Chapter 1
INTRODUCTION TO DIGITAL SIGNAL PROCESSING
1.6 Analog Filters
1.7 Applications of Analog Filters

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Analog filters have been in use since 1915 but with the emergence of digital technologies in the 1960s, they began to be replaced by digital filters in many applications.

This presentation will provide a brief historical background on analog filters and their applications.
Filtering can be used to pass one or more desirable bands of frequencies and simultaneously reject one or more undesirable bands.

For example,

- **lowpass filtering** can be used to pass a band of preferred low frequencies and reject a band of undesirable high frequencies.
- **highpass filtering** can be used to pass a band of preferred high frequencies and reject a band of undesirable low frequencies.
- **bandpass filtering** can be used to pass a band of frequencies and reject certain low- and high-frequency bands.
- **bandstop filtering** can be used to reject a band of frequencies but pass certain low- and high-frequency bands.
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- **bandstop filtering** can be used to reject a band of frequencies but pass certain low- and high-frequency bands.
To illustrate the filtering process consider an arbitrary periodic signal which is made up of a sum of sinusoidal components such as

\[ x(t) = \sum_{i=1}^{9} A_i \sin(\omega_i t + \theta_i) \]

where \( A_i \) is the amplitude and \( \theta_i \) is the phase angle of the \( ith \) sinusoidal component.
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where \( A_i \) is the amplitude and \( \theta_i \) is the phase angle of the \( ith \) sinusoidal component.

Arbitrary amplitudes and phase angles can be assigned to the various sinusoidal components as shown in the next slide.
<table>
<thead>
<tr>
<th>$i$</th>
<th>$\omega_i$</th>
<th>$A_i$</th>
<th>$\theta_i$</th>
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<tr>
<td>9</td>
<td>9</td>
<td>0.4103</td>
<td>-0.2722</td>
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</tbody>
</table>
Time-domain representation:
Frequency-domain representation:

Amplitude spectrum

Phase spectrum

Frame # 7 Slide # 14 A. Antoniou Digital Signal Processing – Secs. 1.6, 1.7
The filtering process can be represented by a block diagram as shown in the figure where \( x(t) \) is the input and \( y(t) \) is the output of the filtering process.

\[
\begin{align*}
\text{Filtering} \\
\begin{array}{c}
\downarrow \quad \text{Filtering} \\
\downarrow \\
\end{array}
\end{align*}
\]

\( x(t) \quad y(t) \)

\( x(t) \quad t \quad y(t) \quad t \)
Lowpass Filtering

Lowpass filtering will pass low frequencies and reject high frequencies as shown in the next two slides.
Lowpass Filtering Cont’d

Input

Output
Lowpass Filtering *Cont’d*

Input

```
-10  -5  0  5  10  15
```

Output

```
-10  -5  0  5  10  15
```
Highpass Filtering

Highpass filtering will pass high frequencies and reject low frequencies as shown in the next two slides.
Highpass Filtering \textit{Cont’d}

\begin{align*}
\text{Input} & \quad \text{Amplitude spectrum} \\
\text{Output} & \quad \text{Phase spectrum}
\end{align*}
Highpass Filtering \textit{Cont’d}

![Input and Output Waveforms](image)
Bandpass Filtering

Input

Output
In the first presentation, the filtering process was described as a process that will manipulate the spectrum of a signal in some way.
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In this fairly broad definition, several other filtering processes can be identified such as

- Differentiation
- Integration
Differentiation

If we differentiate the signal

\[ x(t) = \sum_{i=1}^{9} A_i \sin(\omega_i t + \theta_i) \]

with respect to \( t \), we get

\[ \frac{dx(t)}{dt} = \sum_{i=1}^{9} \frac{d}{dt}[A_i \sin(\omega_i t + \theta_i)] = \sum_{i=1}^{9} \omega_i A_i \cos(\omega_i t + \theta_i) \]

\[ = \sum_{i=1}^{9} \omega_i A_i \sin(\omega_i t + \theta_i - \frac{1}{2} \pi) \]
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\[ = \sum_{i=1}^{9} \omega_i A_i \sin(\omega_i t + \theta_i - \frac{1}{2}\pi) \]

We note that the amplitude and phase spectrums of the signal have become

\{\omega_i A_i : i = 1, 2, \ldots, 9\} and \{\theta_i - \frac{1}{2}\pi : i = 1, 2, \ldots, 9\} respectively.
\{\omega_i A_i : i = 1, 2, \ldots, 9\} \quad \text{and} \quad \{\theta_i - \frac{1}{2}\pi : i = 1, 2, \ldots, 9\}

In effect,

- the amplitude spectrum has been multiplied by the frequency $\omega_i$, and
...\[
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\]

In effect,

▶ the amplitude spectrum has been multiplied by the frequency $\omega_i$, and

▶ an angle of $\pi/2$ has been subtracted from the phase spectrum.
Differentiation \textit{Cont’d}

Input

Output
Differentiation Cont’d

Input

Output

Digital Signal Processing – Secs. 1.6, 1.7
Evidently, differentiation tends to enhance high-frequency components and weaken low-frequency components somewhat like highpass filtering.
If we integrate the signal

\[ x(t) = \sum_{i=1}^{9} A_i \sin(\omega_i t + \theta_i) \]

with respect to \( t \), we get

\[
\int x(t) \, dt = \sum_{i=1}^{9} \int A_i \sin(\omega_i t + \theta_i) \, dt = \sum_{i=1}^{9} \left[ -\frac{A_i}{\omega_i} \cos(\omega_i t + \theta_i) \right]
\]

\[ = \sum_{i=1}^{9} \frac{A_i}{\omega_i} \sin(\omega_i t + \theta_i - \frac{1}{2}\pi) \]
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\end{align*}

Evidently,

- the amplitude spectrum has been divided by the frequency $\omega_i$, and
\{A_i/\omega_i : i = 1, 2, \ldots, 9\} \quad \text{and} \quad \{\theta_i - \frac{1}{2}\pi : i = 1, 2, \ldots, 9\}

Evidently,

- the amplitude spectrum has been divided by the frequency \(\omega_i\),
- and
- an angle of \(\pi/2\) has been subtracted from the phase spectrum.
Integration Cont’d

Input

Output
Integration Cont’d
Evidently, integration tends to enhance low-frequency components and weaken high-frequency components somewhat like lowpass filtering.
Electrical engineers have known about filtering processes for well over 80 years and through the years they invented a great variety of circuits and systems that can perform filtering, which are known collectively as filters.
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Electrical filters can be classified on the basis of their operating signals as analog or digital.

In analog filters the input, output, and internal signals are in the form of continuous-time signals whereas in digital filters they are in the form of discrete-time signals.
Analog filters were originally invented for use in radio receivers and long-distance telephone systems and continue to be critical components in all types of communication systems.
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Various families of analog filters have evolved over the years, which can be classified as follows on the basis of their constituent elements and the technology used:

- Passive $RLC$ filters
- Discrete active $RC$ filters
- Integrated active $RC$ filters
- Switched-capacitor filters
- Microwave filters
Passive RLC Filters

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- They are made of interconnected resistors, inductors, and capacitors and are said to be passive in view of the fact that they do not require an energy source, like a power supply, to operate.

Filtering action is achieved through the property of electrical resonance which occurs when an inductor and a capacitor are connected in series or in parallel.

The importance of filtering in communications motivated engineers and mathematicians between the thirties and fifties to develop some very powerful and sophisticated methods for the design of passive RLC filters.
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Passive $RLC$ Filters \textit{Cont’d}

Passive $RLC$ lowpass filter
Discrete Active RC Filters

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- They are made up of discrete resistors, capacitors, and amplifying electronic circuits.
- Inductors are absent and it is this feature that makes active RC filters attractive.
Inductors have always been bulky, expensive, and generally less ideal than resistors and capacitors particularly for low-frequency applications. Unfortunately, without inductors electrical resonance cannot be achieved and with just resistors and capacitors only crude types of filters can be designed. However, through the clever use of amplifying electronic circuits in $RC$ circuits, it is possible to simulate resonance-like effects that can be utilized to achieve filtering of high quality. These filters are said to be active because the amplifying electronic circuits require an energy source in the form of a power supply.
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Discrete Active RC Filters Cont’d

Discrete active bandpass filter
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Interest in these filters has been strong during the eighties and nineties and research is continuing.
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Switched-Capacitor Filters

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△ These are essentially active *RC* filters except that switches are also utilized along with amplifying devices.

△ In this family of filters, *switches are used to simulate high resistance values which are difficult to implement in integrated-circuit form*.

△ Like integrated active *RC* filters, switched-capacitor filters are compatible with integrated-circuit technology.
At microwave frequencies *in the range 0.5 to 500 GHz*, *inductors and transistors do not work very well* and, therefore, passive *RLC* or active *RC* filters have poor performance; hence, *microwave filters* are used.
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Microwave filters are built from a variety of microwave components and devices such as waveguides, dielectric resonators, and surface acoustic devices.
Microwave bandpass filter
Applications of Analog Filters

Radios and TVs
Applications of Analog Filters

- Radios and TVs
- Communication and radar systems
Applications of Analog Filters

- Radios and TVs
- Communication and radar systems
- Telephone systems
Applications of Analog Filters

- Radios and TVs
- Communication and radar systems
- Telephone systems
- Sampling systems
Applications of Analog Filters

- Radios and TVs
- Communication and radar systems
- Telephone systems
- Sampling systems
- Audio equipment
When we select our favorite radio station or TV channel, we are actually tuning a bandpass filter inside the radio or TV to the frequencies of the radio or TV station.

The signal from our favorite radio station is the desirable signal and the signals from all the other stations are undesirable.
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The signal from our favorite radio station is the desirable signal and the signals from all the other stations are undesirable.

The same principle can be used to prevent radar signals from interfering with communications channels and vice-versa at an airport.
Signals are often corrupted by spurious signals known collectively as *noise*.

Such signals may originate from a large number of sources, e.g., lightnings, electrical motors, transformers, and power lines.

Noise signals are characterized by frequency spectrums that stretch over a wide range of frequencies. They can be eliminated through the use of bandpass filters that would pass the desired signal but reject everything else, namely, the noise content.
We all talk to our friends and relatives, who may live in another city or another country, almost daily through the telephone system. The telephone signals are transmitted through expensive communications channels.

If these channels were to carry just a single voice, as in the days of Alexander Graham Bell, no one would ever be able to afford a telephone call to anyone, even the very rich.
What makes long-distance calls affordable is our ability to transmit thousands upon thousands of conversations through one and the same communications channel.
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An FDM communication system requires a multitude of filters to operate properly.
The operation of a typical FDM communications system is as follows:

1. At the transmit end, the different voice signals are superimposed on different carrier frequencies using a process known as modulation.

2. The different carrier frequencies are combined by using an adder circuit.

3. At the receive end, carrier frequencies are separated using bandpass filters.

4. The voice signals are then extracted from the carrier frequencies through demodulation.

5. The voice signals are distributed to the appropriate persons through the local telephone lines.
Frequency-Division Multiplex System Cont’d

(a) Basic FDM system

(b) Voice signals arranged into a group
The transmit section adds the frequency of a unique carrier to the frequencies of each voice signal, thereby, shifting its frequency spectrum by the frequency of the carrier.

In this way, the frequency spectrums of the different voice signals are arranged one after the other to form the composite signal $g(t)$ shown in figure (a) of the previous slide, which is referred to as a group by telephone engineers.

The amplitude spectrum of $g(t)$, designated as $G(\omega)$, is illustrated in Fig. (b).
The receive section separates the translated voice signals and restores their original spectrums.

The FDM system requires as many bandpass filters as there are voice signals, and this is why thousands upon thousands of bandpass filters are required.

The FDM system also uses a large number of modulators and demodulators and these devices, as it turns out, also need filters to operate properly.

In short, communications systems are simply not feasible without filters.
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At the receiving end, a supergroup is subdivided into the individual groups by a bank of bandpass filters. The groups are, in turn, subdivided into the individual voice signals by appropriate banks of bandpass filters.
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Similarly, several supergroups can be combined into a mastergroup, and so on, until the bandwidth capacity of the cable or microwave link is completely filled.
FDM system with two levels of modulation.
In a sampling system, the sampling frequency must be at least twice the highest frequency present in the spectrum of the signal. This is known as the *sampling theorem*. In situations where the sampling frequency is fixed and the highest frequency present in the signal can exceed half the sampling frequency, it is crucial to bandlimit the signal to be sampled to prevent a certain type of signal distortion known as aliasing. This bandlimiting process must be carried out through the use of a lowpass analog filter. A sampling system also requires an analog lowpass filter at the output of the D/A converter to serve as a smoothing device.
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Use of Analog Filters in Sampling Systems

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Use of Analog Filters in Sampling Systems \textit{Cont'd}

\[ x(t) \imes x(t) \times x(nT) \times y(nT) \times y(t) \times y(t) \times c(t) \times F_{LP} A/D D/F D/A F_{LP} \]

Frame # 51 Slide # 95 A. Antoniou Digital Signal Processing – Secs. 1.6, 1.7
Loudspeaker systems behave very much like filters and, consequently, they tend to change the spectrum of an audio signal.
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Loudspeaker systems behave very much like filters and, consequently, they tend to change the spectrum of an audio signal. This is due to the fact that the enclosure or cabinet used can often exhibit mechanical resonances that are superimposed on the audio signal. This is one of the reasons why different makes of loudspeaker systems often produce their own characteristic sound.
To correct for mechanical resonances and other imperfections, sound reproduction equipment, such as stereos, is often equipped with *equalizers* that can be used to reshape the spectrum of the audio signal.

These subsystems typically incorporate a number of sliders that can be adjusted to modify the quality of the sound reproduced.

One can, for example, strengthen or weaken the low-frequency (bass) or high-frequency (treble) content of the audio signal.
Since an equalizer is a device that can modify the spectrum of a signal, \textit{equalizers are filters} in terms of the broader definition adopted in the textbook.

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What the sliders do is to alter the parameters of the filter that performs the equalization.

Through the use of an equalizer, one could adjust the spectrum of the audio signal to one’s preference.

A thick carpet can actually absorb a lot of the high-frequency content of the audio signal, i.e., the room would behave very much like a lowpass filter.

In such a situation, one might need to boost the treble a bit to restore some of the lost high-frequency content of the music.
Transmission cables, telephone lines, and communication channels often behave very much like filters and, as a result, they tend to reshape the spectrums of the signals transmitted through them.

The local telephone lines are particularly notorious in this respect – we often do not even recognize the voice of the person at the other end only because the spectrum of the signal has been changed by the telephone line.
Like the performance of loudspeaker systems, that of telephone lines and communication channels can be improved by using suitable equalizers.

In fact, it is through the use of sophisticated equalizers, in the form of adaptive filters, that it is possible to achieve high data transmission rates through local telephone lines.
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In fact, it is through the use of sophisticated equalizers, in the form of *adaptive filters*, that it is possible to achieve high data transmission rates through local telephone lines.

These equalizers are incorporated in the modems at either end of a telephone line.

So-called ADSL Internet service available through telephone companies is achieved by means of adaptive filters.
This slide concludes the presentation.
Thank you for your attention.