic dipoles of the two “particles,” causing them to merge with neutral net magnetic moment and neutral charge, giving a highly stable hydrogen atom. However given a sufficiently strong distorting field, this highly symmetric state could be disturbed, leading to photon radiation, constrained by the radial eigen states.

Introduction

Abstract:

5.4.6 Notes on Jaynes, QED, and quantization

Notes: I am using this document as a place to keep some notes about E.T. Jaynes and his views on QED, along with some other random thoughts. I seem to have misplaced my notebook, and that would be the proper place for this note. Until I find it, this place must do.

1. In Jaynes paper #70, he makes a good case as to why the formulation of quantum mechanics is seriously lacking. As he clearly points out he is hardly alone in his unhappiness. Jaynes’ problem seems to be the rather sloppy way in which the theory was constructed. Rather than go after the math, which I is what I would rather do, he goes after the approach, which he finds misguided (maybe vile would be a better term).

Let me paraphrase Jaynes’ argument as I can best recall it:11 QM and QED is a bit like a crystal. It has a certain structure that would be totally broken if any single part were removed. So he is at a quandary as to which part to attach. He would like to find a chink in the armor, without killing the baby, so to speak. Jaynes recognizes that QED does a lot of things right, and he would like to preserve this success, but remove other aspects that he finds unacceptable.

---

11 I would love to find the exact place, in his writing, other than #70, where he says this, but have been unable to find it, even after an extensive several day search, and email to Larry Bretthorst, host of the Jaynes website http://bayes.wustl.edu/etj/node1.html, who suggests #18.
This is how I recall Jaynes’ unhappiness with QED: Of the things he finds unacceptable, a major one is that, by design QED, does not allow one to directly test it. This, he says, is outrageous, and I agree. One cannot get away with making a theory that is not testable. That is not acceptable within the scientific method. Another problem is the use of probability. A third is that the equation is first order in time, with a complex coefficient $i\hbar\dot{\psi}$.

Here is what I wrote to Larry Bretthorst on July 29, 2013:

Larry,

The basic line of reasoning I recall reading is very close to #70 “Scattering of light by free electrons” (e.g., page 2-3 and beyond). Other papers of interest are #31 “Survey of the Present Status of Neoclassical Radiation Theory” (1973) (e.g., See around page 42 of this MS) and #66 “Clearing up Mysteries - The Original Goal” (1989).

His argument goes something like this: “one cannot be allowed to come up with a question that might provide new insight because the very theory states that your now allowed to measure beyond certain limits.” Item #70 comes close to this line of reasoning on page 36 (2d page of article): "But the Copenhagen interpretation has become something sacred cow, no longer a set of hypotheses to be tested by experiment, but an ideology . . . .

Jont


In "Survey of ...Radiation theory” (1973)” I believe Jaynes says the one place he suggests we attack QED is in its approach to quantization. I don’t recall that he has any specific ideas on this, other than the thought that this might be the best place to attach (question, revise) QED.

I would like to build on Jaynes’ thought. According to exponential horn theory, there is an abrupt cutoff frequency, below which the resistance goes to zero. This effect is analyzed in detail in this document on Horns. I would suggest that we view this cutoff as the same as $E = h\nu$, where the photon is in a trap similar to the exponential horn. Below frequency $\nu$ the photon is in a standing wave trap, with the reflection coefficient being 1 below the cutoff frequency. Once the Reflection coefficient $\Gamma_L = 1$ looking in both directions, the energy becomes quantized at the eigen frequencies of the trap.

Now at the end of last year’s ECE493 lectures, I eventually came to an understanding that when solving an ordinary partial differential equation of first order, the integration factor is identical to the area function $A(x)$ of horn theory. This is a most amazing observation, in my eyes. For example in this document, $A(x)$ is constant for the uniform horn, is $x$ for the conical horn, as shown in Table I. This is also discussed in Table II on page 11, in terms of the factor $F(x)$ which is the coefficient on $P_x$. The method for finding the integration factor is given in Greenberg, but this reconstructs $A(x)$ from $F(x)$, at least for the case of the Horn (Appendix ??, shows that $F(x) = (N - 1)/r$ where $N$ is the number of dimensions.

I question if this is maybe more general than separation of variables. The argument goes something like this: Based on the Divergence Thm, one may convert Conservation of mass
(Eq. 2) into an integral equation. This is now described in Figure 2. This gives a rigorous derivation of the Horn equation starting from Eq. 4.

Here is the next break-through observation: When the area function increases faster than \(x^2\), the conical case, then a cutoff frequency develops. This is therefore strictly a geometrical factor. I have likened this to the trapping of light on a rolling hot road, giving the well known mirage when looking down the road at a grazing angle. When the wave, traveling at the speed of sound (or light) goes around the corner, so there is no direct path, then the cutoff frequency follows. I view this as a form of trapped energy, that provides the evanescent coupling, in terms of a physical description. This explains the abrupt appearance of a cutoff frequency for the exp horn, relative to the conical horn, as shown in Olson’s figure from his page 101 (my Figure 3 on page 6 of this MS).

So the thoughts of Jaynes in #70 (or perhaps $31) and my views seem to vaguely converge, in that this geometrical effect could give rise to a trapping of the waves, thus lead to quantization of the photon below the cutoff frequency. This in essence converts the photon into an electron. Yes, that is a very crazy thought, but then so is QED, as Jaynes points out.

It is incomprehensible to me that one can drop the use of Maxwell’s equations (ME), and the wave equation that follows from ME, and replace the \(\phi_{tt}\) by \(i\hbar\phi_t\). This replaces a hyperbolic equation, supporting waves in two directions, by a parabolic equation with a complex amplitude. To me this makes no sense. A parabolic equation violates Einstein causality since the speed of sound/light is \(\infty\). This follows from the dispersion relationship which would be something like

\[is = \kappa^2.\]  
(5.33)

Here \(s = \sigma + i\omega\), the Laplace frequency, and \(\kappa = \kappa_r + i\kappa_i\) is the complex wave, with \(\kappa_i = 2\pi/\lambda = 1/\hbar\lambda\), assuming traveling plane waves.

2. Here are several quotes from Jaynes’s papers on QED, quantum mechanics (QM) and related topics on waves or Maxwell’s Equations (ME).

- EJ strongly objected to the view commonly held in QM that one cannot ask physical questions. I would need to go back and reread some history of QM to accurately quote the source(s) of this idea. Basically it does not allow one to think in terms of classical explanations. As ETJ points out in several places, the line between the QM or QED explanation and the classical one, is frequently blurred. In retrospect one may find a classical explanation for an effect that was strictly believed to be purely QM. He gives examples that are hard to recall. Perhaps one of the best is the Stark effect which required the introduction of a 1 GHz spin quantization based on the Stern-Gerlac experiment. The use of microwaves proved there was only spin of \(\pm 1/2\), and nothing between. I don’t really have a good handle on this story, but it is a great place to start.

- John Wheeler once told EJ that physics is all about geometry. Of course Einstein said much of the same, and it seems likely that Wheeler was simply quoting Einstein. In any case, I think that this statement is very true, but the really interesting question is, why. Maybe that question now has an answer. More on this below.

- EJ wanted to open the question of weaknesses of QED, and was not sure where or how to start. His proposal was to address the quantization part of the theory. However to the best of my knowledge, he didn’t seem to have a solid proposal as to how to do so.
Following are a few extended comments by me, triggered by EJ’s many views:

- Here I attempt to provide that next step into the Q of QM and QED, with the following observation: Quantization is simply the merging of delay with zero loss wave propagation. The very simplest form of this is standing waves in a delay line. The modes in such a system are the quantized levels and are multiples of half or quarter wave lengths. The boundary conditions are one form of perturbation from perfect mode relations, such as multiples of a quarter wavelength. This is so obvious one wonders why it was not proposed as the physical explanation behind the Q of QM.

- As the geometry becomes more complex, we quickly find dispersive wave propagation, namely the delay becomes frequency dependent. This gives rise to a richer mode structure yet it is still quite lawful. A simple and beautiful example of this follows from the termination of a transmission line with a capacitor (or inductor), as described in my publication with Susan Voss on ear canal impedance and admittance.

- What we have learned from our studies on middle ear (ME) impedance is that typically the only way to really understand how the system works (i.e., how to extract the physics) is to look at the reflectance.

- Now it turns out that the reflectance as defined here seems to not have been defined earlier. This is likely due to its strong similarity with the reflection coefficient, which has a similar formula, but different physical interpretation. Specifically the reflection coefficient is normalized by the source impedance whereas the reflectance is normalized by the surge impedance. Thus the reflectance only depends on the load, whereas the reflection coefficient depends on both the source impedance and the load impedance. One might view the reflectance as self-normalized, since the surge impedance depends only on the driving-point impedance being characterized.

- The mathematics behind the reflectance is defined by the Möbius transformation as nicely explained in Ralph Boas’s wonderful 1986 book *Invitation to complex analysis*. This is a book written to electrical engineers (and their kin).

- Probably the two most important problems to analyze are the electron/photon states and the electronic/photon states of electrons around the nucleus.

- A photon is its own antiparticle, and appears (my hypothesis) as an electron with mass and charge, once put in an evanescent state, as it is (again my hypothesis) about the nucleus (i.e., in the atom).
Appendix A

Notation

A.1 Number systems

The notation used in this book is defined in this appendix so that it may be quickly accessed. Where the definition is sketchy, page numbers are provided where these concepts are fully explained, along with many other important and useful definitions. For example a discussion of \( \mathbb{N} \) may be found on page 29. Math symbols such as \( \mathbb{N} \) may be found at the top of the index, since they are difficult to alphabetize.

A.1.1 Units

Strangely, or not, classical mathematics (as taught today in schools) does not contain the concept of units. It seems units have been abstracted away. This makes mathematics distinct from physics, where almost everything has units. Presumably this makes mathematics more general (i.e., abstract). But for the engineering mind, this is not ideal, as it necessarily means that important physical meaning has been surgically removed, by design. We shall stick to SI units when ever possible. Spatial coordinates are quoted in meters \([m]\), and time in seconds \([s]\). Angles in degrees have no units, whereas radians have units of inverse-seconds \([s^{-1}]\).

A.1.2 Symbols and functions

We use \( \ln \) as the log function base \( e \), \( \log \) as base 2, and \( \pi_k \) to indicate the \( k \)th prime (e.g., \( \pi_1 = 2, \pi_2 = 3 \)).

When working with Fourier \( \mathcal{F}T \) and Laplace \( \mathcal{L}T \) transforms, lower case symbols are in the time domain while upper case indicates the frequency domain, as \( f(t) \leftrightarrow F(\omega) \). An important exception are Maxwell’s equations, because they are so widely used as upper case bold letters (e.g., \( E(x, \omega) \)). It seems logical to change this to conform to lower case, with \( e(x, t) \leftrightarrow E(x, \omega) \) as the preferred notation.

A.1.3 Special symbols common to mathematical:

There are many pre-defined symbols in mathematics, too many to summarize here. We shall only use a small subset, defined here.

\(^1\)https://en.wikipedia.org/wiki/List_of_mathematical_symbols_by_subject#Definition_symbols

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- A set is a collection of objects that have a common property. A set is defined using braces. For example, if set \( P = \{ a, b, c \} \) such that \( a^2 + b^2 = c^2 \), then members of \( P \) obey the Pythagorean theorem. Thus we could say that \( \{ 1, 1, \sqrt{2} \} \in P \).

- Number sets: \( \mathbb{N}, \mathbb{P}, \mathbb{Z}, \mathbb{Q}, \mathbb{F}, \mathbb{I}, \mathbb{R}, \mathbb{C} \) are briefly discussed below, and in more detail in Section 1.3.1 on p. 29.

- One can define sets of sets and subsets of sets, and this is prone (in my experience) to error. For example, what is the difference between the number 0 and the null set \( \emptyset = \{ 0 \} \)? Is \( 0 \in \emptyset \)? Ask a mathematician. It is not a useful construction in the world of engineering.

- They symbol \( \perp \) is used in many different ways to indicate two things are perpendicular, orthogonal, or in disjoint sets. In set theory \( A \perp B \) is equivalent to \( A \cap B = \emptyset \). If two vectors \( E, H \) are perpendicular \( E \perp H \), then their inner product \( E \cdot H = 0 \) is zero. One must infer the meaning of \( \perp \) from its usage (the context).

A.1.4 Greek letters

The Greek letters used in this text include (at least) \( \alpha, \beta, \gamma, \delta, \epsilon, \kappa, \rho, \xi, \omega, \sigma, \phi, \psi, \zeta \), and upper-case \( \Gamma, \Xi, \Phi, \Psi, \Delta, \Omega \). Unfortunately some common Greek letters have no upper-case symbol, such as \( \alpha, \zeta \).

Many of these are pre-associated in engineering and physics with a specific physical meaning. For example, \( \omega [\text{rad}] \) is the radian frequency \( 2\pi f \), \( \rho [\text{kgm/m}^3] \) is commonly the density. \( \phi, \psi \) are commonly used to indicate angles of a triangle, and \( \zeta(s) \) is the Riemann zeta function. Many of these are so well established it makes no sense to define new terms, so we will adopt these common terms (and define them).

Likely you do not know all of these Greek letters, commonly used in mathematics. Some of them are pronounced in strange ways. The symbol \( \xi \) is pronounced “see,” \( \zeta \) is “zeta,” \( \beta \) is “beta,” and \( \chi \) is “kie” (rhymes with pie and sky). I will assume you know how to pronounce the others, which are more phonetic in English. One advantage of learning \LaTeX{} is that all of these math symbols are built in, and thus more easily learned, once you have adopted this powerful open-source math-oriented word-processing system (e.g., used to write this book).

Table A.1: Double-bold notation for the types of numbers. (#) is a page number. Symbol with an exponent denote the dimensionality. Thus \( \mathbb{R}^2 \) represents the real plane. An exponent of 0 denotes point, e.g., \( j \in \mathbb{C}^0 \).

<table>
<thead>
<tr>
<th>Symbol (p. #)</th>
<th>Genus</th>
<th>Examples</th>
<th>Counter Examples</th>
</tr>
</thead>
<tbody>
<tr>
<td>( \mathbb{N} ) (29)</td>
<td>Counting</td>
<td>1, 2, 17, 3, ( 10^{20} )</td>
<td>0, -10, 5j</td>
</tr>
<tr>
<td>( \mathbb{P} ) (29)</td>
<td>Prime</td>
<td>2, 17, 3, ( 10^{20} )</td>
<td>0, 1, 4, 3, 12, -5</td>
</tr>
<tr>
<td>( \mathbb{Z} ) (30)</td>
<td>Integer</td>
<td>-1, 0, 17, 5j, -( 10^{20} )</td>
<td>( 1/2, \pi, \sqrt{5} )</td>
</tr>
<tr>
<td>( \mathbb{Q} ) (30)</td>
<td>Rational</td>
<td>2/1, 3/2, 1.5, 1.14</td>
<td>( \sqrt{2}, 3^{-1/3}, \pi )</td>
</tr>
<tr>
<td>( \mathbb{F} ) (30)</td>
<td>Fractional</td>
<td>1/2, 7/22</td>
<td>2/1, 1/( \sqrt{2} )</td>
</tr>
<tr>
<td>( \mathbb{I} ) (30)</td>
<td>Irrational</td>
<td>( \sqrt{2}, 3^{-1/3}, \pi, e )</td>
<td>Vectors</td>
</tr>
<tr>
<td>( \mathbb{R} ) (31)</td>
<td>Reals</td>
<td>( \sqrt{2}, 3^{-1/3}, \pi )</td>
<td>( 2\pi j )</td>
</tr>
<tr>
<td>( \mathbb{C} ) (31)</td>
<td>Complex</td>
<td>1, ( \sqrt{2}j ), 3( -j/3 ), ( \pi j )</td>
<td>Vectors</td>
</tr>
</tbody>
</table>
A.1. NUMBER SYSTEMS

A.1.5 Double-Bold notation

Table A.1 indicates the symbol followed by a page number indication where it is discussed, and the Genus (class) of the number type. For example, \( \mathbb{N} > 0 \) indicates the infinite set of \textit{counting numbers} \( \{1, 2, 3, \ldots\} \), not including zero. Starting from any counting number, you get the next one by adding 1. Counting numbers are also known as the \textit{Cardinal numbers}.

We say that a number is in the set with the notation \( 3 \in \mathbb{N} \subset \mathbb{R} \), which is read as “3 is in the set of counting numbers, which in turn is in the set of real numbers,” or in vernacular language “3 is a real counting number.”

Prime numbers (\( \mathbb{P} \subset \mathbb{N} \)) are taken from the counting numbers, but do not include 1.

The signed integers \( \mathbb{Z} \) include 0 and negative integers. Rational numbers \( \mathbb{Q} \) are historically defined to include \( \mathbb{Z} \), a somewhat inconvenient definition, since the more interesting class are the \textit{fractionals} \( \mathbb{F} \), a subset of rationals \( \mathbb{F} \subset \mathbb{Q} \) that exclude the integers (i.e., \( \mathbb{F} \perp \mathbb{Z} \)). This is a useful definition because the rationals \( \mathbb{Q} = \mathbb{Z} \cup \mathbb{F} \) are formed from the union of integers and fractionals.

The rationals may be defined, using set notation (a very sloppy language, with incomprehensible syntax) as

\[
\mathbb{Q} = \{p/q : q \neq 0 \& p, q \in \mathbb{Z}\}
\]

which may be read as “the set \{\ldots\} of all \( p/q \) such that \( : q \neq 0 \), ‘and’ \( p, q \subset \mathbb{Z} \).” The translation of the symbols is in single (‘\ldots’) quotes.

Irrational numbers \( \mathbb{I} \) are very special: They are formed by taking a limit of fractionals, as the numerator and denominator \( \to \infty \), and approach a limit point. It follows that irrational numbers must be approximated by fractionals.

The reals \( \mathbb{R} \) include complex numbers \( \mathbb{C} \) having a zero imaginary part (i.e., \( \mathbb{R} \subset \mathbb{C} \)).

The \textit{size} of a set is denoted by taking the absolute value (e.g., \( |\mathbb{N}| \)). Normally in mathematics this symbol indicates the cardinality, so we are defining it differently from the standard notation.

**Classification of numbers:** From the above definitions there exists a natural heretical structure of numbers:

\[
\mathbb{P} \in \mathbb{N}, \quad \mathbb{Z} : \{\mathbb{N}, 0, -\mathbb{N}\}, \quad \mathbb{F} \perp \mathbb{Z}, \quad \mathbb{Q} : \mathbb{Z} \cup \mathbb{F}, \quad \mathbb{R} : \mathbb{Q} \cup \mathbb{I} \subset \mathbb{C}
\]

1. The primes are a subset of the counting numbers: \( \mathbb{P} \subset \mathbb{N} \).

2. The signed integers \( \mathbb{Z} \) are composed of \( \pm \mathbb{N} \) and 0, thus \( \mathbb{N} \subset \mathbb{Z} \).

3. The fractionals \( \mathbb{F} \) do not include the signed integers \( \mathbb{Z} \).

4. The rationals \( \mathbb{Q} = \mathbb{Z} \cup \mathbb{F} \) are the union of the signed integers and fractionals.

5. Irrational numbers \( \mathbb{I} \) have the special properties \( \mathbb{I} \perp \mathbb{Q} \).

6. The reals \( \mathbb{R} : \mathbb{Q}, \mathbb{I} \) are the union of rationals and \textit{irrationals} \( \mathbb{I} \).

7. Reals \( \mathbb{R} \) may be defined as a subset of those complex numbers \( \mathbb{C} \) having zero imaginary part.
A.2 Periodic functions

Fourier series tells us that periodic functions are discrete in frequency, with frequencies given by $nT_s$, where $T_s$ is the sample period ($T_s = 1/F_{\text{max}}$ and $F_{\text{min}} = F_{\text{max}}/N\text{FT}$).

This concept is captured by the Fourier series, which is a frequency expansion of a periodic function. This concept is quite general. Periodic in frequency implies discrete in time. Periodic and discrete in time requires periodic and discrete in frequency (the case of the DFT). The modulo function $\text{mod}_n$ is periodic with period $n$.

A periodic function may be conveniently indicated using double-parentheses notation. This is sometimes known as modular arithmetic. For example $f(((t))) = f(t) = f(t \pm kT)$, is periodic on $t, T \in \mathbb{R}$ with a period of $T$ and $k \in \mathbb{Z}$. This notation is useful when dealing with Fourier series of periodic functions.

When a discrete valued (e.g., time) sequence is periodic we use square brackets $f[[n]] = f[n] = f[n \pm kN]$, with $n, k, N \in \mathbb{Z}$ and period $N$. This notation will be used with discrete-time signals that are periodic, such as the case of the DFT.

A.3 Differential equations vs. Polynomials

A polynomial has degree $N$ defined by the largest power. A quadratic equation is degree 2, and a cubic has degree 3. We shall indicate a polynomial by the notation

$$P_N(z) = z^N + a_{N-1}z^{N-1} \cdots a_0.$$  

It is a good practice to normalize the polynomial so that $a_N = 1$. This will not change the roots, defined by Eq. 1.27 (p. 59). The coefficient on $z^{N-1}$ is always the sum of the roots $z_n (a_{N-1} = \Sigma_n^N z_n)$, and $a_0$ is always their product ($a_0 = \prod_n^N z_n$).

Differential equations have order (polynomials have degree). If a second order differential equation is Laplace transformed (Lec. 1.4.16, p. 101), one is left with a degree 2 polynomial.

Example:

$$\frac{d^2}{dt^2}y(t) + b\frac{d}{dt}y(t) + cy(t) = \alpha \left( \frac{d}{dt}x(t) + \beta x(t) \right) \leftrightarrow (s^2 + bs + c)Y(s) = \alpha(s + \beta)X(s).$$  

Using the same argument as for polynomials, the lead coefficient must always be 1. The coefficient $\alpha \in \mathbb{R}$ is called the gain. The complex variable $s$ is the Laplace frequency. Discuss half-derivatives, etc.

The ratio of the output $Y(s)$ over the input $X(s)$ is called the system transfer function $H(s)$. When $H(s)$ is the ratio of two polynomials in $s$, the transfer function is said to be bilinear, since it is linear in both the input and output. The roots of the numerator are called the zeros and those of the denominator, the poles. The inverse Laplace transform of the transfer function is called the system impulse response, which describes the system’s output signal $y(t)$ for any given input signal $x(t)$, via convolution (i.e., $y(t) = h(t) \star x(t)$).
Appendix B

Matrix algebra: Systems

B.1 Vectors

Vectors as columns of ordered sets of scalars $\in \mathbb{C}$. When we write them out in text, we typically use row notation, with the transpose symbol:

$$[a, b, c]^T = \begin{bmatrix} a \\ b \\ c \end{bmatrix}.$$

This is strictly to save space on the page. The notation for conjugate transpose is $\dagger$, for example

$$\begin{bmatrix} a \\ b \\ c \end{bmatrix}^\dagger = \begin{bmatrix} a^* \\ b^* \\ c^* \end{bmatrix}.$$

The above example is said to be a 3 dimensional vector, because it has three components.

Row vs. column vectors: With rare exceptions, vectors are columns, denoted column-major.\(^1\) To avoid confusion, it is a good rule to make your mental default column-major, in keeping with most signal processing (vectorized) software.\(^2\) Column vectors are the unstated default of Matlab/Octave, only revealed when matrix operations are performed. The need for the column (or row) major is revealed as a consequence of efficiency when accessing long sequences of numbers from computer memory. For example, when forming the sum of many numbers using the Matlab/Octave command \texttt{sum(A)}, where $A$ is a matrix, by default Matlab/Octave operates on the columns, returning a row vector, of column sums. Specifically

$$\text{sum} \begin{bmatrix} 1 & 2 \\ 3 & 4 \end{bmatrix} = [4, 6].$$

If the data were stored in “row-major” order, the answer would have been the column vector $\begin{bmatrix} 3 \\ 7 \end{bmatrix}$.\(^3\)

\(^1\)https://en.wikipedia.org/wiki/Row-_and_column-major_order

\(^2\)In contrast, reading words in English is ‘row-major.’

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**Dot products:** A vector dot product is defined to weight vector elements before summing them, resulting in a scalar. The transpose of a vector (a row-vector), typically used as weights on the elements of a vector. Dot products play an important role in vector algebra and calculus. For example

\[
\begin{bmatrix} 1 \\ 1 \\ 1 \end{bmatrix} \cdot \begin{bmatrix} 1 \\ 2 \\ 3 \end{bmatrix} = \begin{bmatrix} 1^T \end{bmatrix} \begin{bmatrix} 1 \\ 2 \\ 3 \end{bmatrix} = \begin{bmatrix} 1 \\ 1 \\ 1 \end{bmatrix} \begin{bmatrix} 2 \\ 3 \end{bmatrix} = 1 + 2 + 3 = 6.
\]

**Dialects of vector notation:** Physical fields are, by definition, functions of space \(x\) [m], and in the most general case, time \(t\) [s]. When Laplace transformed, the fields become functions of space and complex frequency (e.g., \(E(x, t) \leftrightarrow E(x, s)\)). As before, there are several equivalent vector notations. For example, vector \(E(x, t)\) may also written as

\[
E(x, t) = \begin{bmatrix} E_x(x, t) \\ E_y(x, t) \\ E_z(x, t) \end{bmatrix} = \begin{bmatrix} E_x(x) \\ E_y(x) \\ E_z(x) \end{bmatrix}(x, t) = \begin{bmatrix} E_x, \ E_y, \ E_z \end{bmatrix}^T \equiv E_x \hat{x} + E_y \hat{y} + E_z \hat{z}.
\]

The above equation, with an equation number, is called a “displayed” equation. The in-line dialect is \(E(x, t) = [E_x, E_y, E_z]^T E_x(x, t) \hat{x} + E_y(x, t) \hat{y} + E_z(x, t) \hat{z}\). These are all shorthand notations for expressing the vector. Such usage is similar to a dialect in a language. Another dialect is column-transpose or “in-line” notation, so to place the vector on one line, saving space: \(E(x, t) = [E_x, E_y, E_z](x, t)^T\). These several variants on vector notation are frequently used, even interchangeably.

**Complex elements:** When the elements are complex \((\in \mathbb{C})\), the transpose is defined as the complex conjugate of the elements. In such complex cases the transpose conjugate is denoted with a \(†\) rather than \(T\). Vectors are also frequency written using a bold font

\[
a^T = \begin{bmatrix} 2j \\ -3j \\ 1 \end{bmatrix} \in \mathbb{C}.
\]

For this case when the elements are complex, the dot product is a real number

\[
a \cdot b = a^\dagger b = \begin{bmatrix} a_1^* \\ a_2^* \\ a_3^* \end{bmatrix} \begin{bmatrix} b_1 \\ b_2 \\ b_3 \end{bmatrix} = a_1^* b_1 + a_2^* b_2 + a_3^* b_3 \in \mathbb{R}.
\]

**Norm of a vector:** The dot product of a vector with itself is called the norm of \(a\), designated as

\[
\|a\| = \sqrt{a^\dagger a} \geq 0
\]

which is always non-negative.

Such a construction is useful when \(a\) and \(b\) are related by an impedance matrix

\[
V(s) = Z(s) I(s)
\]

and we wish to compute the power. For example, the impedance of a mass is \(ms\) and a capacitor is \(1/sC\). When given a system of equations (a mechanical or electrical circuit) one may define an impedance matrix.
Complex power: In this special case, the complex power $P(s) \in \mathbb{R}(s)$ is defined, in the complex frequency domain $(s)$ as

$$P(s) = I^\dagger(s)V(s) = I^\dagger(s)Z(s)I(s) \leftrightarrow p(t).$$

The real part of the complex power must be positive. The imaginary part corresponds to available stored energy.

GIVE MORE EXAMPLES

The case of three-dimensions is special, allowing definitions that are not defined in more dimensions. A vector in $\mathbb{R}^3$ labels the point having the coordinates of that vector.

B.1.1 Vectors in $\mathbb{R}^3$

Dot product in $\mathbb{R}^3$: The dot $B \cdot C = ||B|| ||C|| \cos(\theta)$, and $\cos(\theta)$ is called the direction cosine between $B$ and $C$.

Norm of a vector in $\mathbb{R}^3$: The norm of a vector is the dot product of it with itself $||A|| = \sqrt{A \cdot A}$.

Euclidean distance between two points in $\mathbb{R}^3$: The dot product of the difference between two vectors $(A - B) \cdot (A - B)$ is the Euclidean distance between the points they define $||A - B|| = \sqrt{(a_1 - b_1)^2 + (a_2 - b_2)^2 + (a_3 - b_3)^2}$.

Cross product: and cross product $A \times B = ||A|| ||B|| \sin(\theta)$ are defined between the two vectors $A$ and $B$.

The triple product: This is defined between three vectors as $A \cdot (B \times C) = \det \begin{vmatrix} a_1 & a_2 & a_3 \\ b_1 & b_2 & b_3 \\ c_1 & c_2 & c_3 \end{vmatrix}$ also defined in Fig. 1.16. This may be indicated without the use of parentheses, since there can be no other meaningful interpretation. However for rigor, parentheses should be used. The triple product is the volume of the parallelepiped (3D-crystal shape) outlined by the three vectors, shown in Fig. 1.16, p. 82.

B.2 NxM Matrices

When working with matrices, the role of the weights and vectors can change, depending on the context. A useful way to view a matrix is as a set of column vectors, weighted by the elements of the column-vector of weights multiplied from the right. For example

$$\begin{bmatrix} a_{11} & a_{12} & a_{13} & \cdots & a_{1M} \\ a_{21} & a_{22} & a_{32} & \cdots & a_{3M} \\ \vdots & & & & \vdots \\ a_{N1} & a_{N2} & a_{N3} & \cdots & a_{NM} \end{bmatrix} \begin{bmatrix} w_1 \\ w_2 \\ \cdots \\ w_M \end{bmatrix} = \begin{bmatrix} a_{11} & a_{12} & a_{13} & \cdots & a_{1M} \\ a_{21} & a_{22} & a_{32} & \cdots & a_{3M} \\ \vdots & & & & \vdots \\ a_{N1} & a_{N2} & a_{N3} & \cdots & a_{NM} \end{bmatrix} \begin{bmatrix} w_1 \\ w_2 \\ \cdots \\ w_M \end{bmatrix} + \begin{bmatrix} a_{11} \\ a_{21} \\ \vdots \\ a_{N1} \end{bmatrix}.$$
where the weights are \( [w_1, w_2, \ldots, w_M]^T \). Alternatively the matrix is a set of row vectors of weights, each of which are applied to the column vector on the right \(([w_1, w_2, \ldots, W_M]^T)\).

The determinant of a matrix is denoted either as \( \det A \) or simply \( |A| \), (as in the absolute value.) The inverse of a square matrix is \( A^{-1} \) or \( \text{inv} A \). If \( |A| = 0 \), the inverse does not exist.

\[ AA^{-1} = A^{-1}A. \]

Matlab’s notional convention for a row-vector is \([a, b, c]\) and a column-vector is \([a; b; c]\). A prime on a vector takes the complex conjugate transpose. To suppress the conjugation, place a period before the prime. The : argument converts the array into a column vector, without conjugation. A tacit notation in Matlab is that vectors are columns and the index to a vector is a row vector. Matlab defines the notation \( 1:4 \) as the “row-vector” \([1, 2, 3, 4]\), which is unfortunate as it leads users to assume that the default vector is a row. This can lead to serious confusion later, as Matlab’s default vector is a column. I have not found the above convention explicitly stated, and it took me years to figure this out for myself.

When writing a complex number we shall adopt \( 1\mathbf{j} \) to indicate \( \sqrt{-1} \). Matlab/Octave prefer this as well, as its explicit.

Units are SI; Angles in degrees \([\text{deg}]\) unless otherwise noted. The units for \( \pi \) are always in radians \([\text{rad}]\). Ex: \( \sin(\pi) \), \( e^{\mathbf{j}90^\circ} e^{\mathbf{j}\pi/2} \).

B.2.1 NxN matrices and 2x2 systems

Definitions:

1. **Scalar**: A number, e.g. \( \{a, b, c, \alpha, \beta, \cdots\} \in \{Z, Q, I, R, C\} \)

2. **Vector**: A quantity having direction as well as magnitude, often denoted by a bold-face letter with an arrow, \( \mathbf{x} \). In matrix notation, this is typically represented as a single row \([x_1, x_2, x_3, \ldots] \) or single column \([x_1, x_2, x_3 \ldots]^T \) (where \( T \) indicates the transpose). In this class we will typically use column vectors. The vector may also be written out using unit vector notation to indicate direction. For example: \( \mathbf{x}_{3 \times 1} = x_1 \hat{x} + x_2 \hat{y} + x_3 \hat{z} = [x_1, x_2, x_3]^T \), where \( \hat{x}, \hat{y}, \hat{z} \) are unit vectors in the \( x, y, z \) Cartesian directions (here the vector’s subscript \( 3 \times 1 \) indicates its dimensions). The type of notation used may depend on the engineering problem you are solving.

3. **Matrix**: \( A = [a_1, a_2, a_3, \cdots, a_M]_{N \times M} = \{a_{n,m}\}_{N \times M} \), can be a non-square matrix if the number of elements in each of the vectors \( (N) \) is not equal to the number of vectors \( (M) \). When \( M = N \), the matrix is square. It may be inverted if its determinant \( |A| = \prod \lambda_k \neq 0 \) (where \( \lambda_k \) are the eigenvalues).

We shall only work with \( 2 \times 2 \) and \( 3 \times 3 \) square matrices throughout this course.

4. **Linear system of equations**: \( Ax = b \) where \( x \) and \( b \) are vectors and matrix \( A \) is a square.

(a) **Inverse**: The solution of this system of equations may be found by finding the inverse \( x = A^{-1}b \)

(b) **Equivalence**: If two systems of equations \( A_0 \mathbf{x} = \mathbf{b}_0 \) and \( A_1 \mathbf{x} = \mathbf{b}_1 \) have the same solution (i.e., \( \mathbf{x} = A_0^{-1}b_0 = A_1^{-1}b_1 \)), they are said to be equivalent.
(c) **Augmented matrix**: The first type of augmented matrix is defined by combining the matrix with the right-hand-side. For example, given the linear system of equations \( Ax = y \)

\[
\begin{bmatrix}
  a & b \\
  c & d \\
\end{bmatrix}
\begin{bmatrix}
  x_1 \\
  x_2 \\
\end{bmatrix}
= 
\begin{bmatrix}
  y_1 \\
  y_2 \\
\end{bmatrix},
\]

then the augmented matrix is

\[
A|y = \begin{bmatrix}
  a & b \\
  c & d \\
\end{bmatrix}
\begin{bmatrix}
  y_1 \\
  y_2 \\
\end{bmatrix}
\]

A second type of augmented matrix may be used for finding the inverse of a matrix (rather than solving a specific instance of linear equations \( Ax = b \)). In this case the augmented matrix is

\[
A|I = \begin{bmatrix}
  a & b & 1 & 0 \\
  c & d & 0 & 1 \\
\end{bmatrix}
\]

Performing Gaussian elimination on this matrix, until the left side becomes the identity matrix, yields \( A^{-1} \). This is because multiplying both sides by \( A^{-1} \) gives \( A^{-1}A|A^{-1}I = I|A^{-1} \).

5. **Permutation matrix** (\( P \)): A matrix that is equivalent to the identity matrix, but with scrambled rows (or columns). Such a matrix has the properties \( \det(P) = \pm 1 \) and \( P^2 = I \). For the 2x2 case, there is only one permutation matrix:

\[
P = \begin{bmatrix}
  0 & 1 \\
  1 & 0 \\
\end{bmatrix} \quad P^2 = \begin{bmatrix}
  0 & 1 \\
  1 & 0 \\
\end{bmatrix} \begin{bmatrix}
  0 & 1 \\
  1 & 0 \\
\end{bmatrix} = \begin{bmatrix}
  1 & 0 \\
  0 & 1 \\
\end{bmatrix}
\]

A permutation matrix \( P \) swaps rows or columns of the matrix it operates on. For example, in the 2x2 case, pre-multiplication swaps the rows

\[
PA = \begin{bmatrix}
  0 & 1 \\
  1 & 0 \\
\end{bmatrix}
\begin{bmatrix}
  a & b \\
  \alpha & \beta \\
\end{bmatrix}
= \begin{bmatrix}
  \alpha & \beta \\
  a & b \\
\end{bmatrix},
\]

whereas post-multiplication swaps the columns

\[
AP = \begin{bmatrix}
  a & b \\
  \alpha & \beta \\
\end{bmatrix}
\begin{bmatrix}
  0 & 1 \\
  1 & 0 \\
\end{bmatrix}
= \begin{bmatrix}
  b & a \\
  \beta & \alpha \\
\end{bmatrix}
\]

For the 3x3 case there are \( 3 \cdot 2 = 6 \) such matrices, including the original 3x3 identity matrix (swap a row with the other 2, then swap the remaining two rows).

6. **Gaussian elimination (GE) operations** \( G_k \): There are 3 types of elementary row operations, which may be performed without fundamentally altering a system of equations (e.g. the resulting system of equations is equivalent). These operations are (1) swap rows (e.g. using a permutation matrix), (2) scale rows, or (3) perform addition/subtraction of two scaled rows. All such operations can be performed using matrices.

For lack of a better term, we’ll describe these as ‘Gaussian elimination’ or ‘GE’ matrices.\(^3\) We will categorize any matrix that performs only elementary row operations (but any number of them) as a ‘GE’ matrix. Therefore, cascade of GE matrices is also a GE matrix.

---

\(^3\)The term ‘elementary matrix’ may also be used to refer to a matrix that performs an elementary row operation. Typically, each elementary matrix differs from the identity matrix by one single row operation. A cascade of elementary matrices could be used to perform Gaussian elimination.
Consider the GE matrix

\[ G = \begin{bmatrix} 1 & 0 \\ 1 & -1 \end{bmatrix} \]

(a) This pre-multiplication scales and subtracts row (1) from (2) and returns it to row (2).

\[ GA = \begin{bmatrix} 1 & 0 \\ 1 & -1 \end{bmatrix} \begin{bmatrix} a \\ b \\ \alpha \\ \beta \end{bmatrix} = \begin{bmatrix} a \\ a - \alpha \\ b - \beta \end{bmatrix} \]

The shorthand for this operation is Gaussian elimination operation is \((1) \leftarrow (1)\) and \((2) \leftarrow (1) - (2)\).

(b) Post-multiplication adds and scales columns.

\[ AG = \begin{bmatrix} a \\ \alpha \\ b \\ \beta \end{bmatrix} \begin{bmatrix} 1 & 0 \\ -1 & 1 \end{bmatrix} = \begin{bmatrix} a - b \\ \alpha - \beta \\ b \end{bmatrix} \]

Here the second column is subtracted from the first, and placed in the first. The second column is untouched. This operation is not a Gaussian elimination. Therefore, to put Gaussian elimination operations in matrix form, we form a cascade of pre-multiply matrices.

Here \( \det(G) = 1 \), \( G^2 = I \), which won’t always be true if we scale by a number greater than 1. For instance, if \( G = \begin{bmatrix} 1 \\ m \end{bmatrix} \) (scale and add), then we have \( \det(G) = 1 \),

\[ G^n = \begin{bmatrix} 1 \\ n \cdot m \\ 1 \end{bmatrix} \].

**Exercise:** Find the solution to the following 3x3 matrix equation \( Ax = b \) by Gaussian elimination. Show your intermediate steps. You can check your work at each step using Matlab.

\[
\begin{bmatrix}
1 & 1 & -1 \\
3 & 1 & 1 \\
1 & -1 & 4
\end{bmatrix}
\begin{bmatrix}
x_1 \\
x_2 \\
x_3
\end{bmatrix} =
\begin{bmatrix}
1 \\
9 \\
8
\end{bmatrix}.
\]

1. Show (i.e., verify) that the first GE matrix \( G_1 \), which zeros out all entries in the first column, is given by

\[ G_1 = \begin{bmatrix} 1 & 0 & 0 \\ -3 & 1 & 0 \\ -1 & 0 & 1 \end{bmatrix} \]

Identify the elementary row operations that this matrix performs. **Sol:** Operate with GE matrix on \( A \)

\[ G_1[A|b] = \begin{bmatrix}
1 & 0 & 0 \\
-3 & 1 & 0 \\
-1 & 0 & 1
\end{bmatrix}
\begin{bmatrix}
1 & 1 & -1 & 1 \\
3 & 1 & 1 & 9 \\
1 & -1 & 4 & 8
\end{bmatrix} =
\begin{bmatrix}
1 & 1 & -1 & 1 \\
0 & -2 & 4 & 6 \\
0 & -2 & 5 & 7
\end{bmatrix} \]

The second row of \( G_1 \) scales the first row by -3 and adds it to the second row

\[(2) \leftarrow -3(1) + (2).\]

The third row of \( G1 \) scales the first row by -1 and adds it to the third row \([(3) \leftarrow -(1) + (3)]\).
2. Find a second GE matrix, $G_2$, to put $G_1A$ in upper triangular form. Identify the elementary row operations that this matrix performs. Sol:

$$G_2 = \begin{bmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & -1 & 1 \end{bmatrix}$$

or $[(2) \iff -(2) + (3)]$. Thus we have

$$G_2G_1[A|b] = \begin{bmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & -1 & 1 \end{bmatrix} \begin{bmatrix} 1 & 0 & 0 \\ -3 & 1 & 0 \\ -1 & 0 & 1 \end{bmatrix} \begin{bmatrix} 1 & 1 & -1 \\ 3 & 1 & 1 \\ 1 & -1 & 4 \end{bmatrix} = \begin{bmatrix} 1 & 1 & -1 \\ 0 & -2 & 4 \\ 0 & 0 & 1 \end{bmatrix}$$

3. Find a third GE matrix, $G_3$, which scales each row so that its leading term is 1. Identify the elementary row operations that this matrix performs. Sol:

$$G_3 = \begin{bmatrix} 1 & 0 & 0 \\ 0 & -1/2 & 0 \\ 0 & 0 & 1 \end{bmatrix}$$

which scales the second row by $-1/2$. Thus we have

$$G_3G_2G_1[A|b] = \begin{bmatrix} 1 & 0 & 0 \\ 0 & -1/2 & 0 \\ 0 & 0 & 1 \end{bmatrix} \begin{bmatrix} 1 & 1 & -1 \\ 3 & 1 & 1 \\ 1 & -1 & 4 \end{bmatrix} = \begin{bmatrix} 1 & 1 & -1 \\ 0 & 1 & -2 \\ 0 & 0 & 1 \end{bmatrix}$$

4. Finally, find the last GE matrix, $G_4$, that subtracts a scaled version of row 3 from row 2, and scaled versions of rows 2 and 3 from row 1, such that you are left with the identity matrix ($G_4G_3G_2G_1A = I$). Sol:

$$G_4 = \begin{bmatrix} 1 & -1 & -1 \\ 0 & 1 & 2 \\ 0 & 0 & 1 \end{bmatrix}$$

Thus we have

$$G_4G_3G_2G_1[A|b] = \begin{bmatrix} 1 & -1 & -1 \\ 0 & 1 & 2 \\ 0 & 0 & 1 \end{bmatrix} \begin{bmatrix} 1 & 1 & -1 \\ 3 & 1 & 1 \\ 0 & 0 & 1 \end{bmatrix} = \begin{bmatrix} 1 & 0 & 0 \\ 0 & 1 & 0 \\ 0 & 0 & 1 \end{bmatrix}$$

5. Solve for $[x_1, x_2, x_3]^T$ using the augmented matrix format $G_4G_3G_2G_1[A|b]$ (where $[A|b]$ is the augmented matrix). Note that if you’ve performed the preceding steps correctly, $x = G_4G_3G_2G_1b$. Sol: From the preceding problems, we see that $[x_1, x_2, x_3]^T = [3, -1, 1]^T$. 
B.3 Inverse of the 2x2 matrix

We shall now apply Gaussian elimination to find the solution \([x_1, x_2]\) for the 2x2 matrix equation \(Ax = y\) (Eq. 1.61, left). We assume to know \([a, b, c, d]\) and \([y_1, y_2]\). We wish to show that the intersection (solution) is given by the equation on the right.

Here we wish to prove that the left equation \((i)\) has an inverse given by the right equation \((ii)\):

\[
\begin{bmatrix}
a & b \\
c & d \\
\end{bmatrix}
\begin{bmatrix}
x_1 \\
x_2 \\
\end{bmatrix}
= 
\begin{bmatrix}
y_1 \\
y_2 \\
\end{bmatrix} \quad \text{(i)};
\begin{bmatrix}
x_1 \\
x_2 \\
\end{bmatrix}
= 
\frac{1}{\Delta}
\begin{bmatrix}
d & -b \\
-c & a \\
\end{bmatrix}
\begin{bmatrix}
y_1 \\
y_2 \\
\end{bmatrix} \quad \text{(ii)}.
\]

How to take inverse:
1) Swap the diagonal, 2) change the signs of the off-diagonal, and 3) divide by \(\Delta\).

B.3.1 Derivation of the inverse of a 2x2 matrix

1. Step 1: To derive \((ii)\) starting from \((i)\), normalize the first column to 1.

\[
\begin{bmatrix}
1 & \frac{b}{a} \\
1 & \frac{c}{a} \\
\end{bmatrix}
\begin{bmatrix}
x_1 \\
x_2 \\
\end{bmatrix}
= 
\begin{bmatrix}
\frac{1}{a} & 0 \\
0 & \frac{1}{c} \\
\end{bmatrix}
\begin{bmatrix}
y_1 \\
y_2 \\
\end{bmatrix}
\]

2. Step 2: Subtract row (1) from row (2): \((2) \leftarrow (2)-(1)\)

\[
\begin{bmatrix}
1 & \frac{b}{a} \\
0 & \frac{d}{a} - \frac{b}{a} \\
\end{bmatrix}
\begin{bmatrix}
x_1 \\
x_2 \\
\end{bmatrix}
= 
\begin{bmatrix}
\frac{1}{a} & 0 \\
-\frac{1}{a} & \frac{1}{c} \\
\end{bmatrix}
\begin{bmatrix}
y_1 \\
y_2 \\
\end{bmatrix}
\]

3. Step 3: Multiply row (2) by \(ca\) and express result in terms of the determinate \(\Delta = ad - bc\).

\[
\begin{bmatrix}
1 & \frac{b}{a} \\
0 & \frac{a}{\Delta} \\
\end{bmatrix}
\begin{bmatrix}
x_1 \\
x_2 \\
\end{bmatrix}
= 
\begin{bmatrix}
\frac{1}{a} & 0 \\
-\frac{1}{a} & \frac{1}{c} \\
\end{bmatrix}
\begin{bmatrix}
y_1 \\
y_2 \\
\end{bmatrix}
\]

4. Step 4: Solve row (2) for \(x_2\): \(x_2 = -\frac{c}{a}y_1 + \frac{a}{\Delta}y_2\).

5. Step 5: Solve row (1) for \(x_1\):

\[
x_1 = \frac{1}{a}y_1 - \frac{b}{a}x_2
\]

\[
= \frac{1}{a}y_1 - \frac{b}{a}\left[-\frac{c}{a}y_1 + \frac{a}{\Delta}y_2\right]
\]

\[
= \left[1 + \frac{bc}{a\Delta}\right]y_1 - \frac{b}{a}y_2.
\]

Rewriting in matrix format, in terms of \(\Delta = ad - bc\), gives:

\[
\begin{bmatrix}
x_1 \\
x_2 \\
\end{bmatrix}
= 
\frac{1}{a + \frac{bc}{\Delta}}
\begin{bmatrix}
y_1 \\
y_2 \\
\end{bmatrix}
= 
\frac{1}{\Delta}
\begin{bmatrix}
\frac{\Delta + bc}{a} & -b \\
-c & a \\
\end{bmatrix}
\begin{bmatrix}
y_1 \\
y_2 \\
\end{bmatrix}
= 
\frac{1}{\Delta}
\begin{bmatrix}
d & -b \\
-c & a \\
\end{bmatrix}
\begin{bmatrix}
y_1 \\
y_2 \\
\end{bmatrix},
\]

since \(d = (\Delta + bc)/a\).

Summary: This is a lot of messy algebra, that is why it is essential you memorize it:
1) Swap diagonal, 2) change off-diagonal signs, 3) normalize by \(\Delta\).
Appendix C

Eigen Analysis

In the following discussion we show how to determine $E$ and $D$ (i.e., $\Lambda$), given $A$.

Calculating the eigenvalue matrix ($\Lambda$): The matrix equation for $E$ is

$$AE = E\Lambda.$$  \hspace{1cm} (C.1)

Pre-multiplying by $E^{-1}$ diagonalizes $A$, given the eigenvalue matrix ($D$ in Matlab)

$$\Lambda = E^{-1}AE.$$  \hspace{1cm} (C.2)

Post-multiplying by $E^{-1}$ recovers $A$

$$A = E\Lambda E^{-1}.$$  \hspace{1cm} (C.3)

Matrix power formula: This last relation is the entire point of the eigenvector analysis, since it shows that any power of $A$ may be computed from powers of the eigen values. Specifically

$$A^n = E\Lambda^n E^{-1}.$$  \hspace{1cm} (C.4)

For example, $A^2 = AA = E\Lambda (E^{-1}E) \Lambda E^{-1} = E\Lambda^2 E^{-1}$.

Equations C.1, C.2 and C.3 are the key to eigenvector analysis, and you need to memorize them. You will use them repeatedly throughout this course, and possibly for a long time after it is over.

Showing that $A - \lambda_\pm I_2$ is singular: If we restrict Eq. C.1 to a single eigenvector (one of $e_\pm$), along with the corresponding eigenvalue $\lambda_\pm$, we obtain a matrix equations

$$Ae_\pm = E_\pm \lambda_\pm = \lambda_\pm E_\pm.$$  \hspace{1cm} (C.5)

Note the important swap in the order of $E_\pm$ and $\lambda_\pm$. Since $\lambda_\pm$ is a scalar, this is legal (and critically important), since this allows us to remove (factored out) $E_\pm$

$$(A - \lambda_\pm I_2)E_\pm = 0.$$  \hspace{1cm} (C.5)

This means that the matrix $A - \lambda_\pm I_2$ must be singular, since when it operates on $E_\pm$, which is not zero, it gives zero. It immediately follows that its determinant is zero (i.e., $|(A - \lambda_\pm I_2)| = 0$). This equation is used to uniquely determine the eigenvalues $\lambda_\pm$. Note the important difference between $\lambda_\pm I_2$ and $\Lambda$ (i.e., $|(A - \Lambda)| \neq 0$).
Calculating the eigenvalues $\lambda_{\pm}$: The eigenvalues $\lambda_{\pm}$ of $A$ may be determined from $| (A - \lambda_{\pm} I_2) | = 0$

\[
\begin{vmatrix}
1 - \lambda_{\pm} & N \\
1 & 1 - \lambda_{\pm}
\end{vmatrix} = (1 - \lambda_{\pm})^2 - N^2 = 0.
\]

For our case of $N = 2$, $\lambda_{\pm} = (1 \pm \sqrt{2})$.

Calculating the eigenvectors $e_{\pm}$: Once the eigenvalues have been determined, they are substitute them into Eq. C.5, which determines the eigenvectors $E = [e_+, e_-]$, by solving

\[
(A - \lambda_{\pm}) e_{\pm} = \begin{bmatrix} 1 - \lambda_{\pm} & 2 \\ 1 & 1 - \lambda_{\pm} \end{bmatrix} E_{\pm} = 0,
\]

where $1 - \lambda_{\pm} = 1 - (1 \pm \sqrt{2}) = \mp \sqrt{2}$.

Recall that Eq. C.5 is singular, because we are using an eigenvalue, and each eigenvector is pointing in a unique direction (This is why it is singular). You might respectively suggest that this equation has no solution. In some sense you would be correct. When we solve for $E_{\pm}$, the two equations defined by Eq. C.5 co-linear (the two equations describe parallel lines). This follows from the fact that there is only one eigenvector for each eigenvalue.

Expecting trouble, yet proceeding to solve for $E = [e_1^+, e_2^+]^T$,

\[
\begin{bmatrix}
-\sqrt{2} & 2 \\
1 & -\sqrt{2}
\end{bmatrix}
\begin{bmatrix}
e_1^+ \\
e_2^+
\end{bmatrix} = 0.
\]

This gives two identical equations $-\sqrt{2} e_1^+ + 2 e_2^+ = 0$ and $e_1^+ - \sqrt{2} e_2^+ = 0$. This is the price of an over-specified equation (the singular matrix is degenerate). The most we can determine is $e_+ = c [ -\sqrt{2}, 1 ]^T$, where $c$ is a constant. We can determine eigenvector direction, but not its magnitude.

Following exactly the same procedure for $\lambda_-$, the equation for $e_-$ is

\[
\begin{bmatrix}
\sqrt{2} & 2 \\
1 & \sqrt{2}
\end{bmatrix}
\begin{bmatrix}
e_1^- \\
e_2^-
\end{bmatrix} = 0.
\]

In this case the relation becomes $e_1^- + \sqrt{2} e_2^- = 0$, thus $E_- = c [ \sqrt{2}, 1 ]^T$ where $c$ is a constant.

Normalization of the eigenvectors: The constant $c$ may be determined by normalizing the eigenvectors to have unit length. Since we cannot determine the length, we set it to 1. In some sense the degeneracy is resolved by this normalization. Thus $c = 1/\sqrt{3}$, since

\[
(\pm \sqrt{2})^2 + 1^2 = 3 = 1/c^2.
\]

Summary: Thus far we have shown

\[
E = [e_+, e_-] = \frac{1}{\sqrt{3}} \begin{bmatrix} \sqrt{2} & -\sqrt{2} \\ 1 & 1 \end{bmatrix}
\]

and

\[
\Lambda = \begin{bmatrix} \lambda_+ & 0 \\ 0 & \lambda_- \end{bmatrix} = \begin{bmatrix} 1 + \sqrt{2} & 0 \\ 0 & 1 - \sqrt{2} \end{bmatrix}.
\]

\[1^{st} It is a convention to order the eigenvalues from largest to smallest.
Verify that $\Lambda = E^{-1}AE$: To find the inverse of $E$, 1) swap the diagonal values, 2) change the sign of the off diagonals, and 3) divide by the determinant $\Delta = 2\sqrt{2}/\sqrt{3}$ (see Appendix B)

$$E^{-1} = \frac{\sqrt{3}}{2\sqrt{2}} \begin{bmatrix} 1 & \sqrt{2} \\ -1 & \sqrt{2} \end{bmatrix} = \begin{bmatrix} 0.6124 & 0.866 \\ -0.6124 & 0.866 \end{bmatrix}.$$  

By definition for any matrix $E^{-1}E = EE^{-1} = I_2$. Taking the product gives

$$E^{-1}E = \frac{\sqrt{3}}{2\sqrt{2}} \begin{bmatrix} 1 & \sqrt{2} \\ -1 & \sqrt{2} \end{bmatrix} \cdot \frac{1}{\sqrt{3}} \begin{bmatrix} \sqrt{2} & -\sqrt{2} \\ 1 & 1 \end{bmatrix} = \begin{bmatrix} 1 & 0 \\ 0 & 1 \end{bmatrix} = I_2.$$  

We wish to show that $\Lambda = E^{-1}AE$

$$\begin{bmatrix} 1 + \sqrt{2} & 0 \\ 0 & 1 - \sqrt{2} \end{bmatrix} = \frac{\sqrt{3}}{2\sqrt{2}} \begin{bmatrix} 1 & \sqrt{2} \\ -1 & \sqrt{2} \end{bmatrix} \cdot \frac{1}{\sqrt{3}} \begin{bmatrix} \sqrt{2} & -\sqrt{2} \\ 1 & 1 \end{bmatrix} \cdot \frac{1}{\sqrt{3}} \begin{bmatrix} 1 & 1 \\ 1 & 1 \end{bmatrix},$$

which is best verified with Matlab.

Verify that $A = E\Lambda E^{-1}$: We wish to show that

$$\begin{bmatrix} 1 & 2 \\ 1 & 1 \end{bmatrix} = \frac{1}{\sqrt{3}} \begin{bmatrix} \sqrt{2} & -\sqrt{2} \\ 1 & 1 \end{bmatrix} \cdot \frac{1 + \sqrt{2}}{0} \begin{bmatrix} 0 & 1 - \sqrt{2} \end{bmatrix} \cdot \frac{\sqrt{3}}{2\sqrt{2}} \begin{bmatrix} 1 & \sqrt{2} \\ -1 & \sqrt{2} \end{bmatrix},$$

which is best verified with Matlab (or Octave). All the above equations have been verified both with Matlab and Octave.
Appendix D

Symbolic analysis of $T E = E \Lambda$

Here we derive the eigen-matrix $E$, and eigen-value matrix $\Lambda$ given a 2x2 Transmission matrix

$$T = \begin{bmatrix} A & B \\ C & D \end{bmatrix},$$

such that $T E = E \Lambda$, using symbolic algebra methods, given by the Matlab/Octave’s script

```matlab
syms A B C D T E L %Use symbolic Matlab/Octave
T=[A B;C D] %Given matrix T
[E,L]=eig(T) %Find eigen-vector matrix E and %eigen-value matrix L
```

D.1 General case

The eigenvectors $e_{\pm}$ are

$$e_{\pm} = \left( \frac{1}{ \pi} \left[ (A - D) \mp \sqrt{(A - D)^2 + 4BC} \right] \right)$$

(D.1)

and eigenvalues are

$$\lambda_{\pm} = \frac{1}{2} \left( \frac{(A + D) - \sqrt{(A - D)^2 + 4BC}}{(A + D) + \sqrt{(A - D)^2 + 4BC}} \right)$$

(D.2)

The term under the radical (i.e., the discriminant) may be written in terms of the determinant of $T$

$$(A - D)^2 + 4BC = A^2 + D^2 - 4(AD - BC) = A^2 + D^2 - 4\Delta_T.$$

This becomes especially important for the case of reciprocal systems where $\Delta_T = 1$, or for anti-reciprocal systems where $\Delta_T = -1$.

D.2 Special cases having symmetry

Each 2x2 matrix has four entries, each of which can be complex. This leads to $4 \times 4 = 8$ possible special symmetries (an eightfold way), discussed next, in quasi-order of their importance. Each
APPENDIX D. SYMBOLIC ANALYSIS OF $TE = E\Lambda$

symmetry is related to properties of $T$. For example if $T = T^\dagger$, the matrix is said to have Hermitian symmetry. When $T = T^\ddagger$ the matrix is symmetric, and when $T = -T^T$ is said to be skew-symmetric. Each of these eightfold symmetries corresponds to some sort of physical constraint, as discussed below.

**D.2.1 Reversible systems**

When the values of $T$ on its diagonal are equal ($A = D$), the matrix is called reversible, and the eigenvectors and eigenvalues greatly simplify to

$$E = \begin{bmatrix} -\sqrt{\frac{B}{C}} & +\sqrt{\frac{B}{C}} \\ 1 & 1 \end{bmatrix}, \quad \Lambda = \begin{bmatrix} A - \sqrt{BC} & 0 \\ 0 & A + \sqrt{BC} \end{bmatrix} \quad (D.3)$$

This is a common symmetry, thus it is very important. It is useful in describing transmission lines, as been discussed in length in Sections 1.4.11 (p. 92), 3.3.2 and 3.3.3 (pp 208-210).

**D.2.2 Reciprocal systems**

When the matrix is symmetric ($B = C$), the corresponding system is said to be reciprocal. Most physical systems are reciprocal. The determinant of the transmission matrix of a reciprocal network $\Delta_T = AD - BC = 1$. For example Electrical networks, composed of inductors, capacitors and resistors are always reciprocal. It follows that the complex impedance matrix is symmetric (Van Valkenburg, 1964a).

Magnetic systems, such as dynamic loudspeakers are anti-reciprocal, and correspondingly $\Delta_T = -1$ The impedance matrix of a loudspeaker is skew symmetric (Kim and Allen, 2013). All impedance matrices are either symmetric or anti-symmetric, depending on if they are reciprocal (LRC networks) or anti-reciprocal (magnetic networks). These system have complex eigen values with negative real parts, corresponding to lossy systems. In some sense, all of this follows from conservation of energy, but the precise general case is waiting for enlightenment. The impedance matrix is never Hermitian. It is easily proved that Hermitian matrices have real eigen-values, which correspond to lossless networks. Any physical system of equations that has any type of loss, cannot be Hermitian.

In summary, given a reciprocal system, the $T$ matrix has $\Delta_T = 1$, and the corresponding impedance matrix is symmetric (*not* Hermitian).

**D.2.3 Impedance**

As previously discussed in Section 1.4.11 (p. 92), the $T$ matrix corresponding to an impedance matrix is

$$\begin{bmatrix} V_1 \\ V_2 \end{bmatrix} = Z(s) \begin{bmatrix} I_1 \\ I_2 \end{bmatrix} = \frac{1}{C} \begin{bmatrix} A & \Delta_T \\ 1 & D \end{bmatrix} \begin{bmatrix} I_1 \\ I_2 \end{bmatrix}.$$

Reciprocal systems have skew-symmetric impedance matrices, namely $z_{12} = -z_{21}$. When the system is both reversible $A = D$ and reciprocal, the impedance matrix simplifies to

$$Z(s) = \frac{1}{C} \begin{bmatrix} A & 1 \\ 1 & A \end{bmatrix}$$
For such systems there are only two degrees of freedom, $A$ and $C$. As discussed previously Section 1.4.11, p. 92, these each have a physical meaning, $1/C$ is the Thévenin source voltage and $A/Cc$ is the Thévenin impedance.

D.2.4 Transmission matrices and symmetry

The transmission matrix fully characterizes a two-port network (Sect. 1.4.11, p. 92).

D.2.5 Hermitian symmetry

When a system is Hermitian its matrix is conjugate symmetric

$$Z(s) = Z^\dagger(s),$$

a stronger condition than reciprocal, but not the symmetric symmetry of a Brune impedance matrix. Impedance is not Hermitian, unless the diagonal elements are real, but it does have symmetric symmetry.

In the case of a Hermitian matrix, the eigenvalues are always real. To show this start from the definition of an impedance eigen-equation ($V$ is a vector of voltages, $I$ a current vector $Z$ and an impedance matrix)

$$V = ZI = I\Lambda,$$

where $Z, I, V, \Lambda \in \mathbb{C}$, $A = A^\dagger$ is a square conjugate-symmetric matrix, and $I, V$ are vectors of the size of $Z$. Here $Z^\dagger$ is the complex transpose (see Appendix A, p. 256). The power $P$ is the real part of the voltage times the current

$$2P = V^\dagger I + VI^\dagger = (ZI)^\dagger I + ZII^\dagger = I^\dagger ZI + ZII^\dagger$$

??? Subtracting the two equations gives

D.2.6 Double roots

For the 2x2 case of double roots the matrix has Jordan form

$$T = \begin{bmatrix} \lambda & 1 \\ 0 & \lambda \end{bmatrix}.$$ 

Then

$$T^2 = \begin{bmatrix} \lambda^n & n\lambda \\ 0 & \lambda^n \end{bmatrix}.$$ 

This generalizes to $n \times n$ matrices having arbitrary combinations of degeneracies (multiple roots), as in symmetric (square) drums, for example,
APPENDIX D. SYMBOLIC ANALYSIS OF $TE = EA$
Appendix E

Analysis of Pell equation (N=2, 3, M)

Section 2.2.3 (p. 193) showed that the solution \([x_n, y_n]^T\) to Pell’s equation, for \(N = 2\), is given by powers of Eq. 1.10. To find an explicit formula for \([x_n, y_n]^T\), one must compute powers of

\[
A = \begin{bmatrix} 1 & 2 \\ 1 & 1 \end{bmatrix}.
\]

We wish to find the solution to Pell’s equation (Eq. 1.10), based on the recursive solution, Eq. 1.11 (p. 50). Thus we need is powers of \(A\), that is \(A^n\), which gives the a closed form expression for \([x_n, y_n]^T\). By the diagonalization of \(A\), its powers are simply the powers of its eigenvalues. This diagonalization is called an eigenvalue analysis, a very general method rooted in linear algebra. This type of analysis allows us to find the solution to most of the linear the equations we encounter.

From Matlab with \(N = 2\) the eigenvalues of Eq. E.1 are \(\lambda_{\pm} \approx [2.4142, -0.4142j]\) (i.e., \(\lambda_{\pm} = 1j(1 \pm \sqrt{2})\)). The final solution to Eq. E.1 is given in Eq. 2.11 (p. 193). The solution for \(N = 3\) is provided in Appendix E.1.1 (p. 272).

Once the matrix has been diagonalized, one may compute powers of that matrix as powers of the eigenvalues. This results in the general solution given by

\[
\begin{bmatrix} x_n \\ y_n \end{bmatrix} = 1j^n \begin{bmatrix} 1 \\ 0 \end{bmatrix} = 1j^n E \Lambda^n E^{-1} \begin{bmatrix} 1 \\ 0 \end{bmatrix}.
\]

The eigenvalue matrix \(D\) is diagonal with the eigenvalues sorted, largest first. The Matlab command \([E, D] = \text{eig}(A)\) is helpful to find \(D\) and \(E\) given any \(A\). As we saw above,

\[
\Lambda = 1j \begin{bmatrix} 1 + \sqrt{2} & 0 \\ 0 & 1 - \sqrt{2} \end{bmatrix} \approx \begin{bmatrix} 2.414j & 0 \\ 0 & -0.414j \end{bmatrix}.
\]

E.1 Pell equation eigenvalue-eigenvector analysis

Here we show how to compute the eigenvalues and eigenvectors for the 2x2 Pell matrix for \(N = 2\)

\[
A = \begin{bmatrix} 1 & 2 \\ 1 & 1 \end{bmatrix}.
\]

The Matlab command \([E, D] = \text{eig}(A)\) returns the eigenvector matrix \(E\)

\[
E = [e_+, e_-] = \frac{1}{\sqrt{3}} \begin{bmatrix} \sqrt{2} \\ 1 \end{bmatrix}, \begin{bmatrix} -\sqrt{2} \\ 1 \end{bmatrix} = \begin{bmatrix} 0.8165 & -0.8165 \\ 0.5774 & 0.5774 \end{bmatrix}.
\]

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and the eigenvalue matrix $\Lambda$ (Matlab’s D)

$$\Lambda \equiv \begin{bmatrix} \lambda_+ & 0 \\ 0 & \lambda_- \end{bmatrix} = \begin{bmatrix} 1 + \sqrt{2} & 0 \\ 0 & 1 - \sqrt{2} \end{bmatrix} = \begin{bmatrix} 2.4142 & 0 \\ 0 & -0.4142 \end{bmatrix}.$$

The factor $\sqrt{3}$ on $E$ normalizes each eigenvector to 1 (i.e., The Matlab command $\text{norm}([\sqrt{2}, 1])$ gives $\sqrt{3}$).

In the following discussion we show how to determine $E$ and $D$ (i.e., $\Lambda$), given $A$.

Table E.1: This table summarizes the solution of Pell’s equation due to the Pythagoreans using matrix recursion, for the case of $N=3$. The integer solutions are shown in on the right. Note that $x_n/y_n \to \sqrt{3}$, in agreement with the Euclidean algorithm. The Matlab program for generating this data is $\text{PellSol3.m}$. It seem likely that $\beta_0$ could be absorbed in the starting solution, and then be removed from the generating function, other than as the known factor $\beta_0^n$.

### Pell’s Equation for $N = 3$

Case of $N = 3$ & $[x_0, y_0]^T = [1, 0]^T$, $\beta_0 = \sqrt[4]{2}$

Note: $x_n^2 - 3y_n^2 = 1$, $x_n/y_n \to \sqrt{3}$

\[
\begin{bmatrix} x_1 \\ y_1 \end{bmatrix} = \beta_0 \begin{bmatrix} 1 \\ 1 \end{bmatrix} = \beta_0 \begin{bmatrix} 1 & 3 \\ 1 & 1 \end{bmatrix} \begin{bmatrix} 1 \\ 0 \end{bmatrix} \quad (1\beta_0)^2 - 3(1\beta_0)^2 = 1
\]
\[
\begin{bmatrix} x_2 \\ y_2 \end{bmatrix} = \beta_0^2 \begin{bmatrix} 4 \\ 2 \end{bmatrix} = \beta_0^2 \begin{bmatrix} 1 & 3 \\ 1 & 1 \end{bmatrix} \begin{bmatrix} 1 \\ 1 \end{bmatrix} \quad (4\beta_0^2)^2 - 3(2\beta_0^2)^2 = 1
\]
\[
\begin{bmatrix} x_3 \\ y_3 \end{bmatrix} = \beta_0^3 \begin{bmatrix} 10 \\ 6 \end{bmatrix} = \beta_0^3 \begin{bmatrix} 1 & 3 \\ 1 & 1 \end{bmatrix} \begin{bmatrix} 4 \\ 2 \end{bmatrix} \quad (10\beta_0^3)^2 - 3(6\beta_0^3)^2 = 1
\]
\[
\begin{bmatrix} x_4 \\ y_4 \end{bmatrix} = \beta_0^4 \begin{bmatrix} 28 \\ 16 \end{bmatrix} = \beta_0^4 \begin{bmatrix} 1 & 3 \\ 1 & 1 \end{bmatrix} \begin{bmatrix} 10 \\ 6 \end{bmatrix} \quad (28\beta_0^4)^2 - 3(16\beta_0^4)^2 = 1
\]
\[
\begin{bmatrix} x_5 \\ y_5 \end{bmatrix} = \beta_0^5 \begin{bmatrix} 76 \\ 44 \end{bmatrix} = \beta_0^5 \begin{bmatrix} 1 & 3 \\ 1 & 1 \end{bmatrix} \begin{bmatrix} 28 \\ 16 \end{bmatrix} \quad (76\beta_0^5)^2 - 3(44\beta_0^5)^2 = 1
\]

### E.1.1 Pell equation for $N = 3$

In Fig. E.1 for $N = 3$ is given, with $\beta_0 = \sqrt[4]{2}$. Perhaps try other trivial solutions such as $[-1, 0]^T$ and $[\pm \sqrt{}; 0]^T$, to provide clues to the proper value of $\beta_0$ for cases where $N > 3$.

**Exercise:** I suggest that you verify $EA \neq AE$ and $AE = EA$ with Matlab. Here is the Matlab program which does this:

\[
A = \begin{bmatrix} 1 & 2 \\ 1 & 1 \end{bmatrix}; \text{ %define the matrix}
\]
\[
[E,D] = \text{eig}(A); \text{ %compute the eigenvector and eigenvalue matrices}
\]
\[
A*E-E*D \%This should be $\approx 0$, within numerical error.
\]
\[
E*D-D*E \%This is not zero
\]

\[1\text{My student Kehan found the genaral formula for } \beta_0.\]
Appendix F

Tables of Fourier and Laplace Transforms

Properties of Fourier Transforms:
1. Both time \( t \) and frequency \( \omega \) are real.
2. For the forward transform (time to frequency), the sign of the exponential is negative.
3. The limits on the integrals in both the forward and reverse FTs are \([-\infty, \infty]\).
4. When taking the inverse Fourier transform, the scale factor of \( 1/2\pi \) is required to cancel the \( 2\pi \) in the differential \( d\omega \).
5. The Fourier step function may be defined by the use of superposition of 1 and \( \text{sgn}(t) = t/|t| \) as
   \[
   \tilde{u}(t) \equiv \frac{1 + \text{sgn}(t)}{2} = \begin{cases} 
   1 & \text{if } t > 0 \\
   1/2 & \text{if } t = 0 \\
   0 & \text{if } t < 0 
   \end{cases}.
   \]
   Taking the FT of a delayed step function
   \[
   \tilde{u}(t - T_o) \leftrightarrow \frac{1}{2} \int_{-\infty}^{\infty} [1 - \text{sgn}(t - T_o)] e^{-j\omega t} dt = \pi \delta(\omega) + \frac{e^{-j\omega T_o}}{j\omega}
   \]
   Thus the FT of the step function has the term \( \pi \delta(\omega) \) due to the 1 in the definition of the Fourier step. This term introduces a serious flaw with the FT of the step function: While it appears to be causal, it is not.
6. The convolution \( \tilde{u}(t) * \tilde{u}(t) \) is not defined because both \( 1 * 1 \) and \( \tilde{\delta}^2(\omega) \) do not exist (and cannot be defined).
7. The inverse FT has convergence problems whenever there is a discontinuity in the time response. This we indicate with a hat over the reconstructed time response. The error between the target time function and the reconstructed is zero in the root-mean sense, but not point-wise.
   Specifically, at the discontinuity point for the Fourier step function \( (t = 0) \), \( \tilde{u}(t) \neq u(t) \), yet \( \int |\tilde{u}(t) - u(t)|^2 dt = 0 \). At the point of the discontinuity the reconstructed function displays Gibbs ringing (it oscillates around the step, hence does not converge at the jump).\(^1\) The \( \mathcal{LT} \) does not exhibit Gibbs ringing.

\(^1https://en.wikipedia.org/wiki/Gibbs_phenomenon\)
8. The FT is not always analytic in $\omega$, as in this example of the step function. The step function cannot be expanded in a Taylor series about $\omega = 0$, because $\delta'(\omega)$ is not analytic in $\omega$.

9. The Fourier $\delta$ function is denoted $\tilde{\delta}(t)$, to differentiate it from the Laplace delta function $\delta(t)$. They differ because the step functions differ, due to the convergence problem described above.

10. One may define

$$\tilde{u}(t) = \int_{-\infty}^{t} \tilde{\delta}(t) dt,$$

and define the somewhat questionable notation

$$\tilde{\delta}(t) = \frac{d}{dt} \tilde{u}(t),$$

since the Fourier step function is not analytic.

11. The rec($t$) function is defined as

$$\text{rec}(t) = \frac{\tilde{u}(t) - \tilde{u}(t - T_o)}{T_o} = \begin{cases} 0 & \text{if } t > 0 \\ 1/T_o & 0 < t < T_o \\ 0 & \text{if } t < 0 \end{cases}.$$  

It follows that $\tilde{\delta}(t) = \lim_{T_o \to 0} \text{rec}(t)$. Like $\tilde{\delta}(t)$, the rec($t$) has unit area.

---

Table F.1: Summary of key properties of FTs.

<table>
<thead>
<tr>
<th>FT Properties</th>
</tr>
</thead>
<tbody>
<tr>
<td>$\frac{d}{dt} v(t)$ $\leftrightarrow$ $j\omega V(\omega)$</td>
</tr>
<tr>
<td>$f(t) * g(t)$ $\leftrightarrow$ $F(\omega)G(\omega)$</td>
</tr>
<tr>
<td>$f(t)g(t)$ $\leftrightarrow$ $\frac{1}{2\pi}$ $F(\omega) * G(\omega)$</td>
</tr>
<tr>
<td>$f(at)$ $\leftrightarrow$ $\frac{1}{a} F\left(\frac{\omega}{a}\right)$</td>
</tr>
</tbody>
</table>
Table F.2: A brief table of simple Fourier Transforms. Note \( a > 0 \in \mathbb{R} \) has units [rad/s]. To flag this necessary condition, we use \(|a|\) to assure this condition will be met. The other constant \( T_o \in \mathbb{R} [s] \) has no restrictions, other than being real. Complex constants may not appear as the argument to a delta function, since complex numbers do not have the order property.

<table>
<thead>
<tr>
<th>( f(t) ) ↔ ( F(\omega) )</th>
<th>Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>( \tilde{\delta}(t) ) ↔ ( 1(\omega) \equiv 1 \forall \omega )</td>
<td>Dirac</td>
</tr>
<tr>
<td>( 1(t) \equiv 1 \forall t \leftrightarrow 2\pi \tilde{\delta}(\omega) )</td>
<td>Dirac</td>
</tr>
<tr>
<td>( \text{sgn}(t) = \frac{t}{</td>
<td>t</td>
</tr>
<tr>
<td>( \tilde{u}(t) = \frac{1(t) + \text{sgn}(t)}{2} \leftrightarrow \pi \tilde{\delta}(\omega) + \frac{1}{j\omega} \equiv \tilde{U}(\omega) )</td>
<td>step</td>
</tr>
<tr>
<td>( \tilde{\delta}(t - T_o) \leftrightarrow e^{-j\omega T_o} )</td>
<td>delay</td>
</tr>
<tr>
<td>( \tilde{\delta}(t - T_o) \star f(t) \leftrightarrow F(\omega)e^{-j\omega T_o} )</td>
<td>delay</td>
</tr>
<tr>
<td>( \tilde{u}(t)e^{-</td>
<td>a</td>
</tr>
<tr>
<td>( \text{rec}(t) = \frac{1}{T_o} [\tilde{u}(t) - \tilde{u}(t - T_o)] \leftrightarrow \frac{1}{T_o} \left(1 - e^{-j\omega T_o}\right) )</td>
<td>pulse</td>
</tr>
<tr>
<td>( \tilde{u}(t) \star \tilde{u}(t) \leftrightarrow \delta^2(\omega) )</td>
<td>Not defined</td>
</tr>
<tr>
<td>( \tilde{u}(t) \star \tilde{u}(t) )</td>
<td>NaN</td>
</tr>
</tbody>
</table>
Properties of the Laplace Transform

1. Time \( t \in \mathbb{R} \) [s] and \( \text{Laplace frequency} \) [rad] are defined as \( s = \sigma + \omega j \in \mathbb{C} \).

2. Given a \( \text{Laplace transform} \) (\( LT \)) pair \( f(t) \leftrightarrow F(s) \), in the engineering literature, the \textit{time domain} is always lower case \([f(t)]\) and causal (i.e., \( f(t < 0) = 0 \)) and the \textit{frequency domain} is upper-case [e.g. \( F(s) \)]. Maxwell’s venerable equations are the unfortunate exception to this otherwise universal rule.

3. The target time function \( f(t < 0) = 0 \) (i.e., it must be causal). The time limits are \( 0^- < t < \infty \). Thus the integral must start from slightly below \( t = 0 \) to integrate over a delta functions at \( t = 0 \). For example if \( f(t) = \delta(t) \), the integral must include both sides of the impulse. If you wish to include non-causal functions such as \( \delta(t + 1) \) it is necessary to extend the lower time limit. In such cases simply set the lower limit of the integral to \(-\infty\), and let the integrand \( (f(t)) \) determine the limits.

4. When taking the forward transform \( (t \rightarrow s) \), the sign of the exponential is negative. This is necessary to assure that the integral converges when the integrand \( f(t) \rightarrow \infty \) as \( t \rightarrow \infty \). For example, if \( f(t) = e^t u(t) \) (i.e., without the negative \( \sigma \) exponent), the integral does not converge.

5. The limits on the integrals of the forward transform are \( t : (0^-, \infty) \in \mathbb{R} \), and the reverse LTs are \([\sigma_o - \infty j, \sigma_o + \infty j] \in \mathbb{C} \). These limits will be further discussed in Section 1.5.9 (p. 125).

6. When taking the inverse Laplace transform, the normalization factor of \( 1/2\pi j \) is required to cancel the \( 2\pi j \) in the differential \( ds \) of the integral.

7. The frequency for the LT must be is complex, and in general \( F(s) \) is complex analytic for \( \sigma > \sigma_o \). It follows that the real and imaginary parts of \( F(s) \) are related. Given \( \Re\{F(s)\} \) it is possible to find \( \Im\{F(s)\} \) (Boas, 1987). More on this in Section 1.4.16 (p. 101).

8. To take the inverse Laplace transform, we must learn how to integrate in the complex \( s \) plane. This will be explained in Sections 1.5.5-1.5.9 (p. 121-125).

9. The Laplace step function is defined as

\[
    u(t) = \int_{-\infty}^{t} \delta(t) dt = \begin{cases} 
    1 & \text{if } t > 0 \\
    \text{NaN} & t = 0 \\
    0 & \text{if } t < 0
    \end{cases}.
\]

Alternatively one could define \( \delta(t) = du(t)/dt \).

10. It is easily shown that \( u(t) \leftrightarrow 1/s \) by direct integration

\[
    F(s) = \int_{0}^{\infty} u(t) e^{-st} dt = -\frac{e^{-st}}{s}\bigg|_{t=0}^{\infty} = \frac{1}{s}.
\]

With the LT step \((u(t))\) there is no Gibbs ringing effect.
11. In many physical applications, the Laplace transform takes the form of a ratio of two polynomials. In such case the roots of the numerator polynomial are called the zeros while the roots of the denominator polynomial are called the poles. For example the LT of \( u(t) \leftrightarrow 1/s \) has a pole at \( s = 0 \), which represents integration, since

\[
u(t) * f(t) = \int_{-\infty}^{\infty} f(\tau)d\tau \leftrightarrow \frac{F(s)}{s}.
\]

12. The LT is quite different from the FT in terms of its analytic properties. For example, the step function \( u(t) \leftrightarrow 1/s \) is complex analytic everywhere, except at \( s = 0 \). The FT of \( 1 \leftrightarrow 2\pi \tilde{\delta}(\omega) \) is not analytic anywhere.

13. Dilated step function \((a \in \mathbb{R})\)

\[
u(at) \leftrightarrow \int_{-\infty}^{\infty} u(at)e^{-st}dt = \frac{1}{a} \int_{-\infty}^{\infty} u(\tau)e^{-(s/a)\tau}d\tau = \frac{a}{|a|}\frac{1}{s} = \pm\frac{1}{s},
\]

where we have made the change of variables \( \tau = at \). The only effect that \( a \) has on \( u(at) \) is the sign of \( t \), since \( u(t) = u(2t) \). However \( u(-t) \neq u(t) \), since \( u(t) \cdot u(-t) = 0 \), and \( u(t) + u(-t) = 1 \), except at \( t = 0 \), where it is not defined.

Once complex integration in the complex plane has been defined (Section 1.5.2, p. 110), we can justify the definition of the inverse LT (Eq. 1.79).
Table F.3: The following table provides a brief table of Laplace Transforms of $f(t), \delta(t), u(t), \text{rect}(t), T_o, p, e, \in \mathbb{R}$ and $G(s), s, a \in \mathbb{C}$. Given a Laplace transform (LT) pair $f(t) \leftrightarrow F(s)$, the frequency domain will always be upper-case [e.g. $F(s)$] and the time domain lower case [$f(t)$] and causal (i.e., $f(t < 0) = 0$). An extended table of transforms is given in Table F.4 on page 279.

<table>
<thead>
<tr>
<th>$f(t)$ ↔ $F(s)$</th>
<th>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</th>
<th>Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>$\delta(t)$ ↔ 1</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>Dirac</td>
</tr>
<tr>
<td>$\delta(at) \leftrightarrow \frac{1}{</td>
<td>a</td>
<td>}$</td>
</tr>
<tr>
<td>$\delta(t - T_o)$ ↔ $e^{-sT_o}$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>delayed Dirac</td>
</tr>
<tr>
<td>$\delta(t - T_o) \ast f(t) \leftrightarrow F(s)e^{-sT_o}$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>–</td>
</tr>
<tr>
<td>$\sum_{n=0}^{\infty} \delta(t - nT_o) = \frac{1}{1 - \delta(t - T_o)} \leftrightarrow \frac{1}{1 - e^{-sT_o}} = \sum_{n=0}^{\infty} e^{-snT_o}$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>one-sided impulse train</td>
</tr>
<tr>
<td>$u(t) \leftrightarrow \frac{1}{s}$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>step</td>
</tr>
<tr>
<td>$u(-t) \leftrightarrow -\frac{1}{s}$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>anti-causal step</td>
</tr>
<tr>
<td>$u(at) \leftrightarrow \frac{1}{a} \frac{1}{s}$</td>
<td>$a \neq 0 \in \mathbb{R}$</td>
<td>dilated or reversed step</td>
</tr>
<tr>
<td>$e^{at}u(t) \leftrightarrow \frac{1}{s + a}$</td>
<td>$a &gt; 0 \in \mathbb{R}$</td>
<td>damped step</td>
</tr>
<tr>
<td>$e^{\pm at}u(t) \leftrightarrow \frac{1}{s \pm a}$, $a \in \mathbb{C}$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>modulated step</td>
</tr>
<tr>
<td>$\cos(at)u(t) \leftrightarrow \frac{1}{2} \left( \frac{1}{s - a} + \frac{1}{s + a} \right)$</td>
<td>$a \in \mathbb{R}$</td>
<td>cos</td>
</tr>
<tr>
<td>$\sin(at)u(t) \leftrightarrow \frac{1}{2j} \left( \frac{1}{s - a} - \frac{1}{s + a} \right)$</td>
<td>$a \in \mathbb{C}$</td>
<td>“damped” sin</td>
</tr>
<tr>
<td>$u(t - T_o) \leftrightarrow \frac{1}{s} e^{-sT_o}$</td>
<td>$T_o &gt; 0 \in \mathbb{R}$</td>
<td>time delay</td>
</tr>
<tr>
<td>$\text{rect}(t) = \frac{1}{T_o} [u(t) - u(t - T_o)] \leftrightarrow \frac{1}{T_o} \left( 1 - e^{-sT_o} \right)$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>rect-pulse</td>
</tr>
<tr>
<td>$u(t) \ast u(t) = tu(t) \leftrightarrow 1/s^2$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>ramp</td>
</tr>
<tr>
<td>$u(t) \ast u(t) \ast u(t) = \frac{1}{2} t^2 u(t) \leftrightarrow 1/s^3$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>double ramp</td>
</tr>
<tr>
<td>$\frac{1}{\sqrt{t}} u(t) \leftrightarrow \sqrt{\frac{s}{\pi}}$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td></td>
</tr>
<tr>
<td>$t^p u(t) \leftrightarrow \frac{\Gamma(p + 1)}{s^{p+1}}$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td>$\Re p &gt; -1, q \in \mathbb{C}$</td>
</tr>
<tr>
<td>$J_n(\omega_o t) \leftrightarrow \frac{\left( \sqrt{s^2 + \omega_o^2} - s \right)^n}{\omega_o \sqrt{s^2 + \omega_o^2}}$</td>
<td>$t \in \mathbb{R}; s, F(s) \in \mathbb{C}$</td>
<td></td>
</tr>
</tbody>
</table>


Table F.4: Functional relationships between Laplace Transforms.

<table>
<thead>
<tr>
<th>LT functional properties</th>
</tr>
</thead>
<tbody>
<tr>
<td>$f(t) \ast g(t) = \int_{t=0}^{t} f(t-\tau)g(\tau)d\tau \leftrightarrow F(s)G(s)$</td>
</tr>
<tr>
<td>convolution</td>
</tr>
<tr>
<td>$u(t) \ast f(t) = \int_{0}^{t} f(t)dt \leftrightarrow \frac{F(s)}{s}$</td>
</tr>
<tr>
<td>convolution</td>
</tr>
<tr>
<td>$f(at)u(at) \leftrightarrow \frac{1}{a}F\left(\frac{s}{a}\right)$ $a \in \mathbb{R} \neq 0$</td>
</tr>
<tr>
<td>scaling</td>
</tr>
<tr>
<td>$f(t)e^{-at}u(t) \leftrightarrow F(s+a)$</td>
</tr>
<tr>
<td>damped</td>
</tr>
<tr>
<td>$f(t-T)e^{-a(t-T)}u(t-T) \leftrightarrow e^{-sT}F(s+a)$</td>
</tr>
<tr>
<td>damped and delayed</td>
</tr>
<tr>
<td>$f(-t)u(-t) \leftrightarrow F(-s)$</td>
</tr>
<tr>
<td>reverse time</td>
</tr>
<tr>
<td>$f(-t)e^{-at}u(-t) \leftrightarrow F(a-s)$</td>
</tr>
<tr>
<td>time-reversed &amp; damped</td>
</tr>
<tr>
<td>$\frac{d}{dt}f(t) = \delta'(t) \ast f(t) \leftrightarrow sF(s)$</td>
</tr>
<tr>
<td>deriv</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Additional transforms</th>
</tr>
</thead>
<tbody>
<tr>
<td>$\frac{\sin(t)u(t)}{t} \leftrightarrow \tan^{-1}(1/s)$</td>
</tr>
<tr>
<td>half-sync</td>
</tr>
</tbody>
</table>

F.1 Methods for automating the calculation of residues

In this appendix we shall set up the general problem of finding $K_k$ given Eq. 1.52 (Gustavsen and Semlyen, 1999).

$$Z(s) = \frac{N(s)}{D(s)} = sL_o + R_o + \sum_{k=0}^{K} \frac{K_k}{s - s_k}, \quad (F.1)$$

given the roots $s_k$ of polynomial $D(s) = \prod_{k=1}^{K} (s - s_k) = 0.$

1. First discuss the general properties of $Z(s) = K_{-1}s + K_0 + \sum_{k=k_1}^{K} \frac{K_k}{s - s_k}.$

**Exercise:** The impedance may be written as

$$Z(s) = \frac{N(s)}{D(s)} = \frac{\sum_{m} n_m s^m}{\sum_{k} d_k s^k}$$
Thus the companion matrix for the numerator polynomial $N(s)$ is

$$Z_N = \begin{bmatrix}
-d_{N-1} & d_{N-2} & \cdots & \cdots & -d_0 \\
1 & 0 & \cdots & 0 \\
0 & 1 & \cdots & \vdots \\
\vdots & \vdots & \ddots & \vdots \\
0 & 0 & \cdots & 1 \\
\end{bmatrix}, \quad \text{(F.2)}$$

thus numerator polynomial of $Z(s)$ is $N(s) = |Z_N - sI_N|$.

The companion matrix for the denominator polynomial $D(s)$ is

$$Y_N = \begin{bmatrix}
-n_{N-1} & n_{N-2} & \cdots & \cdots & -n_0 \\
1 & 0 & \cdots & 0 \\
0 & 1 & \cdots & \vdots \\
\vdots & \vdots & \ddots & \vdots \\
0 & 0 & \cdots & 1 \\
\end{bmatrix}, \quad \text{(F.3)}$$

while the denominator polynomial of $Z(s)$ is $D(s) = |Y_N - sI_N|$.

2. Note that these equations need to include the determination of unknowns $L_o$ and $R_o$, which in some cases will be zero.

3. Next transform the residue expansion into its companion matrix (Sect. 1.4.4, Eq. 1.37 p. 66). By definition, the eigen-values of the companion matrix are the same as the roots of the impedance matrix.

4. General method to substitute $s = s_k$ in $Z(s)$, to define a non-degenerate linear system of equations in $k_k$, having nonzero determinant ($\Delta_K \neq 0$) (Gustavsen and Semlyen, 1999). The method is recursive if the roots are not accurately known. It might be useful to review Newton’s method along with the Vandermonde determinant, to see if this might be further optimized.

5. This method has close ties to the classic CFA, where it has been called the Cauer decomposition, named after its inventor Wilhelm Cauer (Cauer and Mathis, 1995; Cauer et al., 2000; Cauer, 1958; Cauer et al., 1958), who acted as the primary thesis advisor for Brune (Brune, 1931b; Van Valkenburg, 1964b).
# Appendix G

## Stillwell’s Intersection vs. composition

<table>
<thead>
<tr>
<th>Page/c Group</th>
<th>Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>6BCE</td>
<td>Pythagoreans: Pythagoreans used recursion to solve PE: $y_n/x_n \rightarrow \sqrt{2}$</td>
</tr>
<tr>
<td>44</td>
<td>Greek: Find sides of $\sqrt{2}$ rectangle with Euclidean Alg -&gt; Matrix recurrence relation</td>
</tr>
<tr>
<td>46</td>
<td>Greeks: Other instances of PE occurred in Greek Math (anthypairesis)</td>
</tr>
<tr>
<td>45</td>
<td>Pythagoreans: First use of 2x2 recurrence gave sols to Pell’s Eq. (N=2): $5x^2-2y^2=1$</td>
</tr>
<tr>
<td>3BCE</td>
<td>Archimedes: Cattle problem: $x^2 - 427,434 y^2 = 1$</td>
</tr>
<tr>
<td>47</td>
<td>Archimedes: Cattle problem: $x^2 - 427,434 y^2 = 1$</td>
</tr>
<tr>
<td>7CE</td>
<td>Greeks: Pell’s Eq. both recurrence and composition methods</td>
</tr>
<tr>
<td>46</td>
<td>Indians: Indian Mathematicians” (Ch 5, p. 69)</td>
</tr>
<tr>
<td>37</td>
<td>Bramagupta: Rediscovered PE, both recurrence and composition methods</td>
</tr>
<tr>
<td>69</td>
<td>Brahmagupta: Composition sol method of Pell’s Eq (limited success, specific cases only)</td>
</tr>
<tr>
<td>76</td>
<td>Bhaskara II: Cyclic sol for N=61 $\pmod 4$ (1160CE)</td>
</tr>
<tr>
<td>12CE</td>
<td>Fermat: Rediscovered N=61 case (hard case p. 80) (1657)</td>
</tr>
<tr>
<td>79</td>
<td>Fermat: Related to continued fractions and $\sqrt{N}$ irrational case</td>
</tr>
<tr>
<td>80</td>
<td>Newton: Works out Diophantus’ chord/tangent methods (Stillwell p. 7, 49, 218)</td>
</tr>
<tr>
<td>18CE</td>
<td>Lagrange: First to prove why cyclic methods worked (Andrew Wiles, 1984) (1768)</td>
</tr>
</tbody>
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Bibliography


Campbell, G. (1903), “On Loaded Lines in Telephonic Transmission,” Phil. Mag. 5, 313–331, see Campbell22a (footnote 2): In that discussion, ... it is tacitly assumed that the line is either an actual line with resistance, or of the limit such with R=0.


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Campbell, G. (1937), \textit{The collected papers of George Ashley Campbell} (AT&T, Archives, 5 Reiman Rd, Warren, NJ), forward by Vannevar Bush; Introduction by E.H. Colpitts.


Fagen, M. (Ed.) (1975), \textit{A history of engineering and science in the Bell System — The early years (1875-1925)} (AT&T Bell Laboratories).


Fine (2007), \textit{Number Theory: An Introduction via the Distribution of Primes} (Birkhäuser Boston, Boston, MA), chap. 4, pp. 133–196, URL \url{http://dx.doi.org/10.1007/978-0-8176-4541-0_4}.


Menzies, G. (2008), 1434: The year a Magnificent Chinese fleet sailed to Italy and ignited the renaissance (HarperCollins, NYC).


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