Welcome Back !!

to

Week # 4 of Class

for

EXTRA CLASS Radio License

Santa Fe Trail ARC

KSØKS

Olathe, KS

George McCarville, WBØCNK Jan, 2026

Chapter 6

Radio Circuits and Systems

George McCarville
WBØCNK

Chapter 6

Overview of Chapter 6

This chapter covers a huge number of topics and large amount of material

- It describes
  - Generating a signal (Transmitting)
  - Receiving a signal (Receiving)
- The sections cover “building blocks” used in Transmitting or Receiving

⇨ It provides for some comparisons between:

Older Technology
- Vacuum Tubes
- Mostly Analog
- Appears Simpler

Newer Technology
- Transistors, FETSMOSFETS
- Greatly Expands Digital Signals & Processing
- Can Appear to Be Very Complex

⇨ Both have advantages and or disadvantages

Chapter 6

➔ 6.1 - Amplifiers

➔ 6.2 - Signal Processing

➔ 6.3 - Digital Signal Processing (DSP)
- Software Defined Radio (SDR)

➔ 6.4 - Filters and Impedance Matching

➔ 6.5 - Power Supplies (will be presented next week)

Amplifiers 6.1

Definitions:

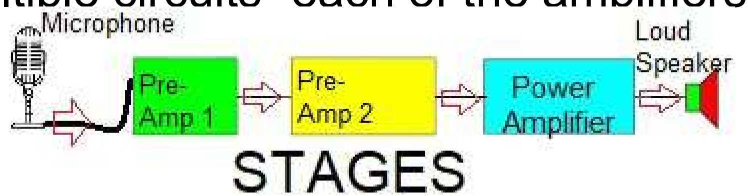
Amplifier: Any circuit that increases the strength of a signal ,voltage, current, or power.

Range of Amplifier types:

- High power signal types
- Numerous small types - internal to radios and test instruments.
- Input voltages from micro volts to hundreds of volts.
- Output powers from billionths of a watt to thousands of watts +.

Terms to know: Gain, input & output Impedance

- Stage: Multiple circuits- each of the amplifiers working together



- Driver: A circuit stage that supplies the input signal to the amplifier.
- Load: A circuit stage that receives the amplifier's output signal.
- Final Amplifier: The last amplifier stage in a transmitter.
- Loading: Attaching a load to the amplifier output

Amplifiers

Amplifier Gain

Amplifier Gain.

- Is the ratio of output signal to input signal
  (can be a voltage, a current, a power.

- Can be measured by any of these:
  - Voltage gain = V_{OUT} / V_{IN}
  - Current gain = I_{OUT} / I_{IN}
  - Power gain = P_{OUT} / P_{IN}

- Can be expressed as simple ratio.
- e.g. – Voltage gain = V_{out} / V_{in} (10V / 1V = 10)

- Can be expressed in decibels.
  e.g. – Power gain = 10 dB

  - Recall Decibels: Is the ratio in terms of a logarithm
    - Example: + 3dB (2 X signal, voltage, current) e.g. 2 watts → 4 watts
    - Example: - 3dB (½ signal , voltage, current) e.g.10 watts → 5 watts

Amplifiers

Input & Output Impedance

Input & Output Impedance:

- Anything connecting to an amplifier is “seen” as an impedance.
- No single component in the amplifier creates the impedance.
- It is a combined effect from all the amplifier components.

Input impedance is the load “seen” by driver.

- Output Impedance(of an amplifier) is source impedance(to the next stage.

  50**Ω** Transmitter final amp “sees” the Antenna impedance – hopefully also 50**Ω**

- Maximum power transfer occurs when source & load impedances are equal. (This is why antennas and RF Signal amplifiers should be both 50**Ω)**

Amplifiers

Basic Circuits

Note: For the discussion on amplifiers :

- The Book uses only Bipolar Transistor circuits

- The techniques explained <u>also apply to FETs</u> and <u>vacuum tubes</u>.

Amplifiers

Basic Circuits

Generally there are 3 Types of bipolar transistor amplifier circuits:
Called:

- Common-emitter
- Common-base
- Common-collector

(more on each of these in follow-on slides)

- "Common" means that a particular electrode serves as a reference for
- both the input and output.
-
- – usually ground is "common".

- Recall: The bipolar transistor is essentially a current amplifier. (5ma. ► 15ma.)
  Current in the base-emitter controls larger currents in the collector-emitter.

- Condition needed for bipolar transistor operation:
  ⇨ The bipolar-transitor base-emitter must be forward biased and the base-collector reversed biased in order to act as an amplifier.(ref: P5.8-5.9)

Forward and Reverse Bias:

- <u>In NPN circuits</u>:
  collector and base must be positive with respect to the emitter.
- <u>Conversely for PNP</u>:
  base and collector are negative with respect to the emitter.
- The combination of bias and collector-emitter current is called the <u>"operating Point"</u>.
- With no input – the operating point is called the *quiescent point* or "Q-Point".   (more when we visit graphs showing operating current).

Amplifiers

Common – Emitter and Common Collector

Common emitter.

- Most common amplifier type

- Provides both voltage gain and current gain
- Input & output voltages 180° out of phase
- Fairly high input impedance
- Internal heating causes an increase of BJT gain. Thermal runaway is controlled with increased degenerative feedback voltage of R3.
- Output impedance depends on RL

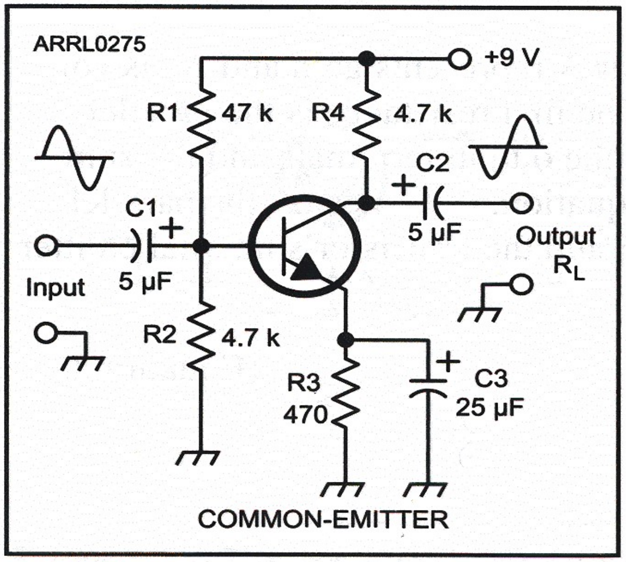


Figure 6-1 — The common-emitter amplifier circuit has a fixed-bias source (R1 and R2), degenerative emitter feedback for bias stabilization (R3), and a resistive collector load (R4). C1 and C2 are dc blocking capacitors for input and output signals. C3 acts as an emitter bypass to increase ac gain while maintaining stable behavior at dc.

You can recognize the Common-Emitter circuit by the value of resistance in the emitter circuit R3 in Figure 6.1, it's smaller than R4 (or absent) than in the collector circuit, the emitter resistor may also be bypassed with a capacitor C3.

Amplifiers

Common Emitter

Common emitter Both input and output use the Emitter as "reference"(it is AC grounded through C3

- R_1 & R_2 provides a fixed bias & stable operating point.
  (voltage divider is powered by +9V)
- C_2 allows maximum AC signal gain
- R3 in series with emitter
  - provides self bias
  - voltage is generated across R3 as current through R3 increases
  - (degenerative feedback)
  - This prevents thermal runaway

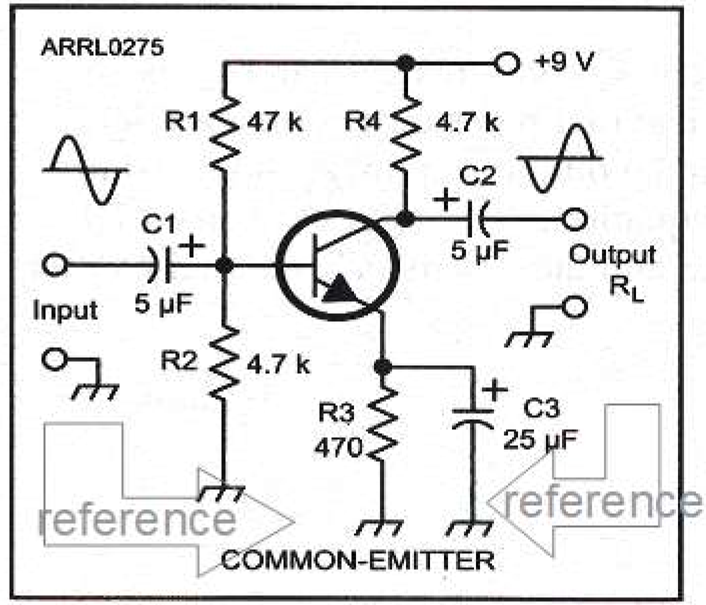


Figure 6-1 — The common-emitter amplifier circuit

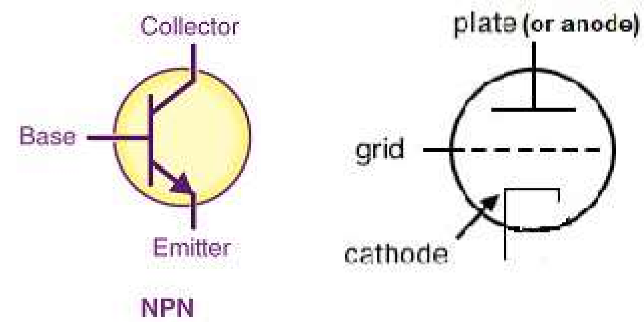
- **The type of amplifier circuit shown in Figure 6-1 is: → Common Emitter**
- **In Fig 6-1, what is the purpose of R1 and R2? → Voltage Divider Bias (fixed)**
- **In Fig 6-1, what is the purpose of R3? → Self Bias**

Amplifiers

Similarities of Vacuum Tubes

Each of the bipolar transistor <u>electrodes</u> has
A comparable vacuum tube <u>electrode.</u>

- Transistor emitter ~ tube cathode
- Transistor base ~ tube grid
- Transistors collector ~ tube anode or tube plate



Each type of transistor amplifier circuit has a corresponding vacuum tube amplifier circuit.

- Common-Emitter ~ Common-Cathode.
- Common-Base ~ Grounded-Grid.
- Common-Collector ~ Common-Anode.
- Emitter Follower ~ Cathode Follower.

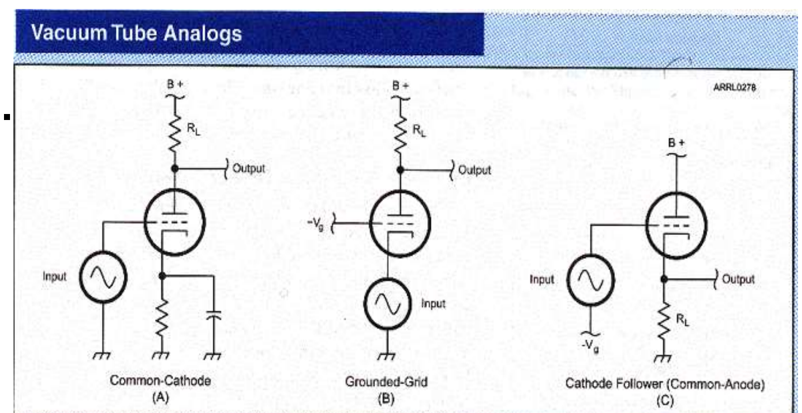
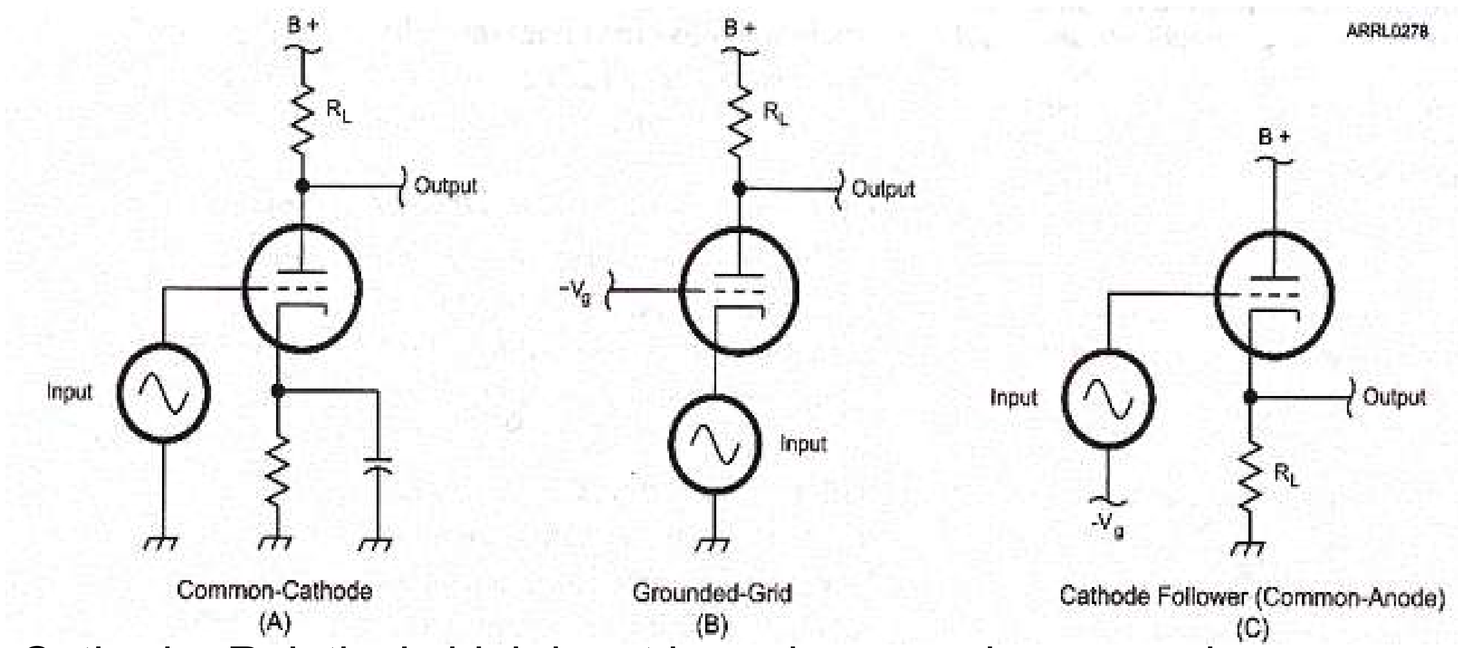


Figure 6-4 — Each of the common tube amplifier configurations — common-cathode (A), grounded-grid (B), common-anode (C) — has its analog in transistor amplifier circuits.

Amplifiers

Similarities of Vacuum Tubes



- Common Cathode: Relatively high input impedance and power gain
- Grounded Grid: Input signal at the cathode, no current gain, low input impedance (50**Ω**)
- Common-anode: Input to grid, high input impedance, no Voltage gain
  (rarely used in amateur equipment)

- **What is a characteristic of a grounded-grid amplifier?** ► **Low input impedance.**

Common - Collector or Emitter Follower Circuit

Both input and output use the Collector as "reference" (Collector is AC grounded)

Common Collector.

- Emitter Follower (another name)
- R3 is a load that acts to stabilize the circuit bias
- Has no current gain
  (collector current = emitter current)
- Input Voltage is "in-phase" with output
- Often used as a *buffer* amplifier
  to isolate low power sensitive stages from heavy varying output loads.

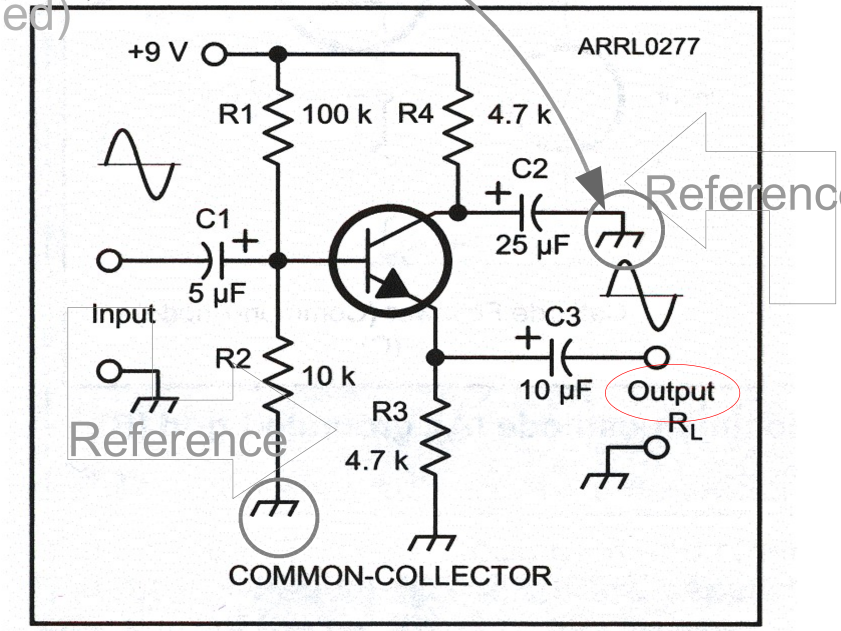


Figure 6-3 — The common-collector amplifier is also known as an emitter follower because the output voltage across the emitter resistor, R3, is in phase with and very nearly equal to that of the input signal.

- Remember: The signal is AC.
- Signal travels though C1 and C3 with ground as reference.
- On output. the AC signal ground is through C2

- **In Fig 6-3, what is the purpose of R3**
  ►**Self Bias**

- **What is characteristic of an emitter follower (or common collector) amplifier?**
  ►**Input and output signals are in-phase.**

For More Information

Common - Base Circuit

Common base.

- “Looks like it's on it's “side”
- R_1 & R_2 provide fixed bias.
  (voltage divider powered by +9V)
- C3 provides a steady dc forward bias
- Has no current gain
  (collector current = emitter current)
- Input Voltage is “in-phase” with output
- Low input impedance (~ 50**Ω**)
- High output impedance
- Used as an <u>impedance converter</u>
- Used as an RF Pre-amplifier
- Amplifier can have large voltage gains

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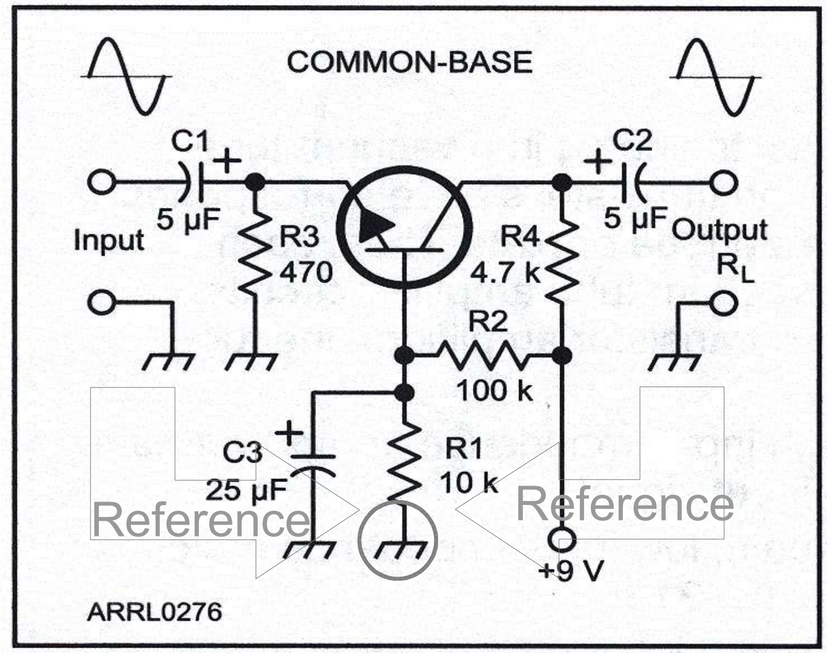


Figure 6-4 — The common-base amplifier

Both input and output use the grounded Base as “reference”

Break ?

Amplifiers
Op Amp Amplifiers
Introduction

Op Amp Amplifiers

- High-gain, direct-coupled, differential amplifier
  - Two inputs
    one inverting (labeled -)
    one non-inverting (labeled +)
  - Differential input → Input signal is the difference between inverting & non-inverting inputs.
  - Direct-coupled → Will amplify DC as well as AC.

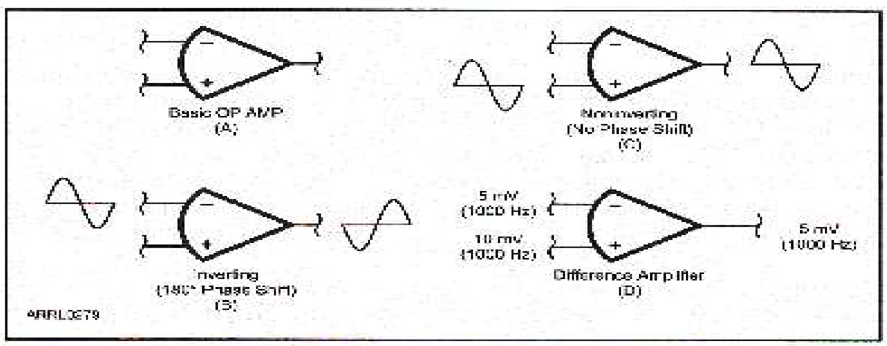


Figure 6-5 — Part A shows the basic schematic symbol for an operational amplifier (op amp). Parts B through D show how the output responds to signals at the op amp's inputs.

- **What is an operational amplifier?**

→ **A high-gain, direct-coupled, differential amplifier with very high input impedance and very low output impedance.**

Op Amp Characteristics

Op Amp Amplifiers

- High-gain, direct-coupled, differential amplifier.

- Ideal operational amplifier:
  - Infinite input impedance.
  - Zero output impedance.
  - Infinite voltage gain.
  - Flat frequency response.
  - Zero offset voltage. $0\ V_{IN} \Rightarrow 0\ V_{OUT}$

- **What is the typical output impedance of an integrated circuit op-amp?**
  ► **Very low.**
- **What is the typical input impedance of an integrated circuit op-amp?**
  ► **Very high**
- **How does the gain of an ideal operational amplifier vary with frequency?**
  ► **It does not vary with frequency.**

Amplifiers

Op Amp Characteristics

Circuit characteristics are totally determined by external components.

- *Loop Gain* (is between input and output)
- *Closed Loop*: Usually some output is fed back to inverting input acts to stabilize the gain.
- Gain remains constant over wide frequency
  - In practical op-amps loop gain decreases linearly with frequency
  - Gain *Bandwidth*: The frequency at which the open-loop gain of the amplifier equals one.

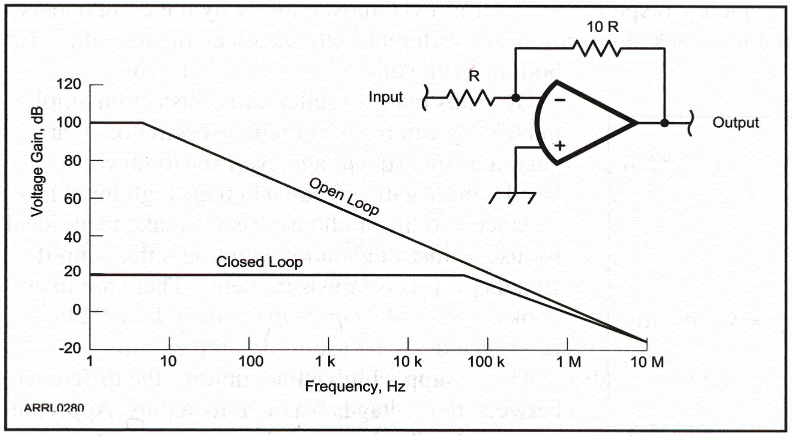


Figure 6-6 — The open-loop gain and closed-loop gain are shown as a function of frequency. The vertical separation between the curves is a measure of the feedback or gain margin.

What is the Gain- *Bandwidth of an operational amplifier?*

► **The frequency at which the open-loop gain of the amplifier equals one**

Amplifiers

Op Amp Characteristics

Operational amplifier characteristics continued:

- *Slew rate*: (max output voltage swing per unit of time)
  - 1V to 30 V per microsecond typical
- *Power bandwidth* (function of slew rate)
- Actual Op-Amps are not perfect
  - Shorted inputs should make 0V output.
  - *Input offset voltage* between inputs that will produce a 0V output.

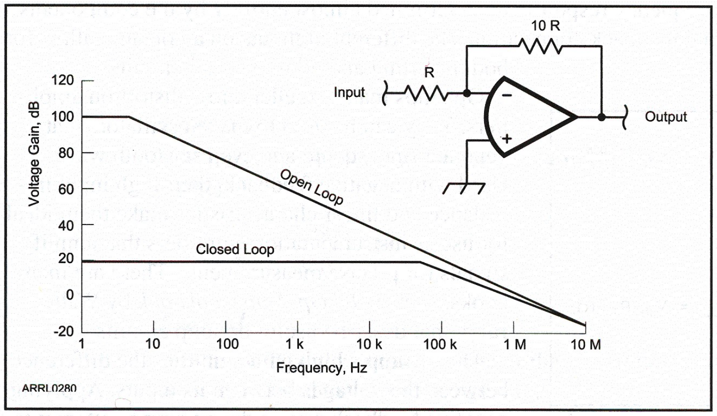


Figure 6-6 — The open-loop gain and closed-loop gain are shown as a function of frequency. The vertical separation between the curves is a measure of the feedback or gain margin.

- **What is the frequency response of the circuit in E7-3 if a capacitor is added across the feedback resistor?**
  - **→ Low-pass filter**
- **What is meant by the term op-amp input offset voltage?**

→ The differential input voltage needed to bring the open loop output voltage to zero.

Amplifiers

Basic Amplifier Circuits (for Op Amps)

- The most common application of op amp is in negative feedback circuits
- Operating from DC up to few hundred Kilohertz.
- The high gain amplifies difference between the inputs.
- Negative feedback caused the amp to try to drive inputs to 0V.
- The op amps high input impedance allows the current into the inputs to be ignored.

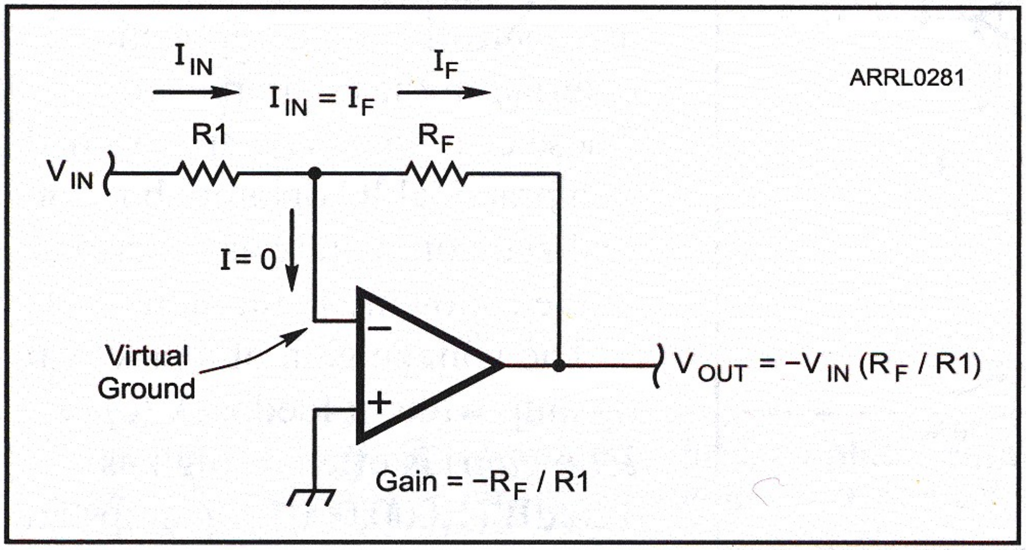


Figure 6-7 — Because of its high gain, negative feedback forces the op amp to keep the inverting and non-inverting inputs at nearly the same voltage. To do so requires balancing input current through R1 and feedback current in R_F, resulting in voltage gain of $-R_F/R1$.

Eq. 6.4

$$A_V = \frac{V_{out}}{V_{in}} = \frac{-R_f}{R1}$$

Basic Amplifier Circuits (for Op Amps)

Examples to work:

Example 6.1

What is the voltage gain of the circuit in Figure 6.7 if R1 = 1800 Ω and R_F = 68 kΩ?

$$|A_V| = \frac{R_F}{R1} = \frac{68000}{1800} = 38$$

- **What absolute voltage gain can be expected from the circuit in Figure ____ when R1 is 1800 ohms and RF is 68 Kilohms? 38.**

Basic Amplifier Circuits (for Op Amps)

Example 6.2

What is the voltage gain of the circuit in Figure 6.7 if R1 = 10 Ω and R_F = 470 Ω?

$$|A_V| = \frac{R_F}{R1} = \frac{470}{10} = 47$$

- **What magnitude of voltage gain can be can be expected from the circuit in Figure ____ when R1 is 10 ohms and RF is 470 ohms: 47**

Amplifiers

Basic Amplifier Circuits (for Op Amps)

Example 6.3

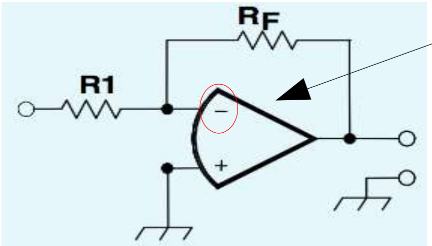
What is the voltage gain of the circuit in Figure 6.7 if R1 = 3300 Ω and R_F = 47 kΩ?

$$|A_V| = \frac{R_F}{R1} = \frac{47000}{3300} = 14$$

- **What absolute voltage gain can be expected from the circuit in Figure ____ when R1 is 3300 ohms and RF is 47 Kilohms?**
  14.

Example 6.4

What will be the output voltage of the circuit in Figure 6.7 if R1 = 1000 Ω and R_F = 10 kΩ and the input voltage = 0.23 V?



Inverting Circuit:

AV = 10,000 / 1000 = 10

Vout = - Av Vin = -10(.23) = -2.3V

- **What will the output voltage of the circuit shown in Figure ___be if R1 is 1000 ohms, RF is 10,000 ohms and 0.23 volts DC is applied to the input?**
  - 2.3 volts

Amplifiers
Op Amp Amplifiers
Comparators

- voltage comparator is another form of a special op amp circuit. Has 2 analog inputs. It "compares" the input voltages.
- External resistors generate a reference voltage Vsp
  - called the *setpoint or threshold*
- If non-inverting voltage is higher than the inverting, the output is driven to positive limit.
- If inverting input is higher than non-inverting the output will be driven to negative limit.
- Changes it's output depending on whether the unknown voltage is above or below the threshold (Vsp)
- Hysteresis is a form of positive feedback (from R3) that moves the setpoint in opposite direction the input crossed the setpoint, and prevents output "Chatter" due to noise on the input.

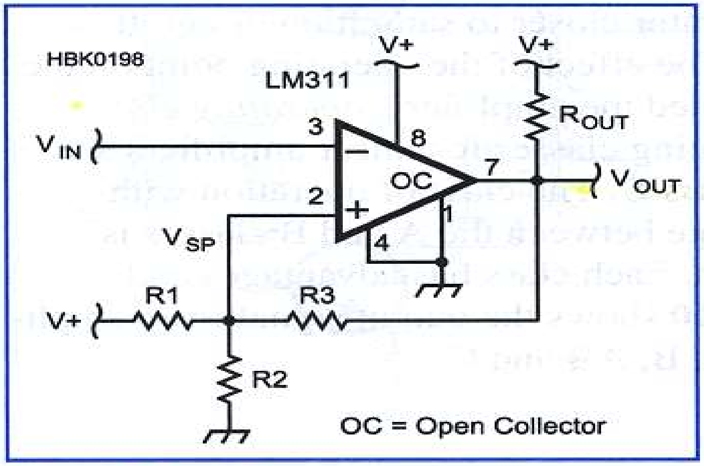


Figure 6.8 — A comparator circuit in which the output voltage is low when voltage at the inverting input is higher than the setpoint voltage, V_{SP}, at the noninverting input. R3 creates hysteresis by allowing more current to flow through R1 when the comparator output is low, shifting the setpoint by a few millivolts.

- **What is the function of hysteresis in a comparator?**
  ➛ To prevent input noise from causing unstable output signals.
- **What happens when the level of a comparators input signal crosses the threshold? ➛ The comparator changes its output state.**

Amplifiers
New Topic
Classes of operation – All Amplifiers

Fig 6.9 shows typical 3 regions of amplifier output versus input.

- As more signal is applied at input→ output increases due to Amp Gain.
- **Cutoff**
  - When collector drain current is very small
- **Saturation:**
  - When collector drain current is very large
  - Additional input causes less & less output.
- **Linearity**:
  - controlled by the Amp's *load line*
  - *operating point* on *load line.*
- ***Operating class:***
  - Is the effect of the operating point on the Amp's linearity.
- **3 Primary Classes: A, B ,C**
  (Other Classes have been developed: AB, D,E)

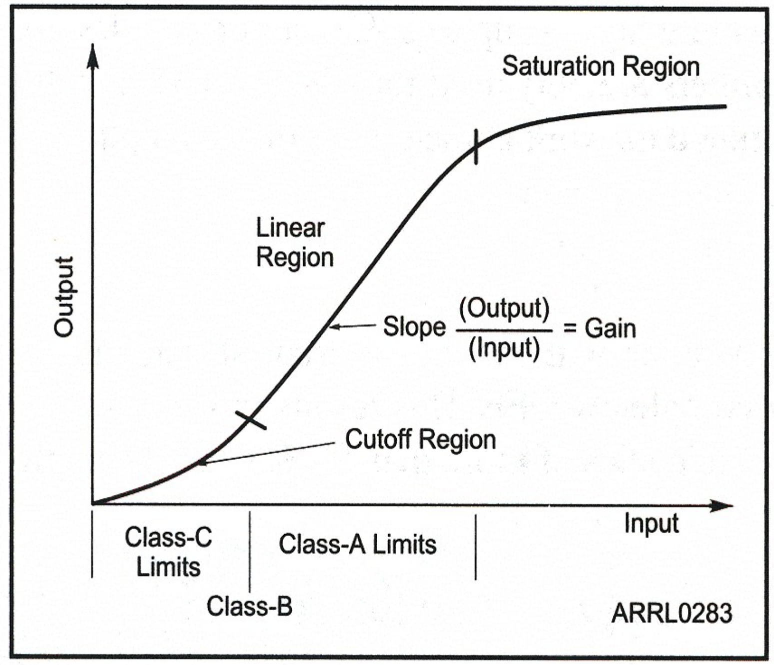


Figure 6-9 — This graph shows the three regions of amplifier operation: cutoff, linear and saturation. Each region of the curve corresponds to a different operating class.

Amplifiers

Classes of operation

What the wave forms are showing:

Class A
- Entire input waveform is used.

Class B
- Only top portion of the wave is used.

Class AB
- The top and some of the bottom of the wave is used.

Class C
- Only a small portion of the top of the wave is used.

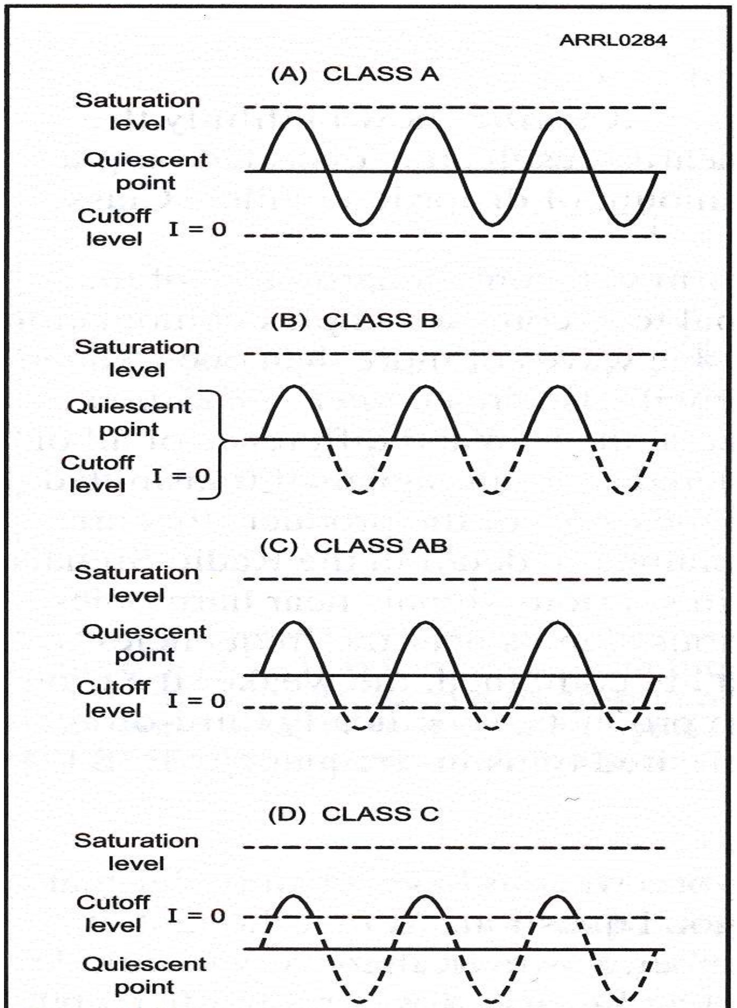


Figure 6-10

Amplifiers
Classes of operation
Class A

- Class A
  - On for 360°
  - Best linearity.
  - Least efficient
  - Only 25%-30% efficient
  - AM Radio Broadcast

Always on so inefficient but sounds great, inefficiencies cause 75% heat and rest great signal. **Hi end audio equip.**

What is the operating point of a Class A common emitter amplifier?

➤ Approximately halfway between saturation and cutoff.

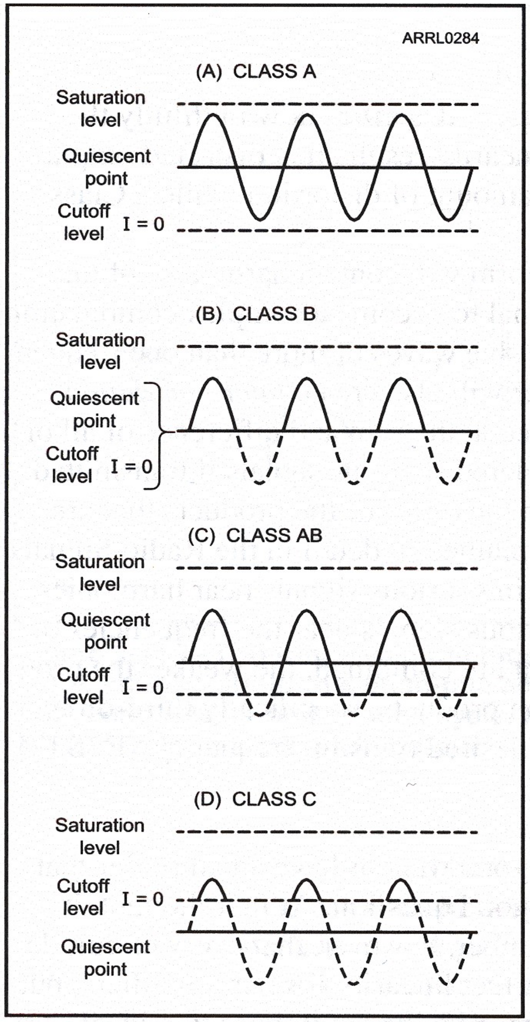


Figure 6-10 — Amplifier output waveforms for various classes of operation. All waveforms assume a sine-wave input signal.

Amplifiers
Classes of operation
Class B

- Class B
  - On for only 180°
  - Non-linear.
    - 2 devices in push-pull configuration is linear.
    - Near 60% efficient
  - Less of the full input audio
  - Non-linear distortion adds new frequencies that were not at the input (Even order and Odd order harmonics)

Not always on, switched on and off so some distortion in signal, but more efficient. Not used in many audio applications.

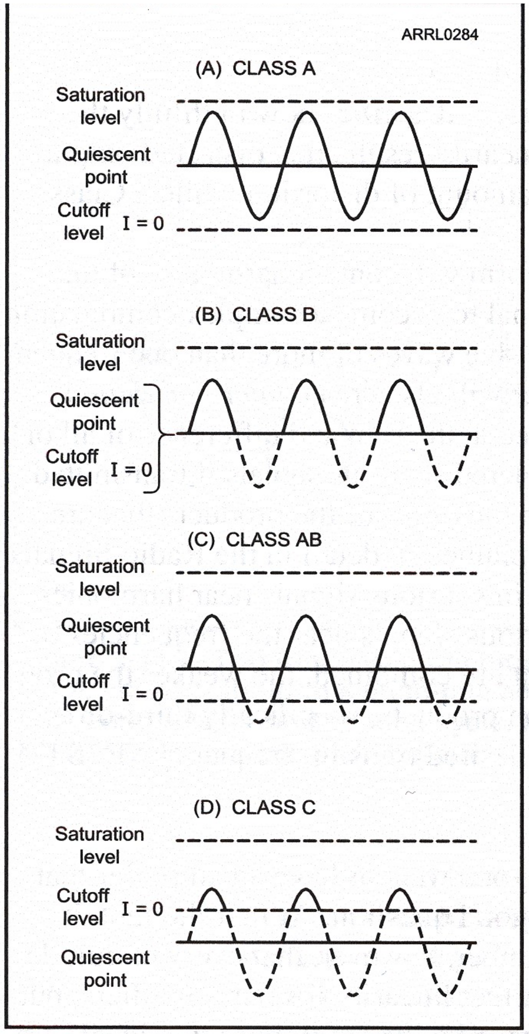


Figure 6-10 — Amplifier output waveforms for various classes of operation. All waveforms assume a sine-wave input signal.

Class B Special Case Push-Pull

Special Case: Class B Push Pull

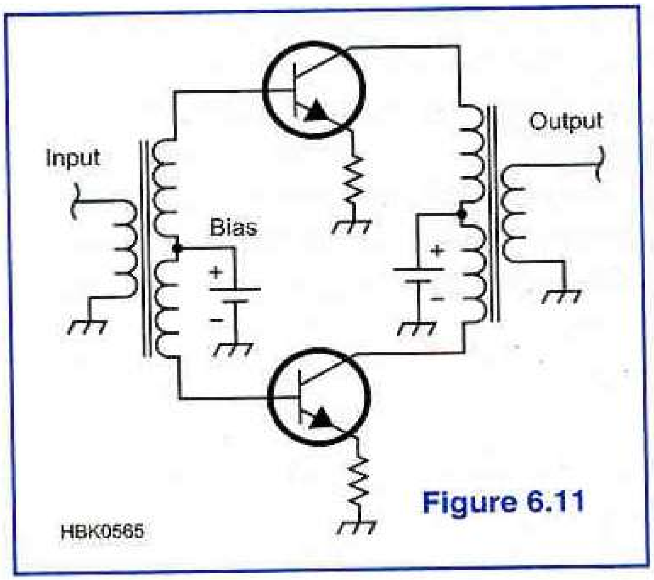


Figure 6.11

- Class B amplifiers are often used in audio and RF frequencies by connecting the two tubes or transistors in a push-pull circuit.
- While one transistor is cut off, the other is conducting, so both halves of the signal waveform are present in the output.
- This reduces the amount of distortion in the output and will reduce even-order harmonics.

Amplifiers
Classes of operation
Class AB

Class AB

- Is a "compromise" between classes A & B
- On for >180° but < 360°
- Bias is set to allow entry beyond the linear region & into the cutoff region on peaks
- Non-linear

A-B: Better efficiencies(40% heat) because not totally off like B. Used in regular audio equip.

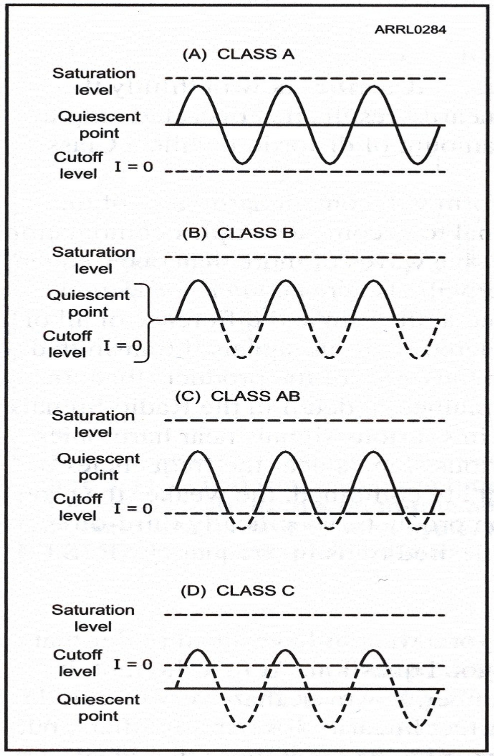


Figure 6-10 — Amplifier output waveforms for various classes of operation. All waveforms assume a sine-wave input signal.

For what portion of the signal cycle does each active element in a push-pull, Class AB amplifier conduct?

→ **More than 180 degrees but less than 360 degrees.**

Amplifiers
Classes of operation
Class C

Class C

- On for much less than 180°
- Highly non-linear.
- Creates pulses at signal freq.
- Very efficient -up to 80%
- Linearity is very poor, so use for SSB would result in too much distortion.

- For Class C Amps, the bias allows the operation past “cut off”region, conducting only during part of a half-cycle, creating pulses. The efficiency can be quite high up to 80%.
- **Which of the following is a likely result when a class C amplifier is used to amplify a single sideband phone signal? → Signal distortion and excessive bandwidth.**

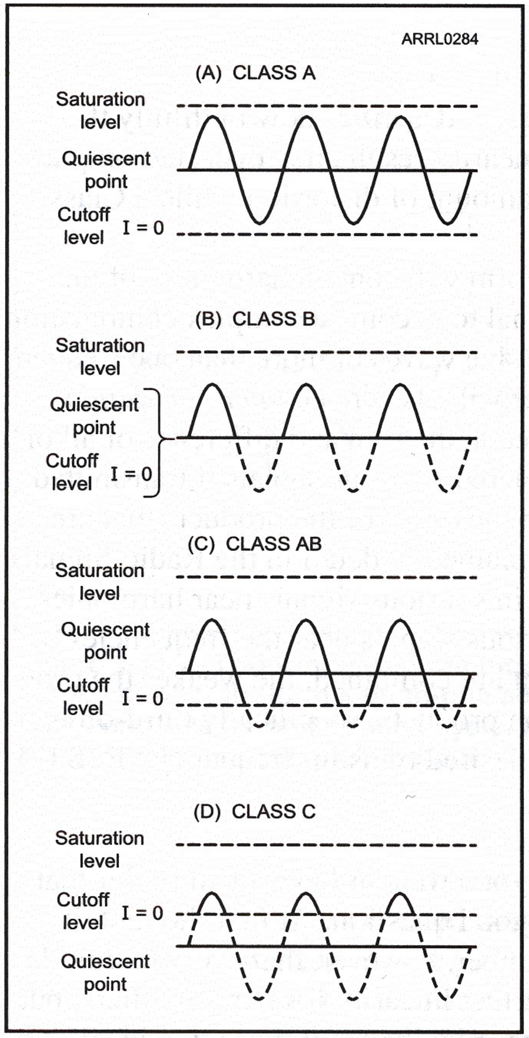


Figure 6-10 — Amplifier output waveforms for various classes of operation. All waveforms assume a sine-wave input signal.

Switching or Switchmode Classes (Classes D,E . . .)

Goes beyond Class C, causing the transistor to act entirely as a switch completely saturated, or completely cutoff.

- Switching speeds well above highest frequency to be amplified.
- Efficiency >90%
- The switching action creates an output waveform that is a series of squared-off pulses very rich in harmonics.
- For Class D Amp, a low pass filter at the output reduces most harmonics, but leaves the signals in the desired range of frequencies.

- **Why are switching amplifiers more efficient than linear amplifiers?**
  → The switching device is at saturation or cutoff most of the time

- **What is a Class D amplifier: →A type of amplifier that uses switching technology to achieve high efficiency.**

- **What circuit is required at the output of an RF switching amplifier?**
  →A filter to remove harmonic content

Introduction to Distortion & Intermodulation

Distortion

- Non-linearity causes the output to contain harmonics and distortion of the input.
- Intermodulation products are also created (intermod) and for a linear amplifier are considered spurious signals if transmitted.
- Even-order products result in spurious signals near harmonics of the input signal
- Lower order odd products - 3rd order products -result in spurious signals **near** the input signals.

Break ?

6.2 - Signal Processing

Pg: 6-13

Circuits useful for radios:

- Oscillators Frequency Synthesis
- Instability , Neutralization
- Mixers , Modulators
- VFOs & crystal oscillators
- SSB, Frequency and Phase Modulation
- Detectors & Demodulators Frequency Synthesis
- Phase-locked loop
- Direct Digital Synthesis

Introduction

Oscillators

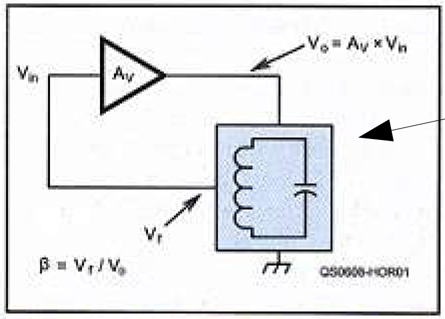


Figure 6.12 — An oscillator consists of an amplifier plus a feedback network, shown here as a parallel LC circuit. The LC circuit acts as a filter, restricting feedback to its resonant frequency.

Parallel LC circuit acts as a filter to restrict feedback to it's resonant frequency

In amplifiers, it was a problem if some output made its way back to the input creating positive feedback. The circuit instability turned the Amp into an Oscillator.

When we want to generate a signal – without any input, oscillation is just what we ! need!

To create an oscillator – 3 things are needed:

- Amplifier with gain at desired frequency
- Circuit that provides positive feedback
- Filter that restricts the feedback to desired frequency

Signal Processing 6.2

Instability and Parasitic Oscillation

Amplifier instability:

- Positive feedback combined with gain and filtering can cause an amplifier to become unstable.
- Can oscillate or generate noise.

We've heard Public Address amps screech at times.

→ Sound from speaker finds a way back into the microphone.

Called: Positive Feedback – the more speaker sound into the mic the more it becomes amplified.

– If it becomes a loop – oscillation may even occur.

Modern microprocessor based Audio feedback eliminators:
Audio device or software that automatically identifies and cancels problematic frequencies causing microphone feedback using digital signal processing to insert narrow EQ notches to regain volume and clarity without degrading the overall sound quality

Signal Processing 6.2

Pg: 6-13
Pg: 6-14

Instability and Parasitic Oscillation

Solutions for: RF Amplifier instability and Parasitic oscillation

- Use **negative feedback** to stabilize for parasitic suppression.
- Use Neutralization: Provides out-of-phase or negative feedback energy through an alternate path back to the input.

Parasitic Oscillations draw their power from the circuit:

Can occur on frequencies that have no relation to intended signal.

- Are caused by resonances in the input or output circuits
- Can absorb power from the signal to be amplified
- Solutions involves interrupting the path with parasitic suppressors which cause high impedance to the parasitic energy. (C1 in Fig 6.13)

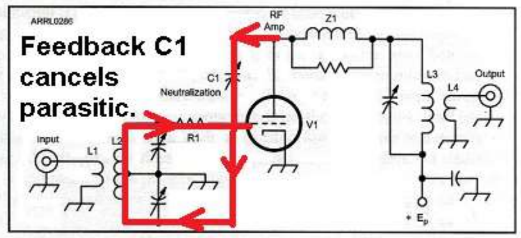


Figure 6.13 — Examples of neutralization and parasitic suppression techniques used with a tube-type RF amplifier.

What can be done to prevent unwanted oscillations in an RF power amplifier? ►Install parasitic suppressors and/or neutralize the stage

Signal Processing 6.2

RF Oscillators

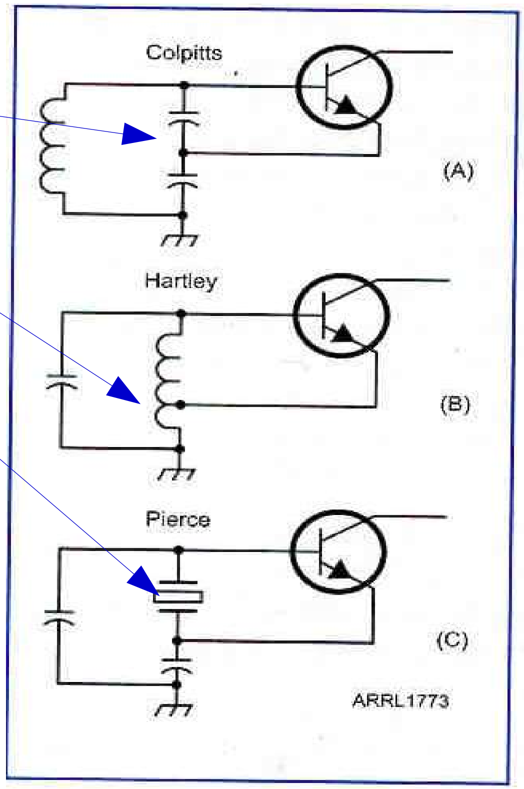
Positive feedback has come about in 3 basic circuits:
In each, feedback is created by routing part of the emitter circuit through a voltage divider created by two reactances using a tap.

Colpitts oscillator. Tapped capacitance divider

Hartley oscillator. Tapped inductance divider.

Pierce Crystal oscillator

To increase the stability over the Colpitts, or Hartley oscillator, Pierce used a crystal (which oscillates at essentially only one frequency and removes the need for an exactly tuned LC tank circuit).

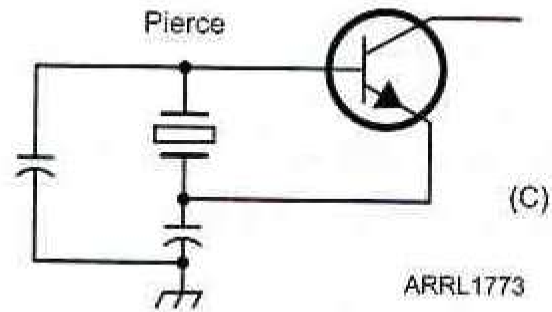


- **What are three common oscillator circuits?** ► **Colpitts, Hartley, and Pierce**
- **How is positive feedback supplied in a colpitts oscillator?** ► **Through a capacitive divider.**
- **How is positive feedback supplied in a Pierce oscillator?** ► **Through a quartz crystal.**

Signal Processing 6.2

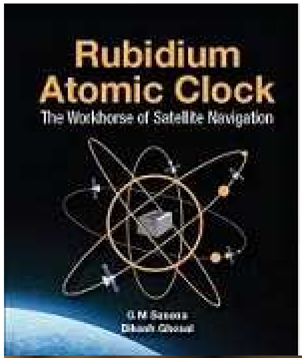
RF Oscillators

Crystals aren't stable enough for direct use at microwave frequencies. Stable frequency reference from GPS satellites signals can be used. Other sources like atomic (rubidium), or temperature stabilized high Q dielectric resonators are used in laboratory test equipment.



From low KHZ to 100s of MHZ: RF, watches @32.768 kHz digital wireless systems.

Rubidium clocks output frequencies derived from the precise 6.834 GHz transition of Rubidium-87 atoms. They typically correct a quartz oscillator to provide common, stable outputs like 10 MHz, 5 MHz & 1 MHz.
The internal atomic frequency is the reference, while the practical output is a much lower, frequency-divided signal.



Which of the following is a technique for providing highly accurate and stable oscillators needed for microwave transmission and reception? → All these choices are correct:

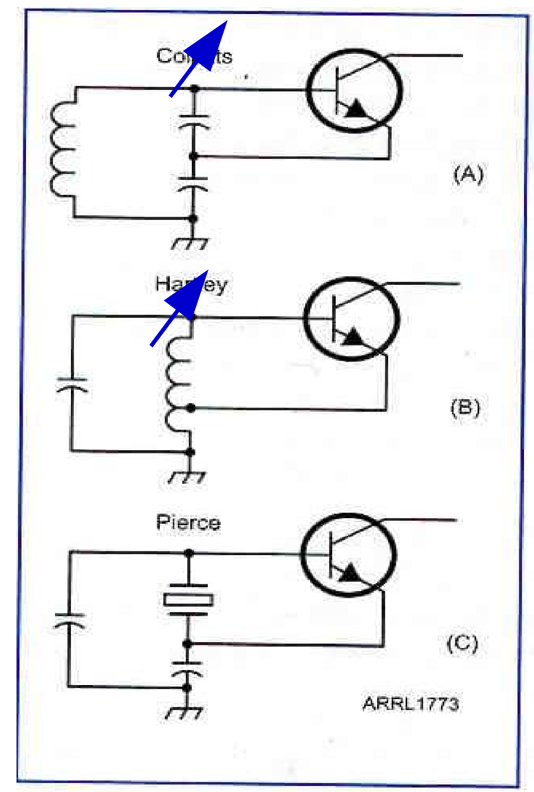
- **Use a GPS signal reference**
- **Use a rubidium stabilized reference oscillator**
- **Use a temperature-controlled high Q dielectric resonator**

Signal Processing 6.2

VFOs: Variable Frequency oscillators

Amateurs need to tune over a wide frequency ranges which requires
A variable frequency oscillator VFO.

= Component made to be variable to get a VFO.



VFOs:

- Are created using a ***variable*** component in the oscillator
- Not as stable as a crystal controlled oscillator.
- Both Hartley and Colpitts oscillators can be used
  - Use either:
    - A variable capacitor in parallel with inductor
    - A variable inductor in parallel with capacitor

- VFO Trade-Offs:
  - Not as stable as crystal oscillators
  - Susceptible to frequency changes due to heat and
  - vibration affecting inductor winding spacings and capacitor plate spacing capacitance.

Signal Processing 6.2

Crystals for Oscillators

Quartz Crystal:

- Is a natural piezoelectric material with ability to change mechanical energy (deformation) into electrical energy an vise versa. This is known as the piezoelectric effect.
- Can be sliced into plates to give vibration frequencies of a few HZ to 10s of MHZ.
- Valuable because it has very HI Q – meaning the vibration is very stable and precise.
- Its properties are very similar to those of a tuned electrical circuit as represented in Fig 6.15.

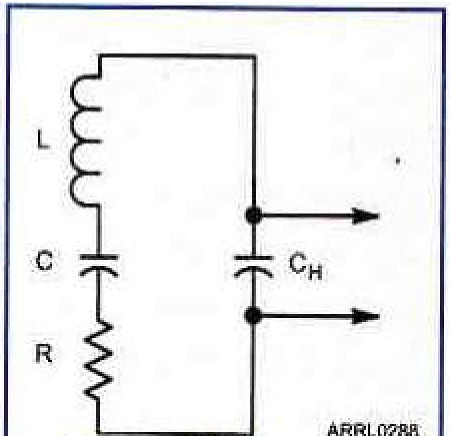


Figure 6.15

Quartz Crystal **Equivalent to series resonant circuit.**

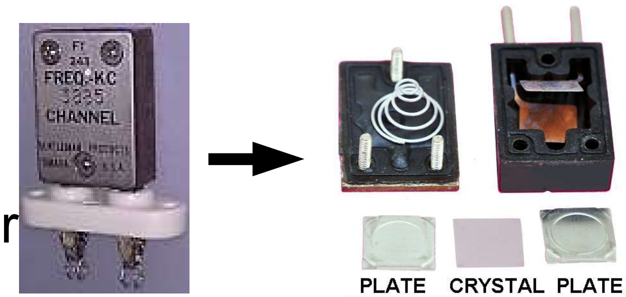
- **What is piezoelectricity?**
  ►A characteristic of materials that generate a voltage when stressed and that flex when a voltage is applied.
- **What is the equivalent circuit of a quartz crystal?**
  ►A series resonant circuit with capacitance of the holder.
- **Which of the following is an aspect of the piezoelectric effect?**
  ► Mechanical deformation of material due to the application of a voltage.

Signal Processing 6.2

Crystals for Oscillators

- Manufactures house them in various kinds of holders.

- The 2 metal plates that sandwich the crystal in it's holder from a capacitor and is part of the units design.



- The Mfr specifies additional external capacitance that causes it to resonate at its intended frequency.
- The crystal combined with all of the capacitance forms an equivalent to series resonant circuit.

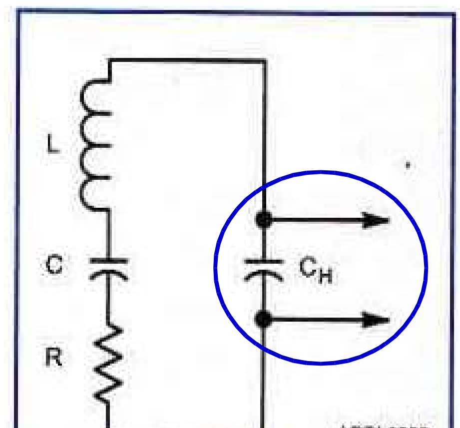


Figure 6.15

Quartz Crystal Equivalent to series resonant circuit.

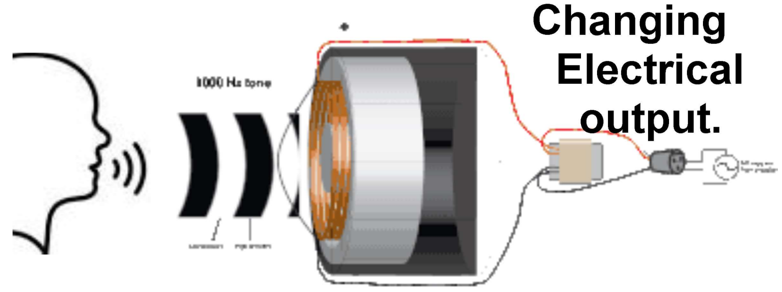
- **Which of the following ensures that a crystal oscillator operates on the frequency specified by the crystal manufacturer?**

  → **Provide the crystal with a specified parallel capacitance**

Microphonics and Thermal Drift

Microphonics and Thermal Drift:

Microphone coil vibrated by sound or tapping



Changing Electrical output.

- Physical change in an oscillator will change the feedback path.
- Two primary sources: mechanical vibration and temperature change.
- The response is called *microphonics* & causes the oscillator to act like a microphone converting sound waves(vibration) into an electrical signal.
- Designers isolate oscillators mechanically with padding & shock absorbers etc.

- **Microphonics are changes in oscillator frequency caused by mechanical vibration**
- **An oscillator's microphonics can be reduced by mechanically isolating the oscillator circuitry from its enclosure.**
- **Which of the following components can be used to reduce thermal drift in crystal oscillators? → NPØ capacitors**

Introduction to Frequency Synthesis

Advances in technology have overcome many of the traditional analog LC component VFO problems we have just studied:

Old Technology Problems →

- Oscillator instability
- Noise
- lack of accuracy
- Lack of repeatability
- Little flexibility

← Not ready for modern GHZ freq.

- WiFi
- Cell Phones
- Satellites (etc.)
- SDR
  (software defined radios)

- Modern radios no longer use <u>tunable</u> oscillators to control signal frequency. **Phase-locked loop (PLL) synthesizers** were once universal but now are replaced with improved :**<u>Direct Digital Synthesizers</u> (DDS)**
- **<u>Frequency Synthesis</u> : DDS controls the frequencies in small steps of 100 Hz or less.**
- Made possible by:
  - Microchip (μchip) development:
  - faster CMOS processing
  - Improved Digital-to-Analog Converters (DACs)
  - Programmable -Read-Only Memory (PROM) --<u>Lookup Tables</u>

Signal Processing

Introduction to Frequency Synthesis

Preview and summary
Of Frequency Synthesis:

- A crystal provides the fundamental accuracy at 1 stable frequency.

- A microprocessor controlled PLL uses the stable crystal frequency and creates, multiplied frequencies

- The result is:
  Direct Digital Synthesizer (DDS)
- Offers amazing digital frequency agility.

- All three work together in modern signal generation systems.

Phase:

Phase of the moon.

Signal Processing

Frequency Synthesis

Direct Digital Synthesizers (DDS)

- The DDS is based on using a sine wave by specifying a series of amplitude values spaced at equal phase angle.
- The frequency of the sine wave is determined by the sampling rate the synthesizer steps through the values.
- The crystal oscillator sets the sampling rate for amplitude values.
- The clock tells the *phase accumulator* to read the data from the adder.
- The phase accumulator varies between 0 & 360 degrees for one cycle of a sine wave.
- **The ROM lookup table contains the amplitude values for the sine(or cosine) of each angle.**
- The DAC changes the digital look up table values into analog output voltage representing the sine wave.

- **What type of frequency synthesizer circuit uses a phase accumulator, lookup table, digital to analog converter, and a low pass anti-alias filter: A direct digital synthesizer.**
- **What information is contained in the lookup table of a direct digital frequency synthesizer? The amplitude values that represent a sine-wave output.**

Signal Processing

Frequency Synthesis

Direct Digital Synthesizers (DDS)

A Synthesizer is run by a microprocessor, and other components.
It steps through a series of successive operations to create a wave.
It accesses stored values of the wave from memory.
Not all values of the wave are usually needed – so only samples from the memory are used.

Figure 6.16 Summary of functions:

Xtal Oscillator: Sets beat or basic timing steps

Adder keeps track of how many samples have been used

Phase Accumulator, Builds the wave amplitude from the samples

Rom Lookup Table, Contains all the possible samples of the stored wave values

Digital to Analog converter (D toA) Converts the digital numbered data to actual voltages

Low Pass Filter which Removes the noise generated by the digital switching that went on.

Next screen shows: The DDS running through the steps multiple times to create the wave:

Signal Processing

Frequency Synthesis

Direct Digital Synthesizers (DDS)

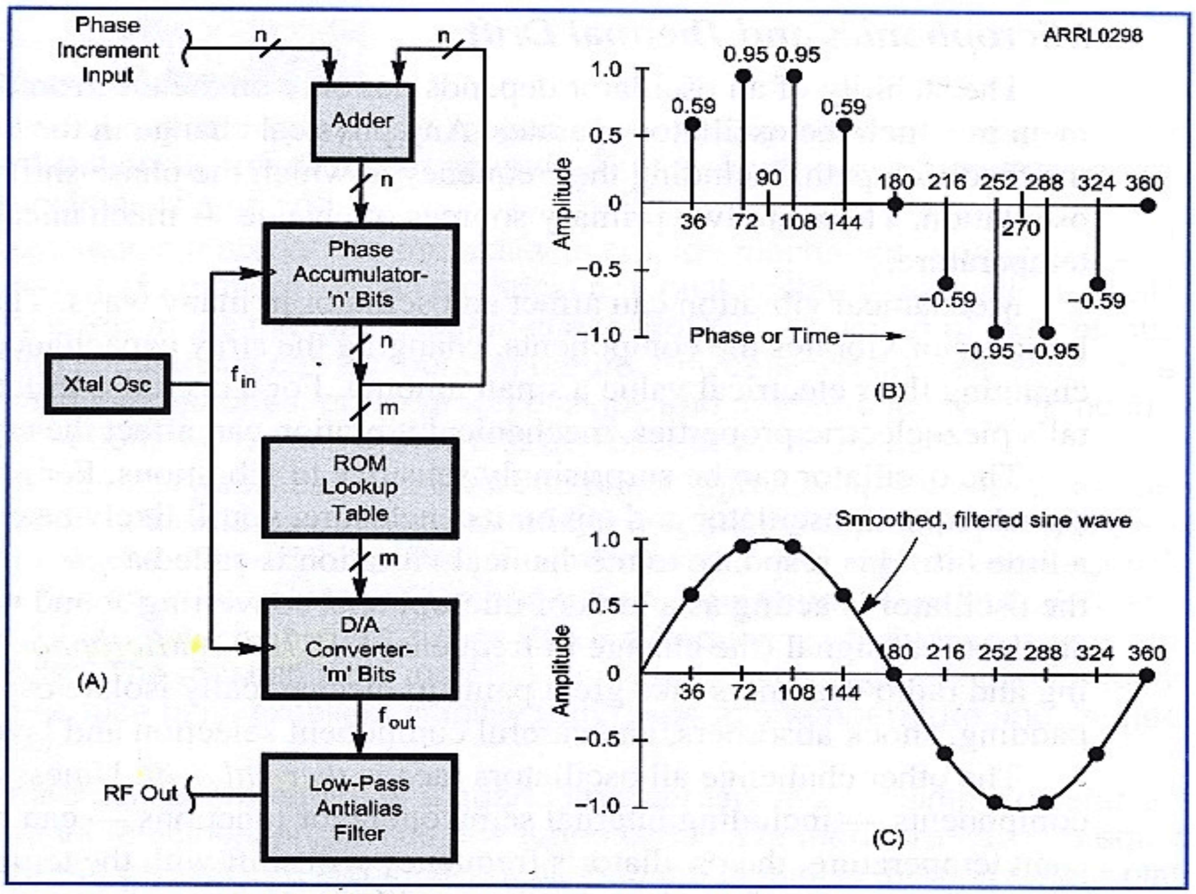


Figure 6.16 — (A) is the block diagram of a direct digital synthesizer (DDS). (B) shows the amplitude values found in the ROM lookup table for a particular sine wave being generated. (C) shows the smoothed output signal from the DDS, after it goes through the low-pass anti-alias filter.

Direct Digital Synthesizers (DDS)

- **What type of frequency synthesizer circuit uses a phase accumulator, lookup table, digital to analog converter, and a low pass anti-alias filter: A direct digital synthesizer.**

- **What information is contained in the lookup table of a direct digital frequency synthesizer? The amplitude values that represent a sine-wave output.**

Signal Processing

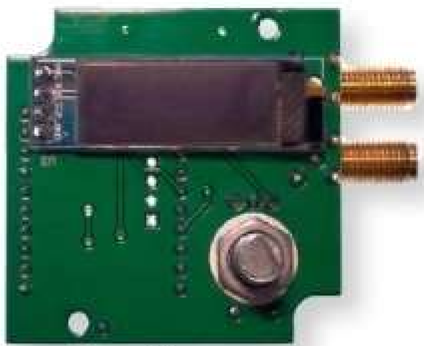
Frequency Synthesis

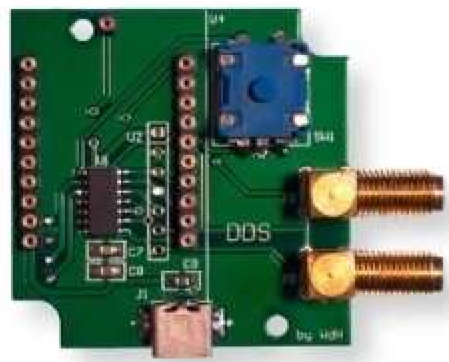
Direct Digital Synthesizers (DDS)

What do DDS Circuits Look Like?

They're not costly to build

A Simple DDS Signal Generator: 0MHZ to 70MHZ From: elektor magazine





DDS Signal Generator
0-70MHZ

OLED display
$4.00 at:
Buy Diplay

PIC18F04Q41
$1.03 at:
Mouser

$26.27 at:
on-bay

AD9851 DDS Module

Signal Processing
Frequency Synthesis
Direct Digital Synthesizers (DDS)

SPURS:

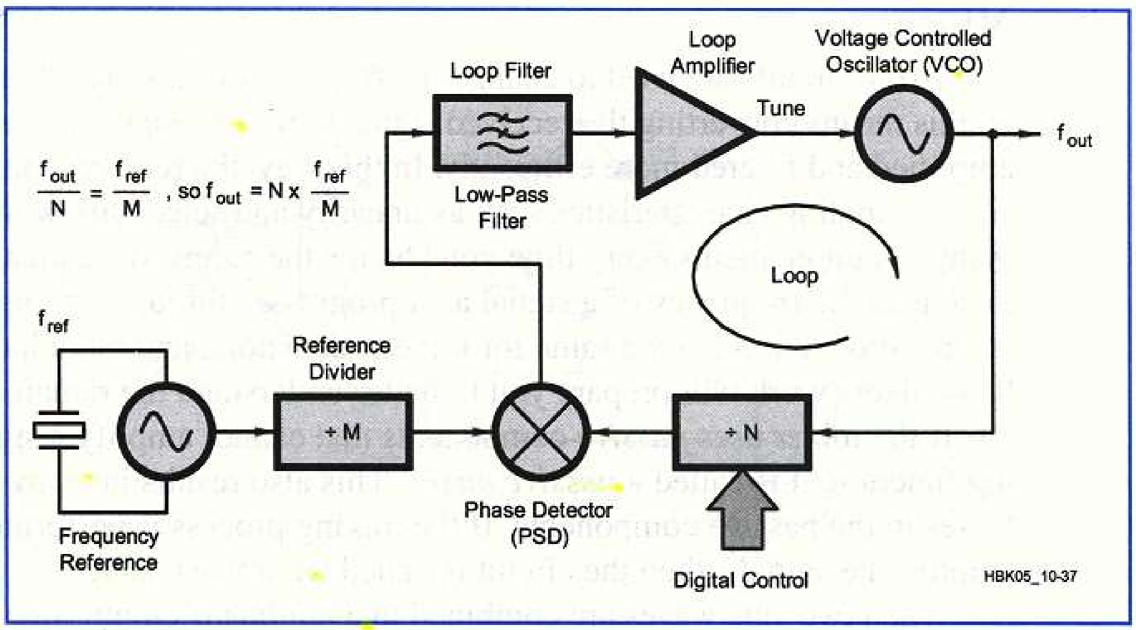
- There are spectral impurity components produced by a DDS (due to rapid switching)
- Called : *Spurs* (**S**purious **U**nwanted **S**ignals)
- Occur at specific discrete frequencies determined by the clock frequency and other digital components of the DDS Synthesizer.
- Careful design usually places these spurs outside of the amateur bands.

- **What are the major spectral impurity components of direct digital synthesizers?**
  Spurious signals at discrete frequencies.

Phase-Locked Loop (PLL)

<u>PLL</u> Phase Locked Loops are now older technology

- Were once universal
- PLL Have been largely replaced by DDS
- DDS requires less analog circuitry
- DDS is easier to integrate into digital ICs



Signal Processing

Frequency Synthesis

Phase-Locked Loop (PLL)

PLL

- A PLL frequency of the variable oscillator is continuously compared to the phase of a stable fixed-frequency reference oscillator
- Servo loop: Error-detecting circuit with negative feedback.
- Allows VFO to have stability of crystal oscillator.
- Can do FM modulation & demodulation when signal added to VCO.
- Capture range – Range of frequencies over which PLL can achieve lock.
- Spectral impurities are mainly broadband phase noise.

- **A phase-locked loop circuit is an electronic servo loop consisting of a phase detector, a low-pass filter, a voltage-controlled oscillator, and a stable reference oscillator**

- **Which of these functions can be performed by a phase-locked loop: Frequency synthesis and FM demodulation.**

Introduction to Mixers

Mixer circuits are used to <u>change the frequency of a signal</u>.

Example:

Superheterodyne receiver:

- Converting received signal to the *intermediate frequency.*(IF)
- So it can be amplified and filtered more efficiently
  (<u>a lower and easier to use frequency</u>)
- Several mixers are used to change the frequency of a signal as it progresses through a transmitter or receiver.
- Lots of necessary filtering is more easy performed at the IF frequency (or frequencies).

Introduction to Mixers

The Process of Mixing:

When two sine waves are combined in a nonlinear circuit like a mixer

- The output is a complex waveform
  that has the principal components of:
    - The two original signals frequencies
    - AND the two product signals.

- The product signals are sine waves whose frequencies are:
    - sum and difference of the two original signals.
    - Also included are are higher orders of combinations of the harmonics from the input signals.

Signal Processing
Mixers
Introduction to Mixers

(more complex explanation)

- Used to change the frequency of a signal.
- Mathematically combine 2 frequencies together, generating 4 output frequencies.
  - 1st: **f1** 2nd **f2** 3rd **f1+f2**, 4th **f1–f2**

Note : Two signals combined / mixed.
RF Signal
Local Oscillator

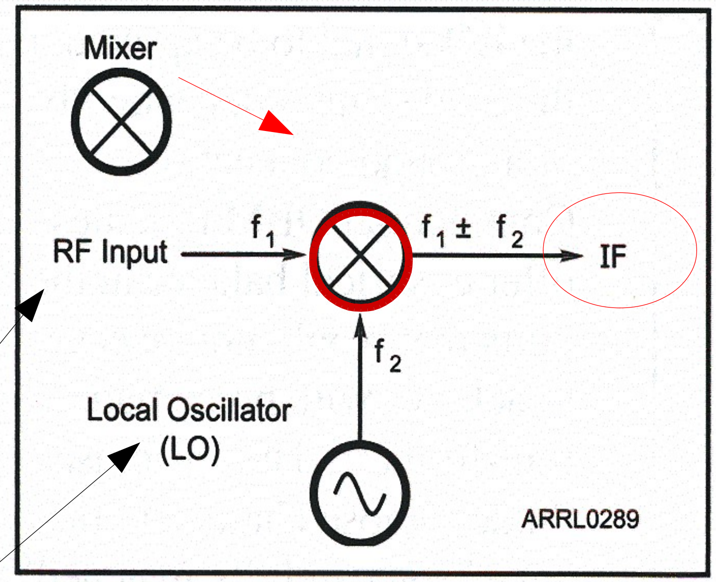


Figure 6.18 — The mixer combines the two input signals (RF and LO). This produces the mixing products that compose the IF signal at the output.

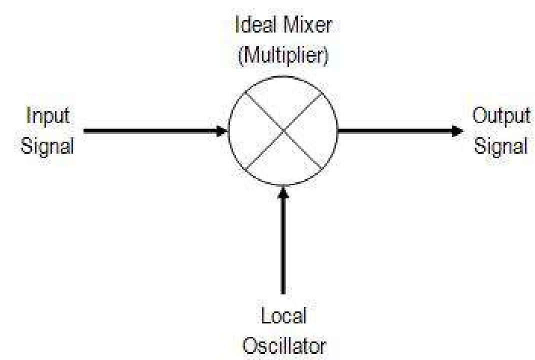
- **What are the principal frequencies that appear at the output of a mixer circuit?**

→ **The two input frequencies along with their sum and difference frequencies**

Passive Mixers

Passive mixers.

- Uses passive components such as diodes.
- Multiplies the RF signal and LO signals together
- No amplification.
- Some conversion loss.
- Requires strong LO signal.
- Generates some noise.
- Fig 6.19



Signal Processing
Mixers
Active Mixers

Active mixers.

- Uses active components such as transistors or FETs.
- Amplification possible.
- No conversion loss.
- Less LO signal needed.
- Generate less noise.
- Strong signal handling capability is not as good as passive mixers.
- Fig 6.20
- Mixer losses : Amplifiers gain should just overcome losses.
- Excessive amplification allows strong signals to cause IMD
- (InterModulation Distortion) or spurious mixer products.

- **What occurs when an excessive amount of signal energy reaches a mixer circuit?**

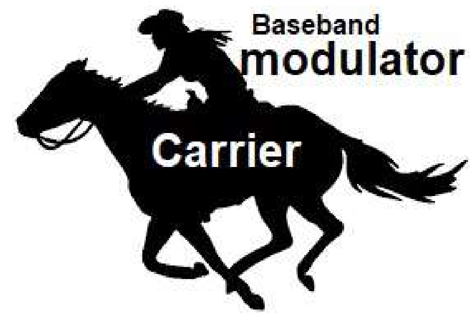
→ **Spurious mixer products are generated.**

Signal Processing

Modulators

Recall your knowledge of Modulators

Modulators:



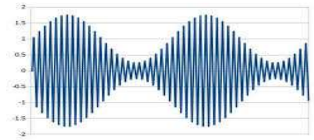
Combining a voice (rider) with an RF signal "carrier" (the horse) results in a signal that can be ***transmitted.***

- The process of joining these two signals is called **modulation**.
- The information voice signal is also known as the "**baseband**" **signal**.
- The baseband signal "modulates" the radio frequency carrier signal.

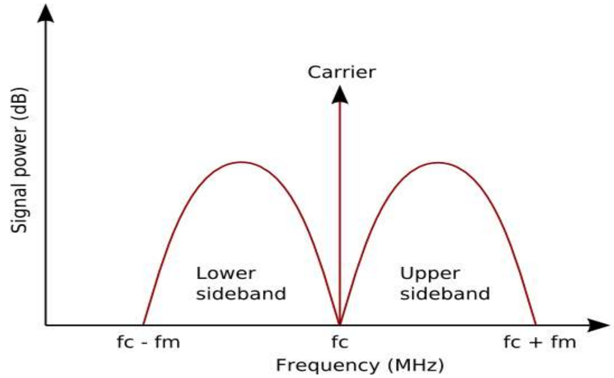
- **What is meant by the term base band in radio communications?**

→ **The frequency range occupied by a message signal prior to modulation**

Amplitude modulation.



- Mixing an audio frequency(AF) signal 100's -1000's of HZ and a carrier, produces amplitude **modulated signal**.



- AM is a composite of **3** individual RF signals.
- RF carrier (carries no information)
- Lower sideband (Left sideband)
- Upper sideband (Right sideband)
  - **The sidebands are images of each other with the same information.**

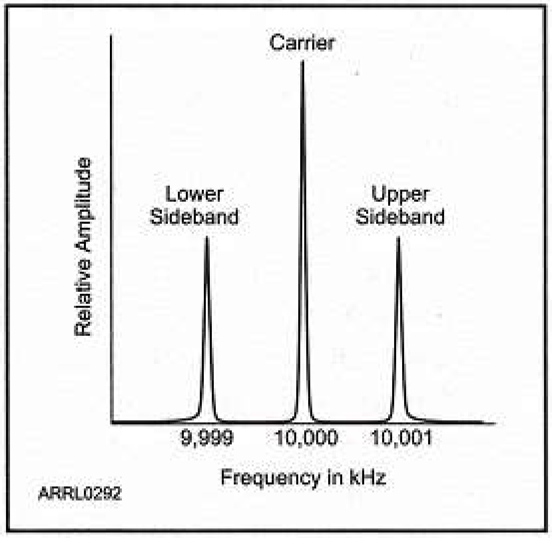


Figure 6-20 — The result of amplitude modulating a 10 MHz carrier with a 1 kHz sine wave shows the upper sideband (USB) at 10 MHz + 1 kHz and \the lower sideband (LSB) at 10 MHz — 1 kHz.

Signal Processing
Modulators
SSB: The Filter Method

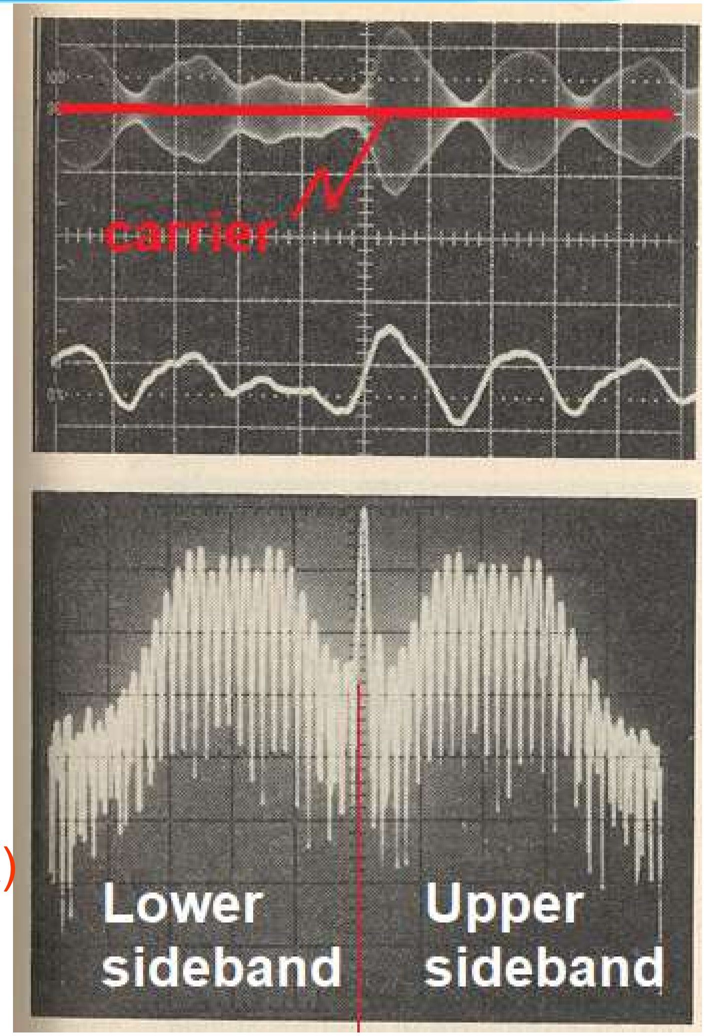
(Ways to increase signal intelligibility)

(not in book)

Single-sideband modulation.

- Since the carrier doesn't contain "information"
- And the sideband of the AM signal are mirror images (match each other)
- Energy can be saved by eliminating both of them
- Various methods –
  - filtering is used a lot.

(not in book)



Note how the lower and upper sidebands are Mirror images of each other.

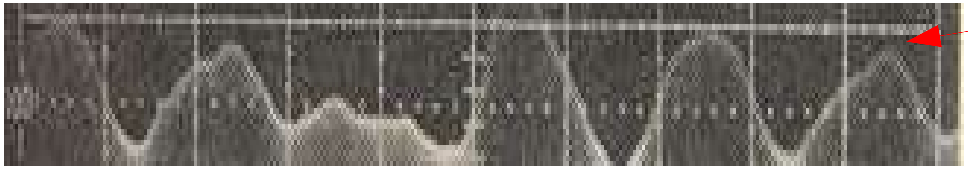
Signal Processing

Modulators

SSB: The Filter Method

Single-sideband modulation: Several ways to create it.

- Filter method.
  - Start with AM double-side band signal & use filter**s** to **<u>remove one sideband & the carrier</u>**.
  - Better idea – use a balanced modulator (mixer) to generate a double-side band <u>suppressed carrier</u> signal. Then all you have to <u>filter out is the the one unwanted sideband.</u>



Only 1 sideband is left

(not in book)

What is one way a single-sideband signal can be generated?

→ By using a balanced modulator followed by a filter.

Signal Processing

Modulators

SSB: The Filter Method

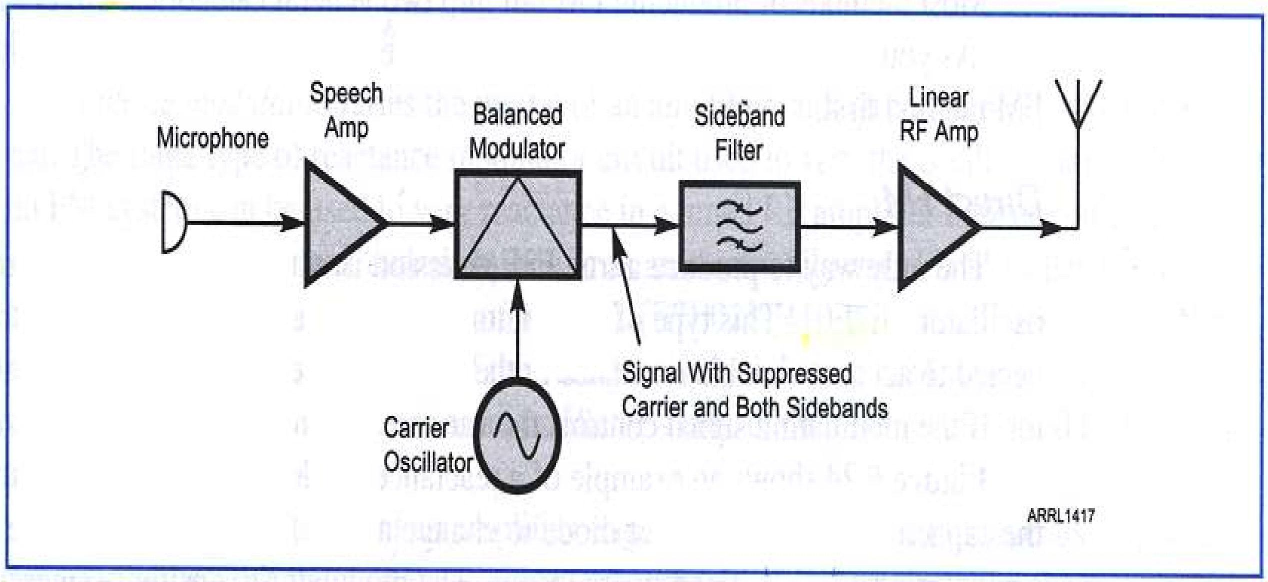


Figure 6.22 — A block diagram showing the filter method of generating and transmitting an SSB signal.

SSB: The Quadrature Method

(For More Information Section) - optional

- **<u>Phasing (quadrature) method.</u>**
  - Generate 2 identical carrier signals, 90° out of phase.
  - Generate 2 identical audio signals, 90° out of phase.
  - Mix these together in a pair of balanced modulators & result is that the carrier & one sideband are canceled out leaving only one sideband.
  - 90° phase shift of band of audio frequencies difficult to accomplish in hardware, but easy in software.
    - Called: Hilbert-transform filters.
  - Most SDR transmitters use the phasing or quadrature method to generate SSB mathematically.

SSB: The Quadrature Method

(For More Information Section) - optional

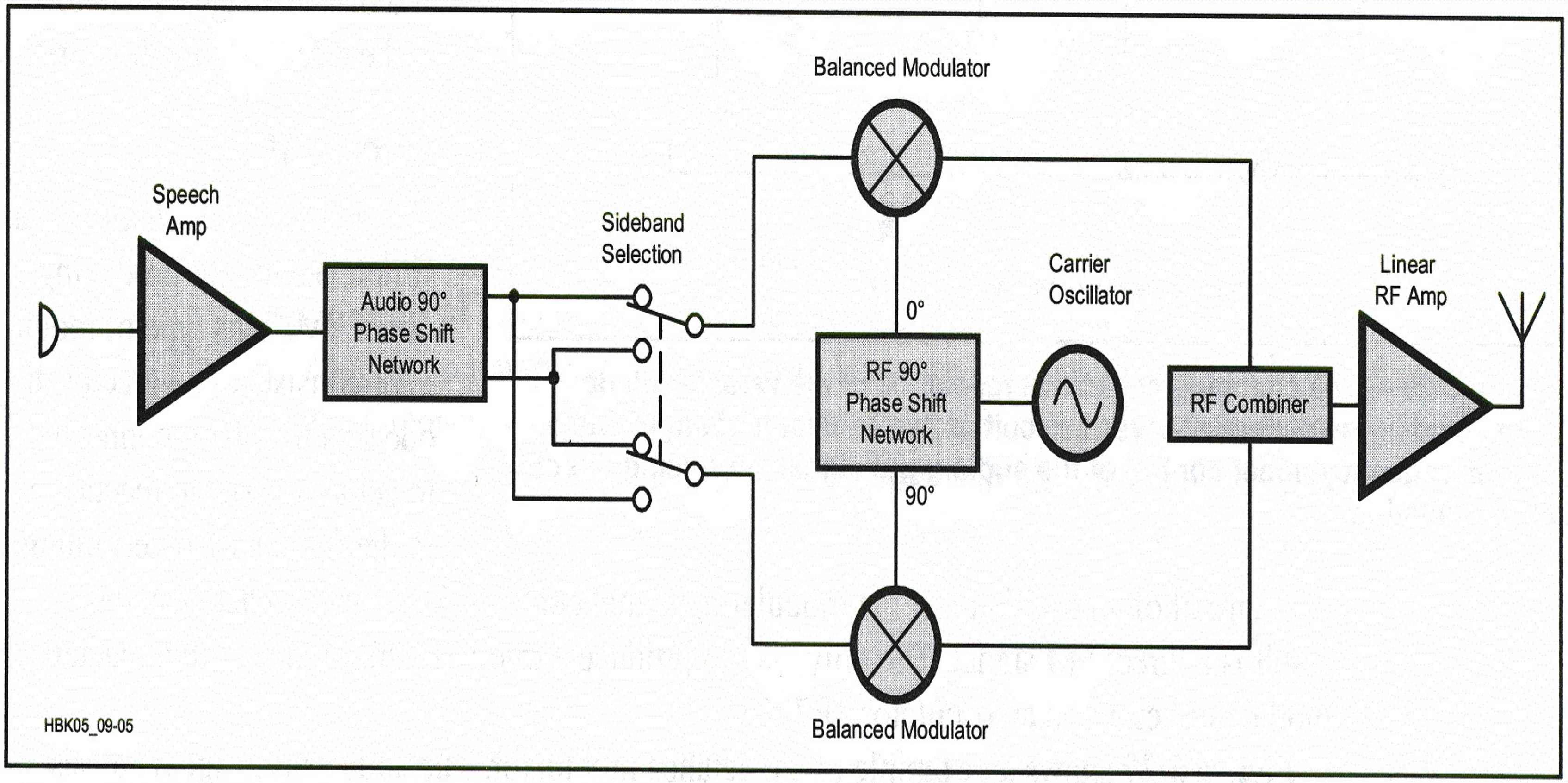


Figure 6-23 — The block diagram of a quadrature SSB generator. By combining the independent DSB signals, one sideband is canceled and an SSB signal results.

Frequency and Phase Modulation

Frequency and Phase modulation

- Frequency Modulation (FM) and Phase Modulation (PM) have some advantages over AM modulation.

  - Can be wider in footprint - hi fidelity.
    (perhaps a disadvantage as takes up more band space)

  - Most methods of producing FM fall into two categories:

    - **Direct FM**

    - **Indirect FM**

Frequency modulation.

- Direct FM
  - Only way to produce true FM (with no phase modulation) is with a *reactance modulator* acting **on** an oscillator.

Which of the following can be used to generate FM phone signals?

➤ **Reactance modulation of a local oscillator.**

Direct FM

- Direct FM by using a Varactor Diode

The audio input signal causes the capacitance of the varactor diode to change, and in turn, that changes the LC ratio of the oscillator's resonant circuit and its frequency. A variable inductance could also be used but is less practical.

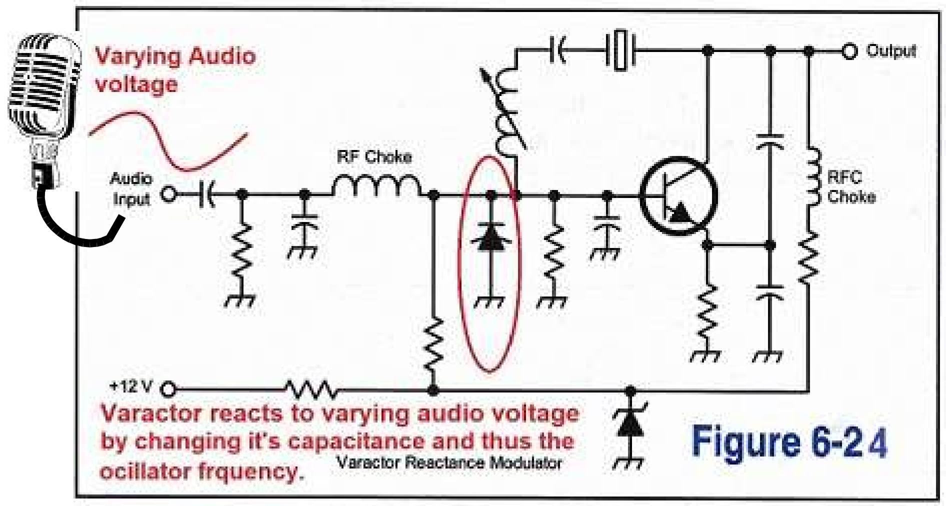


Figure 6-24

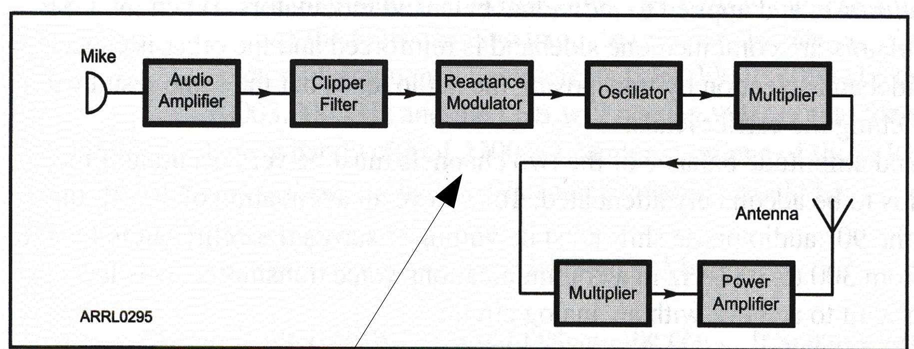


Figure 6-25 — The block diagram of a typical direct-FM transmitter. The amount of frequency deviation produced by the reactance modulator is multiplied along with frequency by the multiplier stages.

What is the function of a reactance modulator?

➤ Produce PM or FM signals by varying a capacitance

Indirect FM

- **Indirect FM (phase modulation).**
  - Frequency deviation increases with increasing modulating frequency.
  - Generated by adding reactance modulator
    to any stage *other than* the oscillator stage.

For *Indirect* FM and **Phase Modulation,** the amplifier tank circuit frequency is varied to produce a PM signal.

Signal Processing

Pre-emphasis and De-emphasis

Frequency and Phase modulation.

- For **PM**: Frequency deviation increases with the modulating audio frequency.
  (Because higher audio frequencies produce greater frequency deviation.)
- For **FM**: Frequency deviation is constant with the modulating audio frequency.
  - So for **FM**, this leaves the high audio frequencies (which are lower in energy) susceptible to high frequency noise.
  - In **FM, *pre-emphasis*** is applied to the high frequencies of the audio.
    - Pre-emphasis is amplifying the higher audio requencies more than the lower frequencies.
    - Adding pre-emphasis to the modulating audio signal of an FM transmitter gives it characteristics of a PM signal. Pre-emphasis yields a better signal-to-noise ratio for FM.
- At the receiver, corresponding **de-emphasis** is added.
- Net result is reduced high frequency noise for both PM and FM)
- **What is added to an FM speech channel to boost the higher audio frequencies? → A pre-emphasis network.**
- **Why is de-emphasis used in FM communications receivers?**
  → For compatibility with transmitters using phase modulation

Introduction to Detectors and Demodulators

We just learned about creating or **TRANSMITTING** AM, FM and PM signals.

The following topic is about **RECEIVING** these signal.

AM Detectors Amplitude Modulation

FM Detectors Frequency Modulation

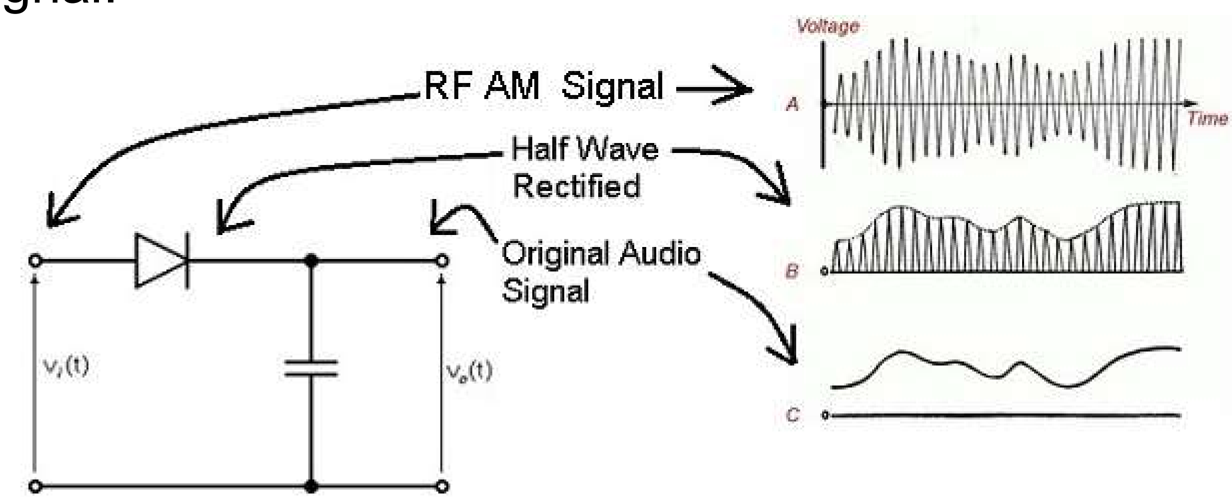
Signal Processing
Detectors and Demodulators
Detectors

Modulating information must be recovered from the received signal. Several methods:

- A **detector** circuit extracts the information directly from the signal.
- A **demodulator** reverses the transmitter modulation process to recover information.
- The simplest detector uses a diode to simply rectify the signal waveform
  - It passes only the positive half cycles. Then a capacitor charges and holds the peak voltages of each ½ cycle (now a filtered dc wave) leaving only the original ac information signal.

Diode Detector for AM:

- The simplest detector is the diode detector.
- It works on the signal amplitude as an " Envelope detector"



Simplified version of Fig. 6.26

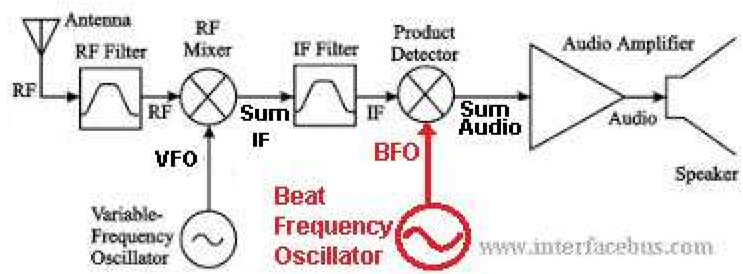
How does a diode envelope detector function?

→ **By rectification and filtering of RF signals**

Product Detectors

Product Detectors are used for CW, SSB and RTTY

- Mixes the received signal
- with a BEAT FREQUENCY OSCILLATOR
- to create an hearable audio signal.
  - $f_{RF} \times f_{LO} \rightarrow f_{RF} - f_{LO} \rightarrow f_{AF}$



- A Product Detector is a type of mixer that combines a locally generated carrier or beat-frequency oscillator(BFO) with the incoming RF signal
- It follows the IF stages.
- The BFO frequency is chosen so one of the "sum and difference" output products is at audio frequencies.

An example: If the receiver's IF is 455 kHz and a 700Hz tone is desired, the BFO is set to 455.7kHZ creating sum-and difference products at 700Hz and 910.7**k**Hz. An audio filter removes the higher 910.7**k**Hz frequency

Which type of detector is used for demodulating SSB signals?
→ **A Product detector.**

Signal Processing
Detectors and Demodulators
Product Detectors

Pg: 6-24
6-25

A Beat Frequency is created when two different frequency signals are combined. Their max and min creases a 3rd signal.

► Note what happens when the top two signal reach max or min peaks at the same time..

1st

2nd

New
3rd

Click the link BELOW to play this video
2 different frequencies (top & Middle waves) together create a 3rd wave or frequency.

THE NEW WAVE on the bottom, IS THE DIFFERENCE IN FREQUENCIES BETWEEN WAVE #1 AND WAVE #2.

Note how the TOPS of Wave #3 occure when Wave #1 and Wave #2 are both at their TOPS, and wave #3's BOTTOMS occure when Wave #1 and Wave #2 are both at their BOTTOMS.

Note how Wave #3 goes to 0 when Wave #1 or #2 is at it's TOP and the other is at it's BOTTOM.

Note that wave #3 is a lower frequency of 20 Hz (the difference of 420Hz - 400Hz).

Example of how a BFO (Beat Frequency Ocillator is used in analog HAM Radio receivers)
===>A 20Meter SSB signal at 14. MHz minus a BFO of 13.999200 MHZ reveals a human voice of 800HZ

Click the link below to view this video

https://drive.google.com/file/d/1bRhIVcgaMaHStNeGRX4t5hDt-BKfe3B1/view?usp=drive_link

Detecting FM:

Frequency Discriminator:

- The most common FM Detector is the frequency discriminator (Fig 6.27).
- It uses a transformer tuned to the IF to detect FM signals.
- Through transformer action, the primary signal is introduced the center tap of the secondary winding via capacitor. Modulation in the received signal causes a phase shift in the two output voltages. The two voltages are rectified by diodes, and their difference in voltage is the recovered audio signal.

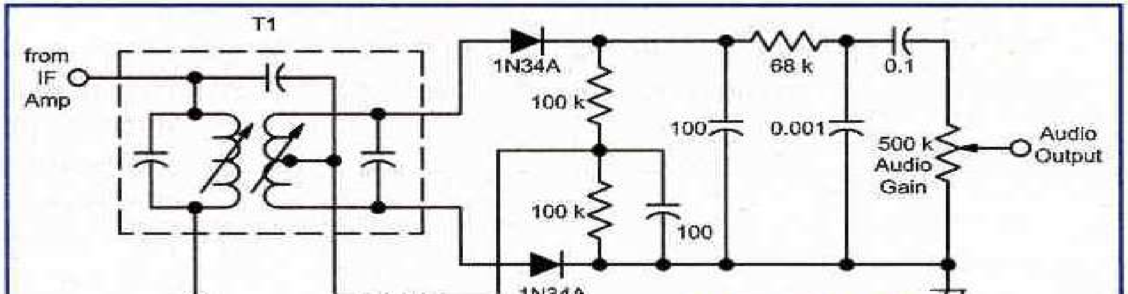


Figure 6.27

What is a frequency discriminator?
→A circuit for detecting FM signals.

Break ?

Pg:

6.3 Digital Signal Processing (DSP) and Software Defined Radio (SDR)

Pg: 6-26

Technology Marches On:

DSP has revolutionized our world the same way as the transitor and microprocessor did.

Kenwood TS-990S

Flex Radio 8600M

Most newer transceivers
- Use a lot of DSP
- Many companies offer full SDR

Icom IC-7300

Yaesu FT-DX10

Digital Signal Processing (DSP) Software Defined Radio (SDR) 6.3 Introduction

Will abbreviate to just “DSP” and “SDR” in follow on slides

- Digital Signal Processing (DSP)
  - Part of practically all modern transceivers.
  - Nearly all of the functions that were analog in older transceivers are now performed inside digital processors as mathematical functions.
  - Allows signal processing that is difficult or impossible to obtain using older analog methods.

- Procedure:
  - Convert analog signal to a series of numbers.
  - Process series of numbers mathematically with microprocessors.
  - Convert resulting series of numbers back to an analog signal.

Sequential Sampling

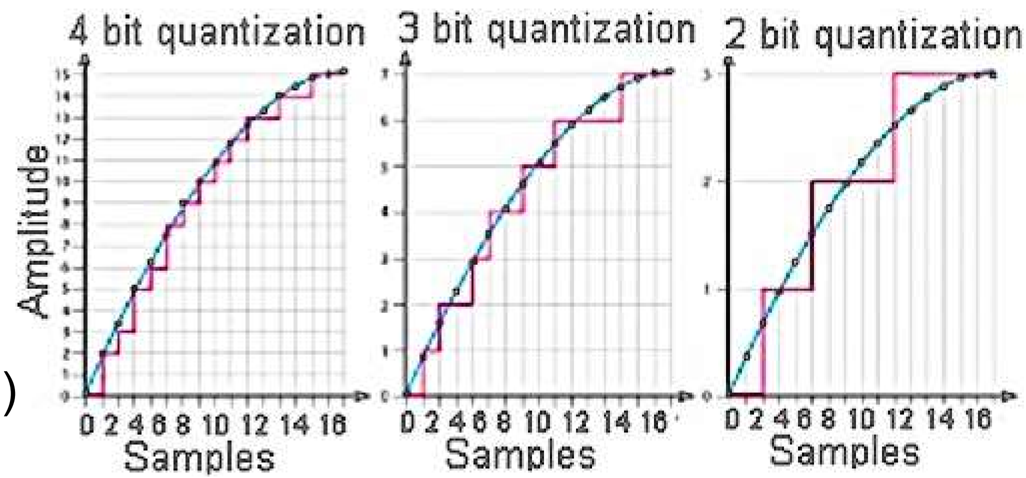
Sequential sampling

1st step in converting the <u>analog waveform</u> to a <u>digital representation</u>

<u>Analog-to-digital conversion.</u>

- **Sample** the signal at <u>regular intervals</u> (sequential sampling).
- Convert signal value to a number.
- Higher sampling rates yields higher accuracy (more bits).
- More “bits” in the number yields higher precision of the representation.

More bits
= Higher precision
(Better Representation)



Fewer bits
=Lower precision
(Poorer Representation)

Step 1 <u>Sample The Signal</u>
This shows how the original signal is better represented with more samples.

Sequential Sampling

Example of:
Adequate Sampling rate.

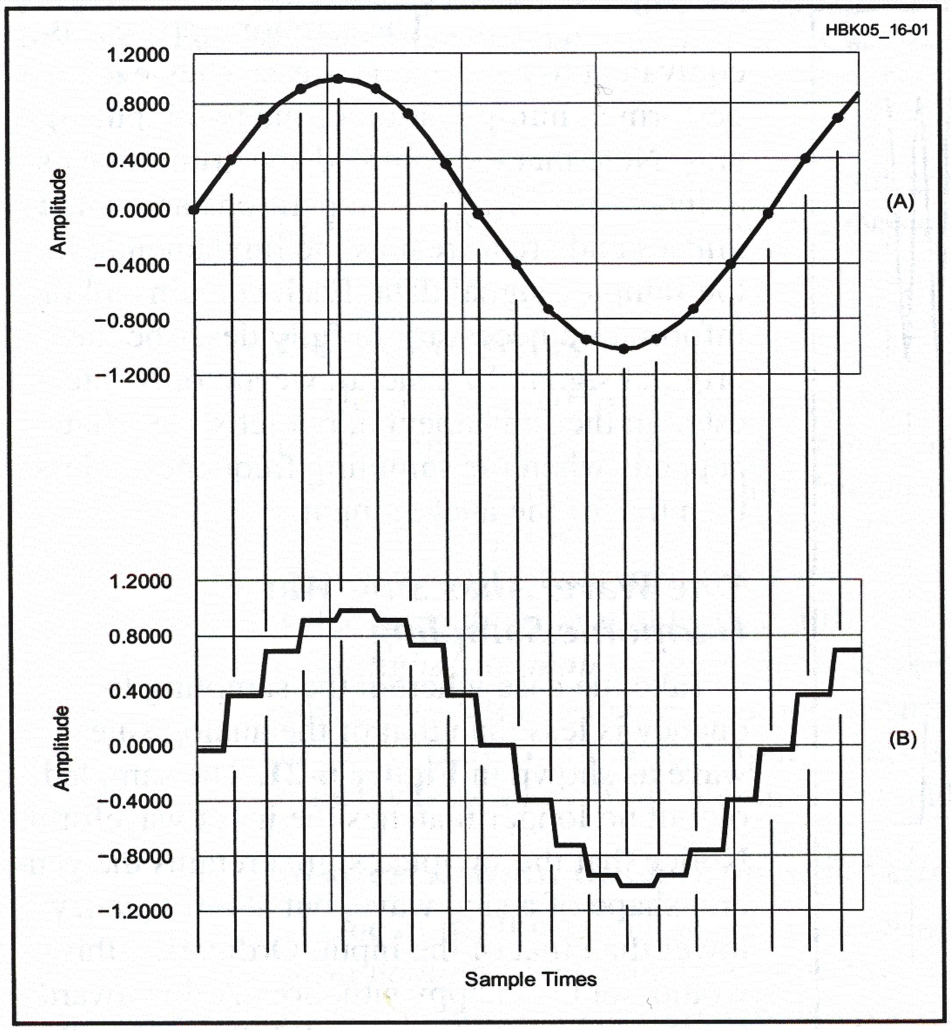


Figure 6-28 — Sine wave of frequency much less than the sampling frequency (A). The sampled sine wave (B).

Note that the frequency of the sine wave being sampled <u>is much lower</u> than the sampling frequency.

<u>Correct:</u>
<u>Many samples must taken during each cycle of the sine wave.</u>

Enough samples allows the signal to be <u>predicted and interpreted correctly</u> from the samples .

Sequential Sampling

- The analog sine wave's frequency spectrum is shown in Fig. 6.29A
  - Recall that the Frequency Spectrum is amplitude vs frequency.
- The spectrum of the sampling function is shown in Fig. 6.29B

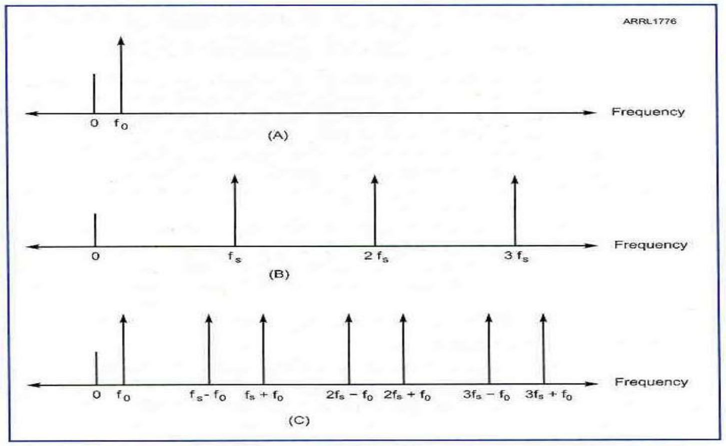


Figure 6.29 — Spectrum of an analog sine wave (A). The spectrum of a sampling function (B). The spectrum of the sampled sine wave (C).

- The sampling process is equivalent to a mixing process, both processes perform a multiplication of the two input signals.
- Note that the sampled spectrum repeats at intervals of f_s .
  These are called *Aliases.*

DSP and SDR

Digital Signal Processing (DSP)

Sine Wave, Alias Sine Wave

- If sample rate is less than 2X the highest frequency of the signal being sampled, result does not match input signal. (Nyquist Sampling Theorem)
- Retains general shape of sine wave but at a lower incorrect frequency.

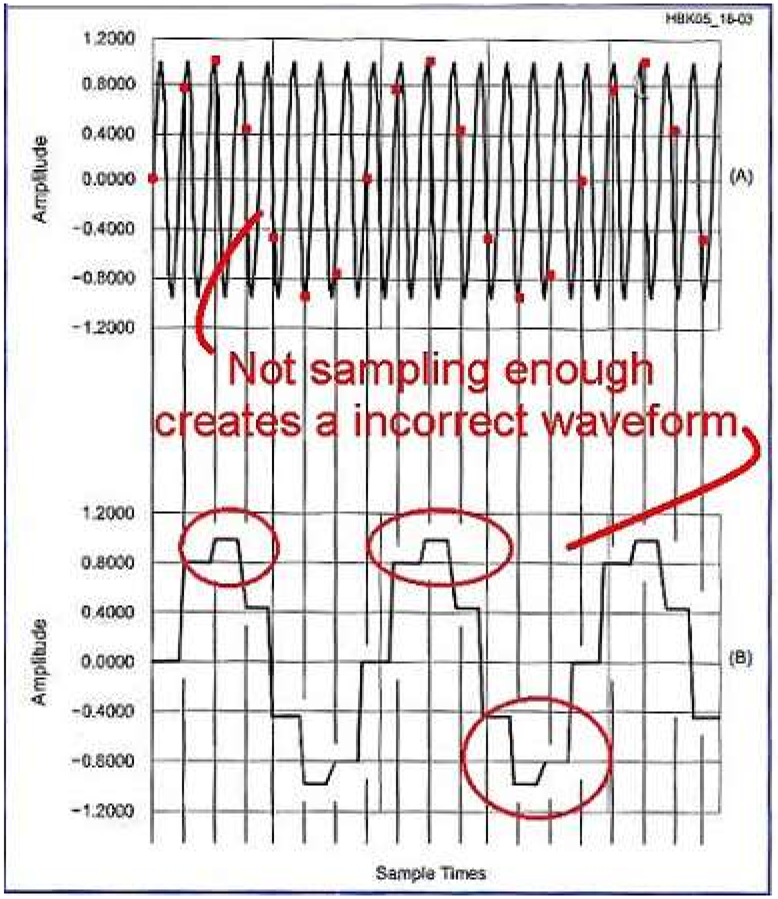


Fig 6.30

Shows in-sufficient sampling to adequately represent the wave and successfully create a digitized form of it.

What's wrong with it:

- wrong frequency
- clipped waveforms
- etc.

How frequently must an analog signal be sampled to be accurately reproduced?

► At least twice the rate of the highest frequency component of the signal.

Data Converters

Analog-To-Digital converter (ADC). Base 2

- A device that performs sampling and analog-to-digital conversion.
- Produces a binary number that is directly proportional to the value of the input voltage.
- More bits in binary number → higher resolution.
    - 8 bits → 256 steps. 2^8 (2 raised to the power of 8: 2x2x2x2x2x2x2x2=256)
    - 16 bits → 65,536 steps. 2^{16}
    - 24 bits → 16,777,216 steps. 2^{24}

- **How many different input levels can be encoded by an analog – to – digital converter with 8-bit resolution?**
  → **256**

Data Converters

ADC resolution

- Is determined by the reference voltage corresponding to the maximum digitized value and the number of bits representing each sample.

Example:
If a 10-bit ADC has a reference voltage of 1 V, the resolution is:
1V/2"=1/ 1024 = 0.976 mV = 1 mV.

- **What is the minimum number of bits required to sample a signal with a range of 1 volt at a resolution of 1 millivolt? ➝ 10 bits.**

- **What sets the minimum detectable signal level for a direct-sampling software defined receiver in the absence of atmospheric or thermal noise? ➝Reference voltage level and sample width in bits.**

Data Converters

ADC resolution (cont.)

- Converting the input signal to digital results in data close to the original signal.

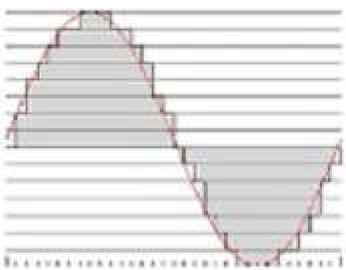
- Other effects include: the bandwidth and slew rate from the converter input circuits resulting in small amounts of distortion as total harmonic distortion or THD which is a measure of converter quality.

- **Which of the following is a measure of the quality of an analog-to-digital converter?**
  → Total harmonic distortion.

Data Converters

The other direction: Digital to Analog Conversion DAC:

- Returns binary numbers back into analog voltages(the reverse
- of ADC), but with some stair-step appearance as each voltage is created.
- The steps between values create unwanted harmonics that are removed with a low-pass filter also called a reconstruction filter.



Digital Stair Step

What is the purpose of a low-pass filter used at the output of a digital-to-analog converter?

→ **Remove spurious sampling artifacts from the output signal**

Fourier Transforms

Fourier Transforms:

Joseph Fourier (1768–1830) was a French mathematician and physicist.

- Is the software equivalent of a hardware spectrum analyzer.
- It converts the signal from Time Domain(amplitude vs time) to the Frequency Domain (amplitude vs frequency).
- This shows the spectral content of the signal
- Performed by DSP algorithm.
- An *Inverse Fourier Transforms* reverses the process
- FFT (*Fast Fourier Transform*) is a special algorithm that reduces the number of calculations required by a factor of 100.

- **The function of a Fast Fourier Transform is to convert digital signals from the time domain to the frequency domain.**

Decimation and Interpolation

The Amazing power of modern DSP
DSP Processors can perform functions on a signals that are impossible to do effectively in analog systems.

Example:
Changing the effective sample rate to shift the frequency of digital signals by ***Decimation*** – which removes every nth sample, reducing the effective sample rate by the same factor.

A similar process called ***Interpolation*** inserts new sample between existing samples to increase the effective sample rate. No anti-aliasing filter is required since the sampling rate is being increased not decreased.

- **What is the function of decimation?**
  ► **Reducing the effective sample rate by removing samples.**

- **Why is an anti-aliasing filter required in a decimator?**
  ► **It removes high-frequency signal components that would otherwise be reproduced as lower frequency components.**

DSP and SDR

Digital Signal Processing (DSP)

Software Defined Radio (SDR) Systems

Software-Defined Radio (SDR)

- **SDR** Software radio hardware is intended to handle almost any modulation format, signal bandwidth and frequency desired.
- Functionality may be altered by downloading new software.
- Uses DSP to perform radio signal processing.
- Hardware must be adequate to sample signals, perform math quickly enough. (keep up with the software)
- Variety of SDRs :
  - Some use an external PC & sound card or USB port
  - Some are self contained (one enclosure), with display and can drive a external LED panel.

DSP and SDR

Digital Signal Processing (DSP)

Software Defined Radio (SDR) Systems

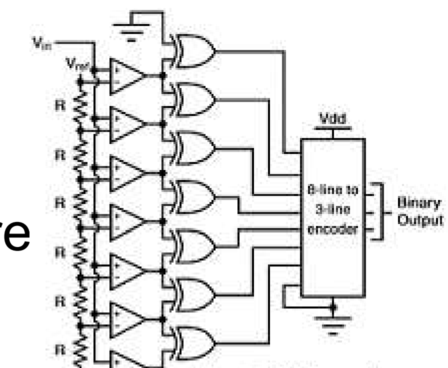
SDR Hardware.

- The transition between analog and digital signals can occur at any of several places in the signal chain between antenna and humans.
- Wide variety of equipment available.
- Some use a PC & sound card.
  - Here the digital signals are received and transmitted over the air using the same process as regular microphone audio.
  - All RF is processed in the transceiver analog supheterodyne or SDR.

- **What is meant by "direct sampling" in software defined radios?**
  - ► **Incoming RF is digitized by an analog-to-digital converter without being mixed with a local oscillator signal.**

SDR Hardware (cont.)

- In some the A-to-D is made at the IF stage and processed by on board DSP or sent to PC for modulation or demodulation.
- State Of The Art: Convert to RF right at the Xmit or Rcv frequency (Antenna) no analog mixers involved.
  Called: Direct Digital Conversion DDC see Fig 6.31
  DDC requires very high sample rate (speeds)
  - Called Flash Conversion ADC. (Pipeline architecture
    - High-speed, high-resolution
    - They are essential in the RF front-end, IF, or for direct conversion of RF signals.



Flash ADC Example Using Comparators and Exclusive OR gates

- **Why are direct or flash conversion analog-to-digital converters used for a software defined radio?**
  → **Very high speed allows digitizing high frequencies**

DSP and SDR

Digital Signal Processing (DSP)

Software Defined Radio (SDR) Systems

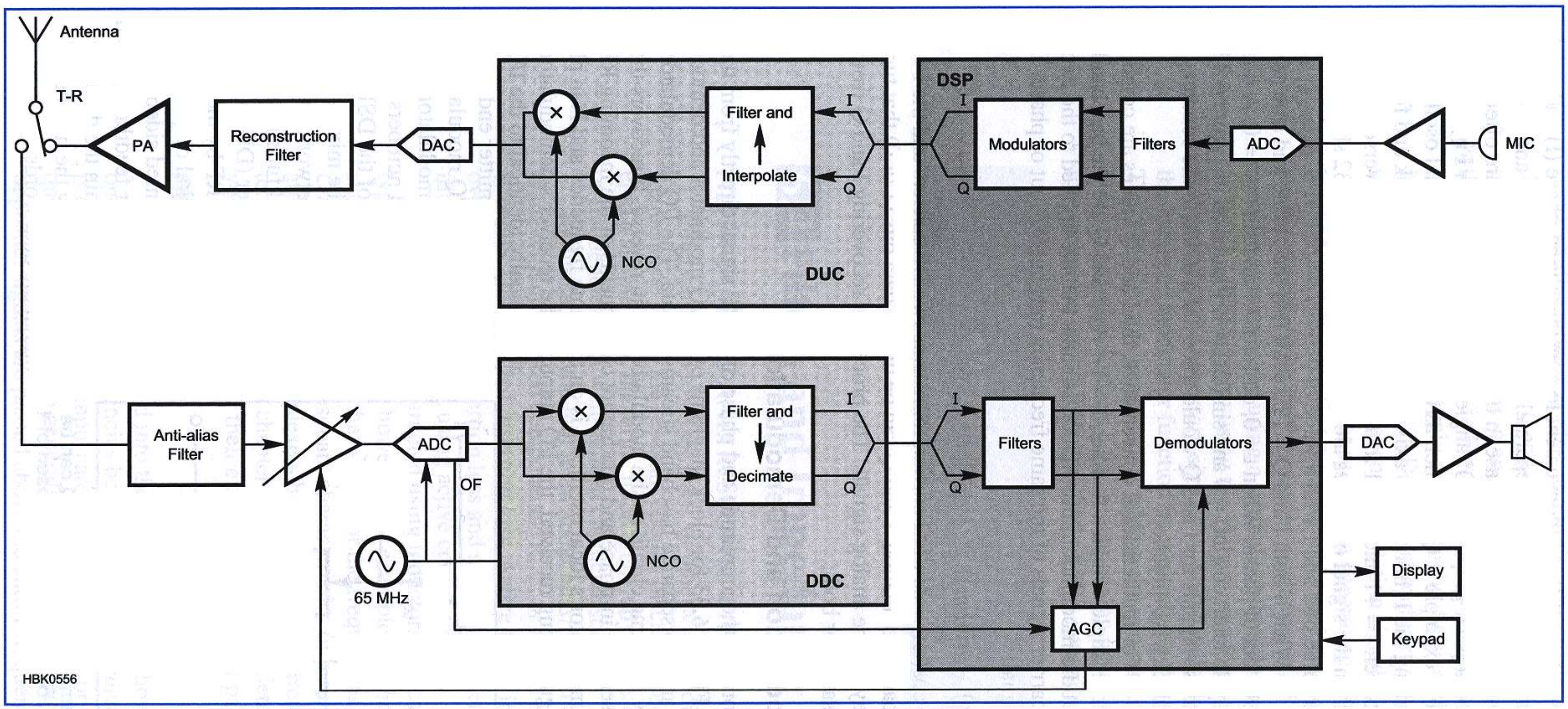


Figure 6.31 — An SDR transceiver based on direct digital conversion (DDC), sampling directly at the RF frequency. DUC stands for digital up-conversion. The block labeled DDC is a digital down-converter.

SDR Hardware (cont.)

- Baseband audio or data signals can use a slower converter called *successive approximation or a sigma-delta ADC.*

- Very few **analog** components remain In DDS transceivers
- The digitization sample rates of ADCs & DACs (their speeds) is the only limit on the size of bandwidths they can process at RF or IF frequencies.

- **Which of the following is a type of analog-to-digital conversion?**
  ► **Successive approximation**

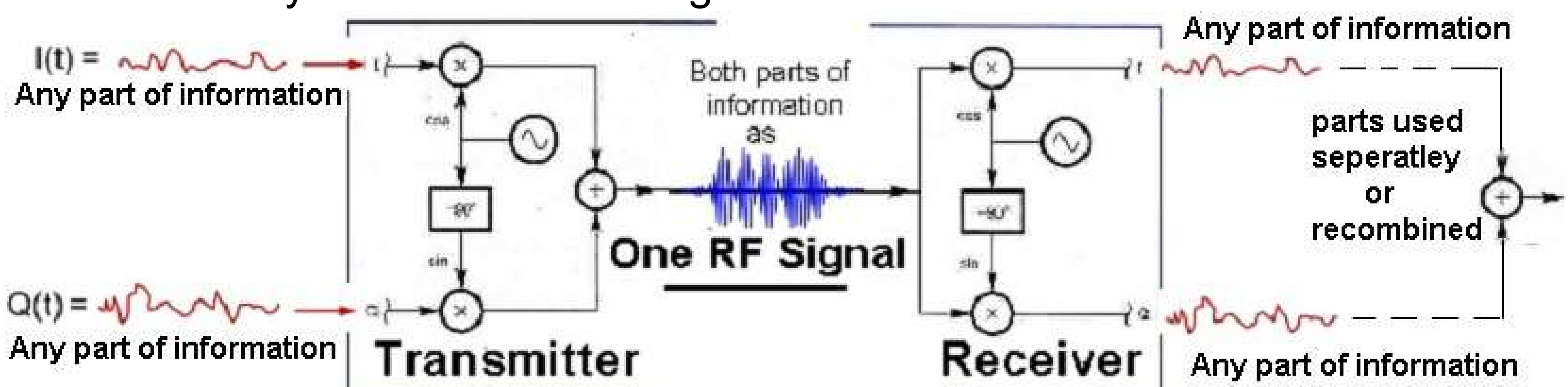
- **What aspect of receiver analog-to-digital conversion determines the maximum receive bandwidth of a direct-sampling software defined radio (SDR)?**
  ► **Sample rate**

I/Q Modulation and Demodulation

DSP Modulation: Encodes (places) binary data 0s and 1s onto an analog carrier wave by shifting its amplitude, frequency, or phase.

I/Q Modulation and Demodulation:

- **Transmitter**:
  - An I/Q modulator controls the amplitude and phase of an RF signal directly from the I and Q components
  - In a digital I/Q modulator, the I and Q signals are streams of digital bits from ADCs.
  - I= is In-Phase, **Q** is a quarter cycle behind I = ***Quadrature***
  - The mixers, oscillator, phase-shift network and summer are all digital functions in the DUE digital up converter block and converted to RF by DAC.
- **Receiver:**
  - Similar (reverse) process is used for the Digital Demodulate for receiving and applying FFT to I and Q to change them to the frequency domain, demodulation and recovery of the baseband signals.



DSP and SDR

Digital Signal Processing (DSP)

I/Q Modulation and Demodulation

- DSP can easily generated SSB signals:
  - Generating precise 90 deg quadrature phase shifts over wide frequency ranges with a combination of filters called the **Hilbert Transform:**
    - Produces the phase shifted message and RF carriers
  - Performs Balanced Modulation by multiplying the sampled signals together as numbers.
  - Finally, a DAC returns the numbers back to analog waveform.
  - This technique makes the quadrature technique of popular method of SSB generation in DSP systems.

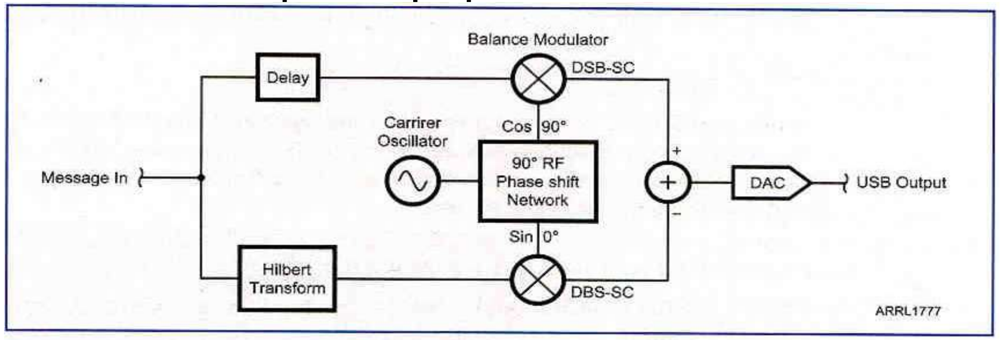


Figure 6.33 — Although very difficult to build in analog circuitry, the Hilbert Transform filter can be implemented by DSP software to generate SSB signals. A delay is required in the non-filtered path so that the message signals remain in phase at the double-balanced modulators.

- **What type of digital signal processing filter is used to generate an SSB signal? ► A Hilbert-transform filter**
- **Which method generates an SSB signal using digital signal processing?**
  ► Signals are combined in quadrature phase relationship

Break ?

6.4 Filters and Impedance Matching

Pg: 6-32

Filter Classifications

Filters: Used to block, pass, or modify signals for some range of frequencies.

- Generally act over a broad ranges of frequencies with well defined characteristics.
- **Passive filters:**
  - Made with un-powered components
  - Use R,C or L components
  - Have some loss called *Insertion loss.*
  - Mechanical filters: Use discs & rods
  - Cavity filters: Use cylinders or rectangular containers
  - Helical Filters – Tuned Coils inside containers
- **Active Filters:**
  - Include power amplifying devices to overcome the filter insertion loss
  - Some times produces signal gain.
  - Some types of filters can only be built using active components.

Filter Classifications (cont'd):

- **Pass Band**: The range of Frequencies passed nearly full strength to ½ power.
- **Stop Band**: The range of Frequencies attenuated ½ power or less of input.

- Lowpass
- Highpass
- Band pass
- Notch

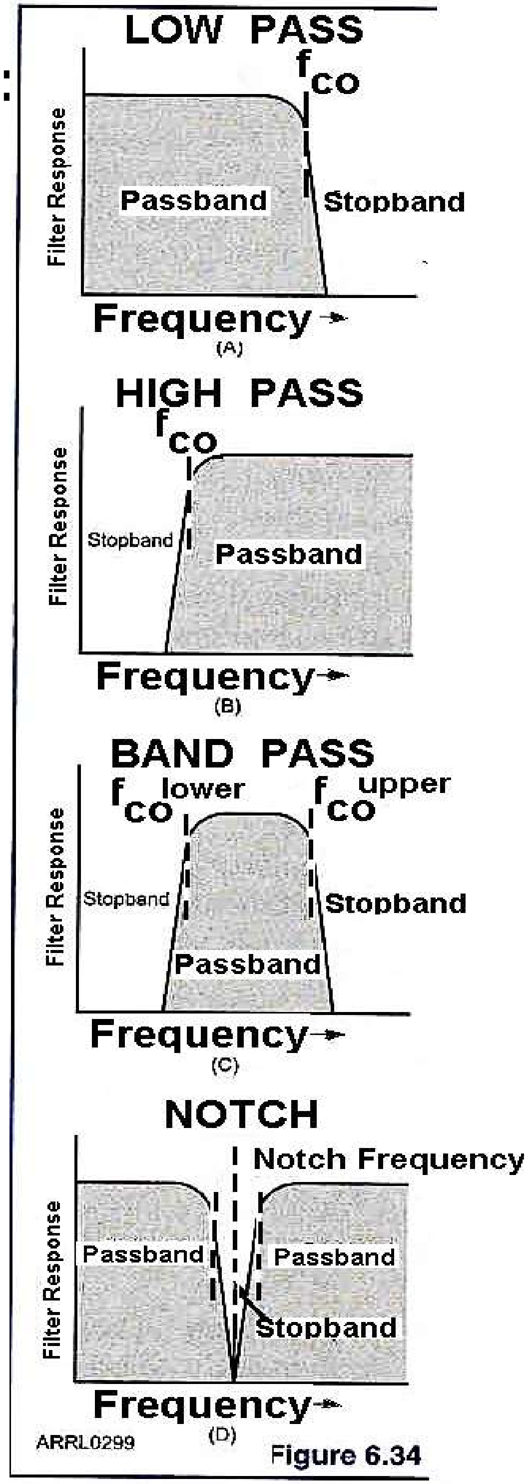
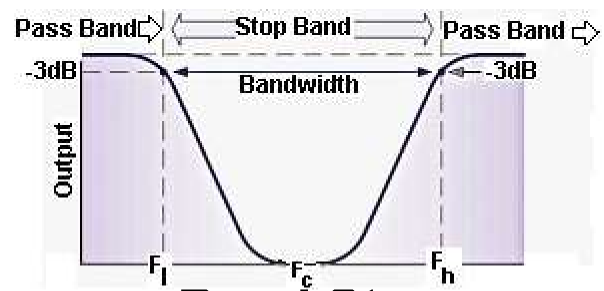


Figure 6.34

- **Cutoff Frequency F_{co}**:

  -3 dB (½ power point)
  - For Lowpass: The point where lower frequencies are ½ strength or stronger.
  - For Highpass: The point where higher frequencies are ½ strength or stronger
  - For Notch and Band Stop: The points were higher and lower frequencies are ½ power or weaker.

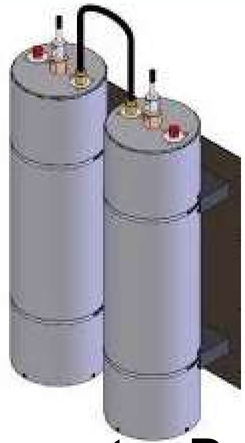


Band Stop
(wide notch)
(opposite of band pass)

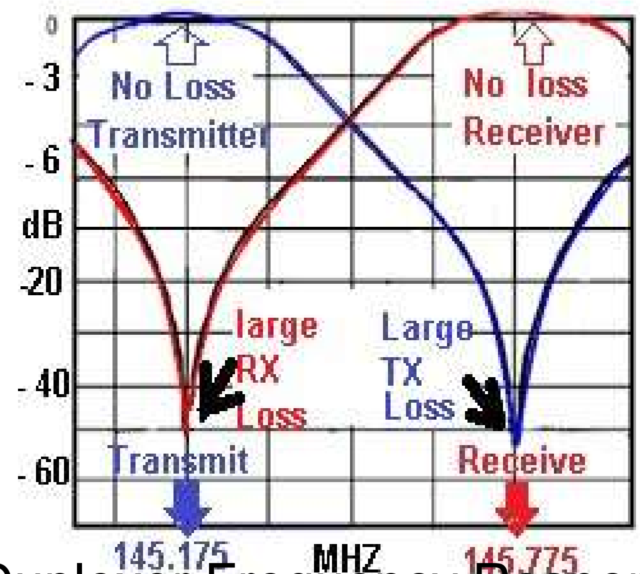
Filter Classifications (cont'd)

Cavity Filters:

- *Use the* resonant characteristic of conducting tubes or boxes, with sharp responses and low loss to act as a filters in repeater duplexers.



Repeater Duplexer Cavity Filters



Duplexer Frequency Response

- Which of the following filters is used in a 2-meter band repeater duplexer?
  ► A cavity filter

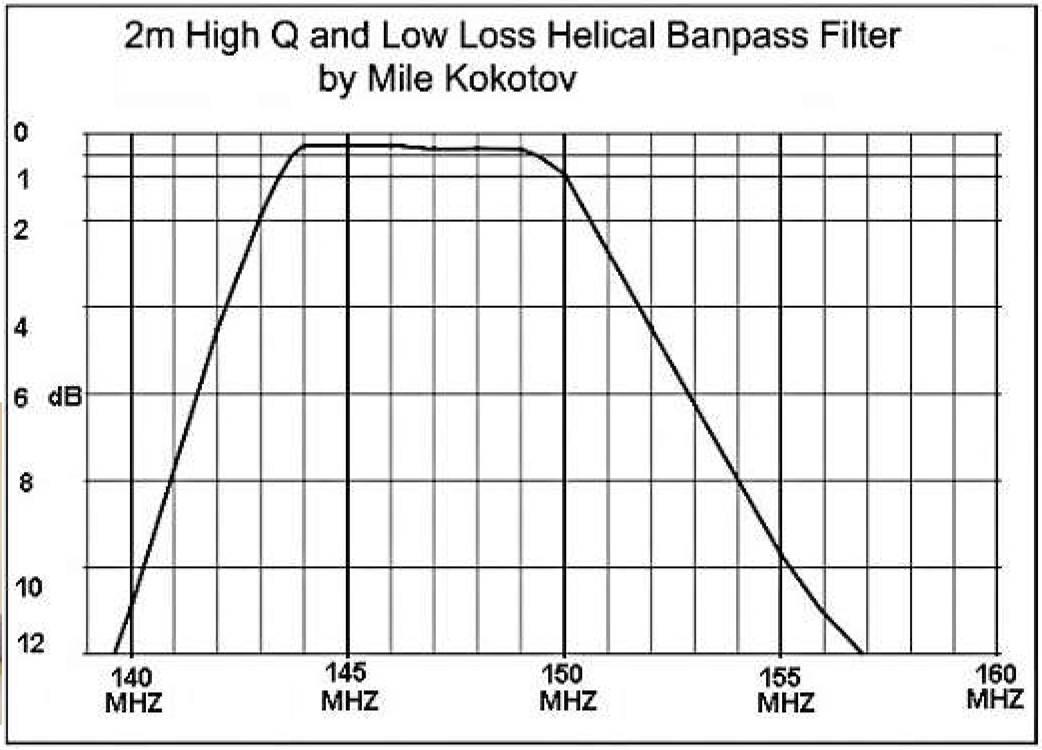
Filter Classifications (cont'd)

Helical Filters:

- Similar to cavity filters, except they are shaped as helical coils inside boxes and are used as notch or band-pass filters.

Helical Filter

Helical Filter inside



- Which of the following is most frequently used as a band-pass or notch filter in VHF and UHF transceivers?
  ► A helical filter

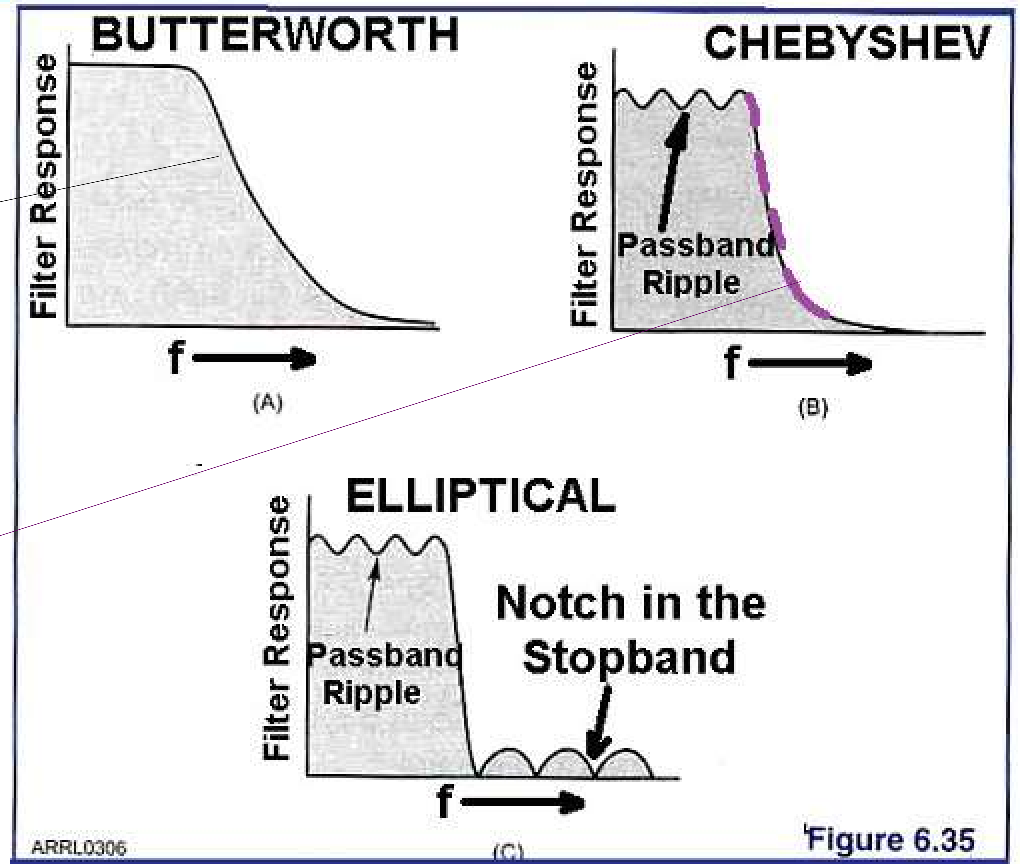
Filters and Impedance Matching
Filter Families and Response Types
Filter Design

Most Common Filters:
- Butterworth
- Chebyshev
- Elliptical

Cutoff transition
- Steepness between passband to stopband

Ripple
- Variations in the pass-band.

Skirts:
- Portion of response curve outside the passband.



- Which filter type has ripple in the passband and a sharp cutoff?
  - ► A Chebyshev filter.
- What are the characteristics of an elliptical filter?
  - ► Extremely sharp cutoff with one or more notches in the stop band

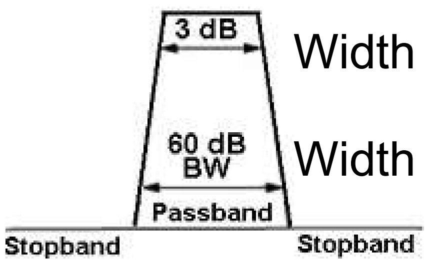
Bandwidth

- Describes the frequency difference between the filters cutoff frequencies or other frequencies with specific amounts of attenuation.

Shape Factor:

- Compares the frequency bandwidth at two levels of attenuation.
  In Amateur Radio the ratio is the -6dB and -60dB points.

Example: A filter with -6dB Bandwidth of 1.8kHz and a -60dB bandwidth of 5.4kHz has -60 to -6 dB shape factor of:



Width at 1.8kHz

Width at 5.4kHZ

$$\frac{5.4}{1.8} = 3.0 \text{ shape factor}$$

Which of the following measures a filter's ability to reject signals in adjacent channels?

→ Shape factor

Filters and Impedance Matching
Crystal Filters

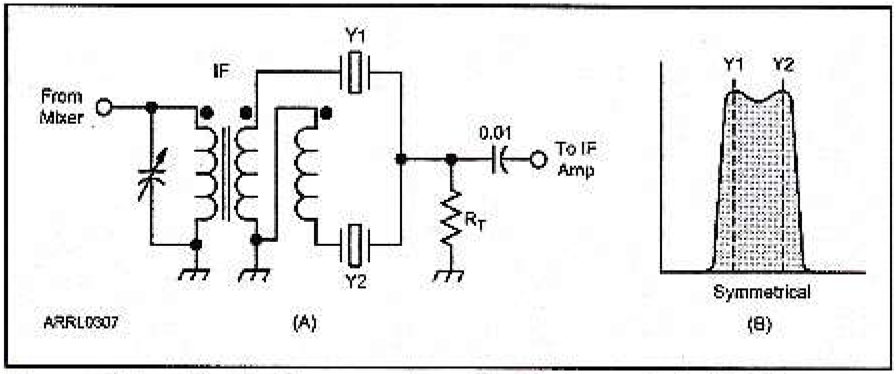


Figure 6-36 — Part A shows a schematic diagram of a half-lattice crystal filter. (B) shows a typical response curve for this type of filter. Note the steep skirts on the response between y_1 and y_2, indicating good rejection of signals outside the passband.

Famous Collins Radio Crystal Lattice Filter.

- Used in low level signal IF sections of analog superheterodyne receivers
- Narrow bandwidths, steep response skirts.
- Provide narrow bandwidths to separate one signal from many.
- Typical Passbands: 2.4Khz for SSB, 250-500Hz for CW

<u>Crystal ladder Filter:</u>
Inferior to Crystal Lattice which have wider bandwidth control & require more complex design

- What is a crystal lattice filter?
  - ► A filter for low-level signals made using quartz crystals

Filters and Impedance Matching
Active Filters

Active filters:

- Use amplification to crate its frequency response.
- LC filters have a fixed frequency response and insertion loss and are usually larger and heavier.
- Advantages of active filters:
  - Provide good gain & frequency selection
  - They are accurately tuned to frequency with a Potentiometer.
  - Need only a few resistors and capacitors

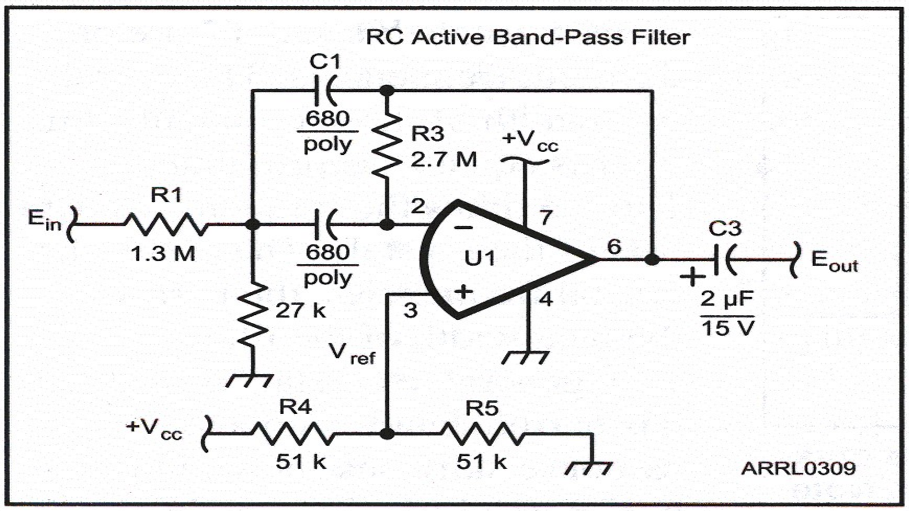
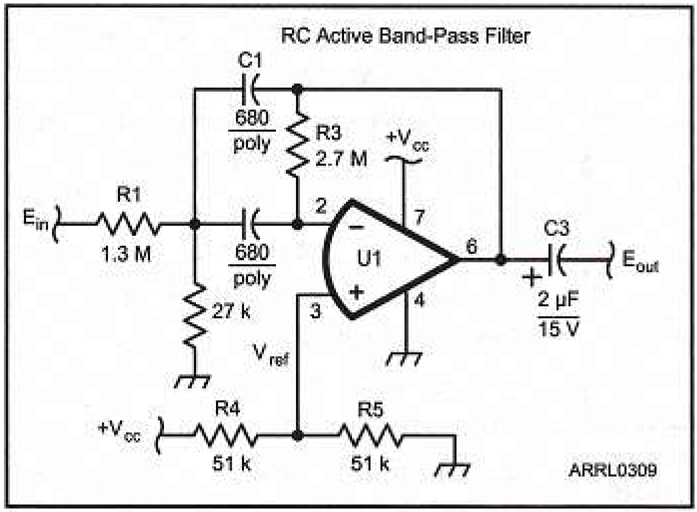


Figure 6.37 — This RC *multiple-feedback* band-pass filter is typical of active filters. This filter has a center frequency of 900 Hz and would be good for use with CW signals. Increasing C1 and C2 would lower the center frequency.

Filters and Impedance Matching
Active Filters

- Disadvantages of Active Filters are:
  - Require a power source
  - Inexpensive Op Amps have limited useful upper frequency of 100s of kHz
  - Output swing must be less than the supply voltage
  - Strong signals may overload them causing distortion.
  - To prevent instability and ringing (sustained oscillations)
    - Gain is typically kept less than 2
    - Q(ratio of center frequency to bandwidth) is typically kept less than 5.



- How can unwanted ringing and audio instability be prevented in an op-amp audio filter?
  → Restrict both gain and Q

Digital Signal Processing (DSP) Filters

There are advantages to filtering signals digitally.

- DSP can create filters that are impractical or impossible with physical components(L,C). Example: "brick wall" filters with extremely steep cutoffs.

- Drawback of DSP filters: They require computing hardware to implement them.
- Advantage: Only change the software program to change the filter characteristics.

- Active programs (in the Transceiver) allow adapting to changing signal conditions = ***Adaptive processing***
  Example:adaptive or ***automatic notch filter***

- What kind of digital signal processing audio filter is used to remove unwanted noise from a received SSBsignal? An adaptive filter

Finite Impulse Response (FIR) Filters

Digital filters are characterized to their response to narrow pulse.

- An infinitely narrow noise is called an *impulse.*
- A filters response to an impulse is called its *impulse response*.

Two primary DSP filter categories:

- Finite Impulse Response Filter (FIR)
- Infinite Impulse Response Filter (IIR)

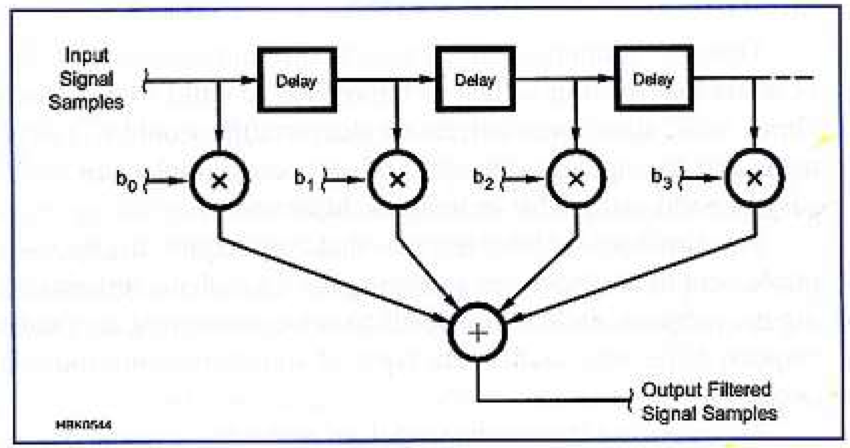


Figure 6.38 — A 4-tap FIR filter. The bn values are the filter coefficients.

A disadvantage of RC filters is long output decays, so high gain & narrow bandwidths can cause ringing.

- FIR: Filter response is finite.
  Inputs are stored in shift registers for delays & shifts called taps which are multiplied by filter coefficients. The completion of registers ends the signal.
- The more taps, the more precise the filter output is.

- **What is the function of taps in a digital signal processing filter?**
  → Provide incremental signal delays for filter algorithms
- **Which of the following would allow a digital signal processing filter to create a sharper filter response?**
  →More taps

Infinite Impulse Response (IIR) Filters

An IIR Filter's response:

- Is designed to lasts "forever", because it contains feedback and feed-forward loops. The output theoretically never goes to zero –(like an analog filter).

- Unlike FIR filters, an IIR Filter's, signal components can be delayed by different amounts, so fewer adders and multipliers are required than in an FIR filter to get the same response. So IIR filters are used where computations are limited.

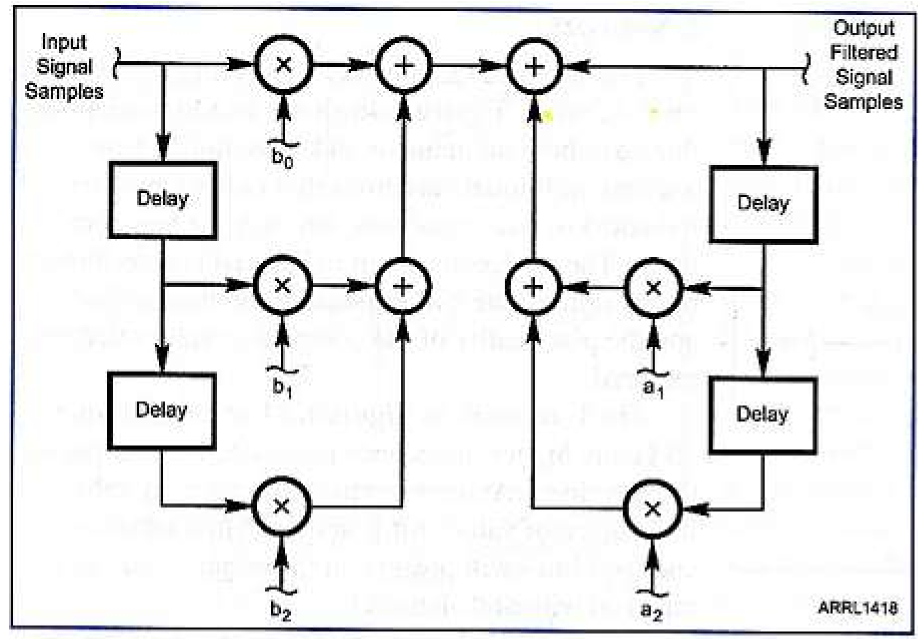


Figure 6.39 — An IIR filter with three feed-forward taps and two feedback taps.

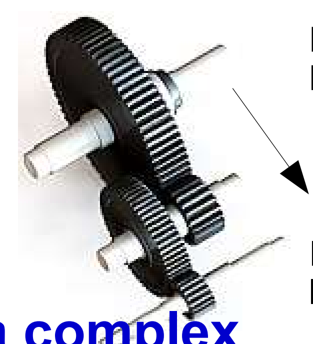
- Which of the following is generally true of Finite Impulse Response (FIR) filters?
  ► FIR filters can delay all frequency components of the signal by the same amount

Filters and Impedance Matching
Impedance Matching

Antenna couplers, transmatches, matchboxes Antenna tuners,etc:

- Purpose of is to convert (match) the antenna and feedline impedance to the output of the transceiver (usually 50 ohms).
- The circuit transforms a complex impedance.
    - <u>Complex Impedance</u>: <u>resistive and reactive (R, L& C)</u>
  - The reactive part is canceled out (with arrangements of L's & C's)
  - It transforms the remaining resist rive portion to the desired value.

- Mechanical analog is the gearbox:
  - Changes one speed and torque to <u>a different combination of speed and torque</u>.



- **How does an impedance-matching circuit transform a complex impedance to a resistive impedance?**
  It cancels the reactive part of the impedance and changes the resistive part to the desired value

Filters and Impedance Matching
Impedance Matching
L-Networks

L-Networks:
- Simplest LC impedance matching network.

Fig 6.40 shows its 4 variations of inductors and capacitors.

- The choice of circuit depends upon the the ratio of the two impedances to matched.

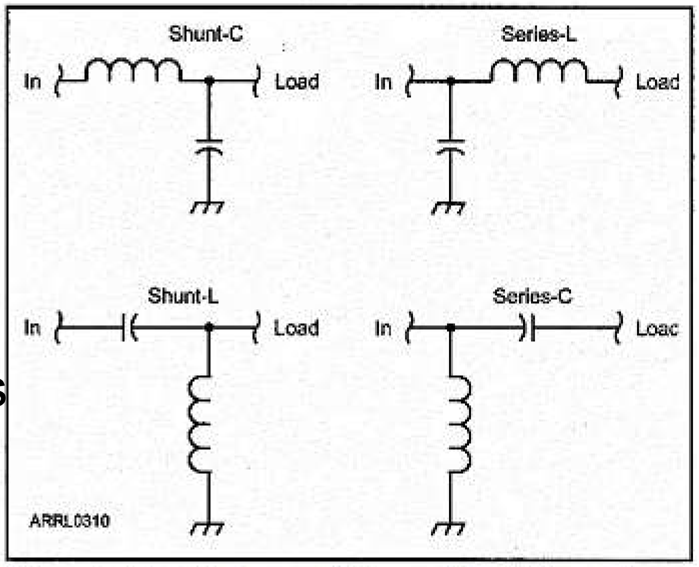


Figure 6-40 —The four variations of the LC impedance-matching L-network. Shunt or series refers to the connection of the component closest to the impedance to be matched.

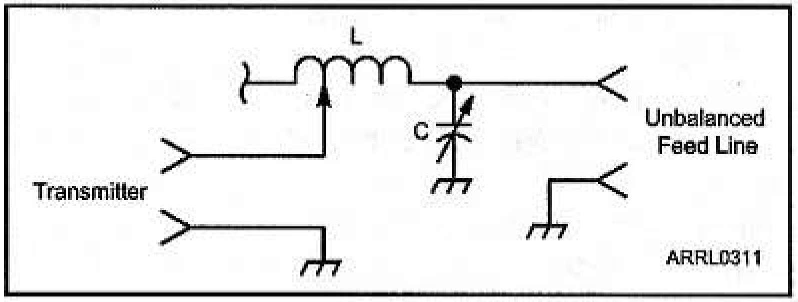


Figure 6-41 — An L-network antenna coupler, useful for an unbalanced feed line, such as coaxial cable. This circuit can transform impedances at the feed line input to the 50-Ω impedances preferred by most transceivers.

L- Network of 6.41 transforms 50 Ohms to higher impedance feedline(~600 Ohms).
Adjust the coil tap and then the variable capacitor for lowest SWR.

Filters and Impedance Matching

Impedance Matching
Pi and Pi-L Networks

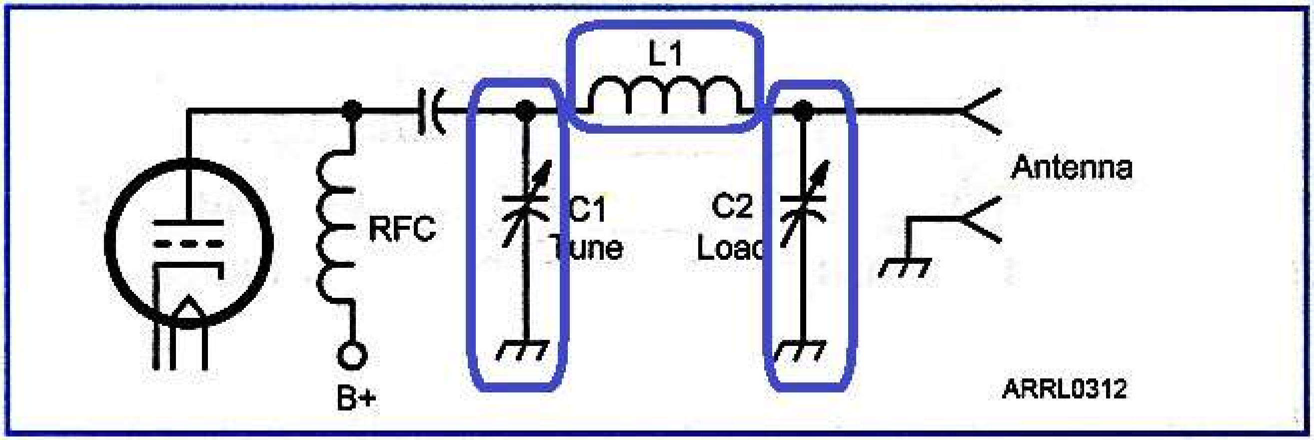
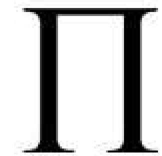


Figure 6.42 — A pi-network output-coupling circuit. C1 adjusts the circuit's tuning to resonance (TUNE) and C2 adjusts the load impedance presented to the tube (LOAD).



Greek
Pi

- How are the capacitors and inductors of a low-pass filter Pi-network arranged between the network's input and output?
  - ► A capacitor is connected between the input and ground, another capacitor is connected between the output and ground, and an inductor is connected between the input and output

Filters and Impedance Matching
Impedance Matching
Pi and Pi-L Networks

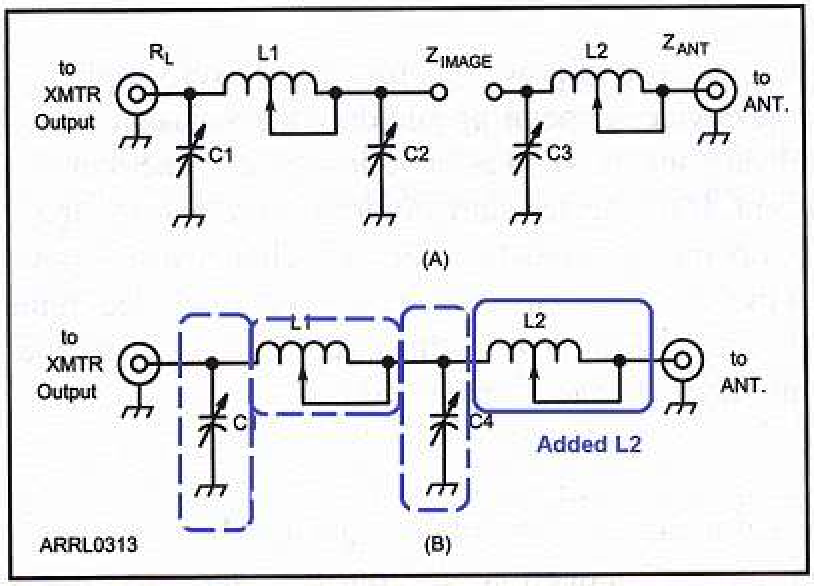


Figure 6-43 — The pi-L-network uses a pi-network to transform the transmitter output impedance (R_L) to an intermediate "image" impedance (Z_{IMAGE}). An L-network then transforms Z_{IMAGE} to the antenna impedance, Z_{ANT}.

- Which describes a Pi-L network? ► A Pi-network with an additional output series inductor.
- What is the purpose of adding an inductor to a Pi-network to create a Pi-L-network? ► Greater harmonic suppression.

Filters and Impedance Matching
Impedance Matching
T-Networks

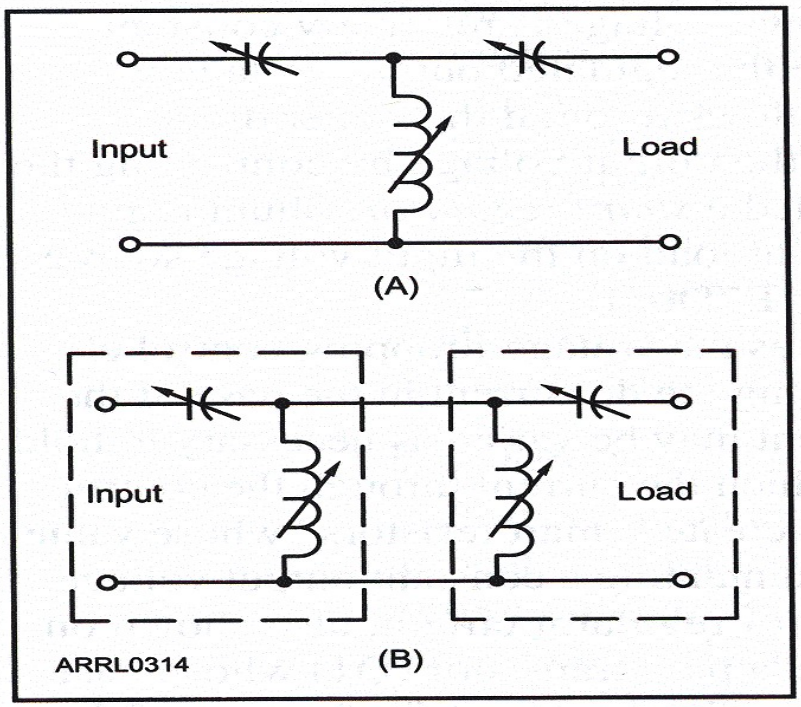


Figure 6-44 — The T-network at (A) can also be thought of as two L-networks back-to-back as shown at (B). This network has low losses and is a widely used circuit in amateur antenna couplers.

- What is the frequency response of a T-network with series capacitors and a shunt inductor? ► High-pass

6.5 Power Supplies

Pg: 6-41

Power Supplies will be covered **next week**