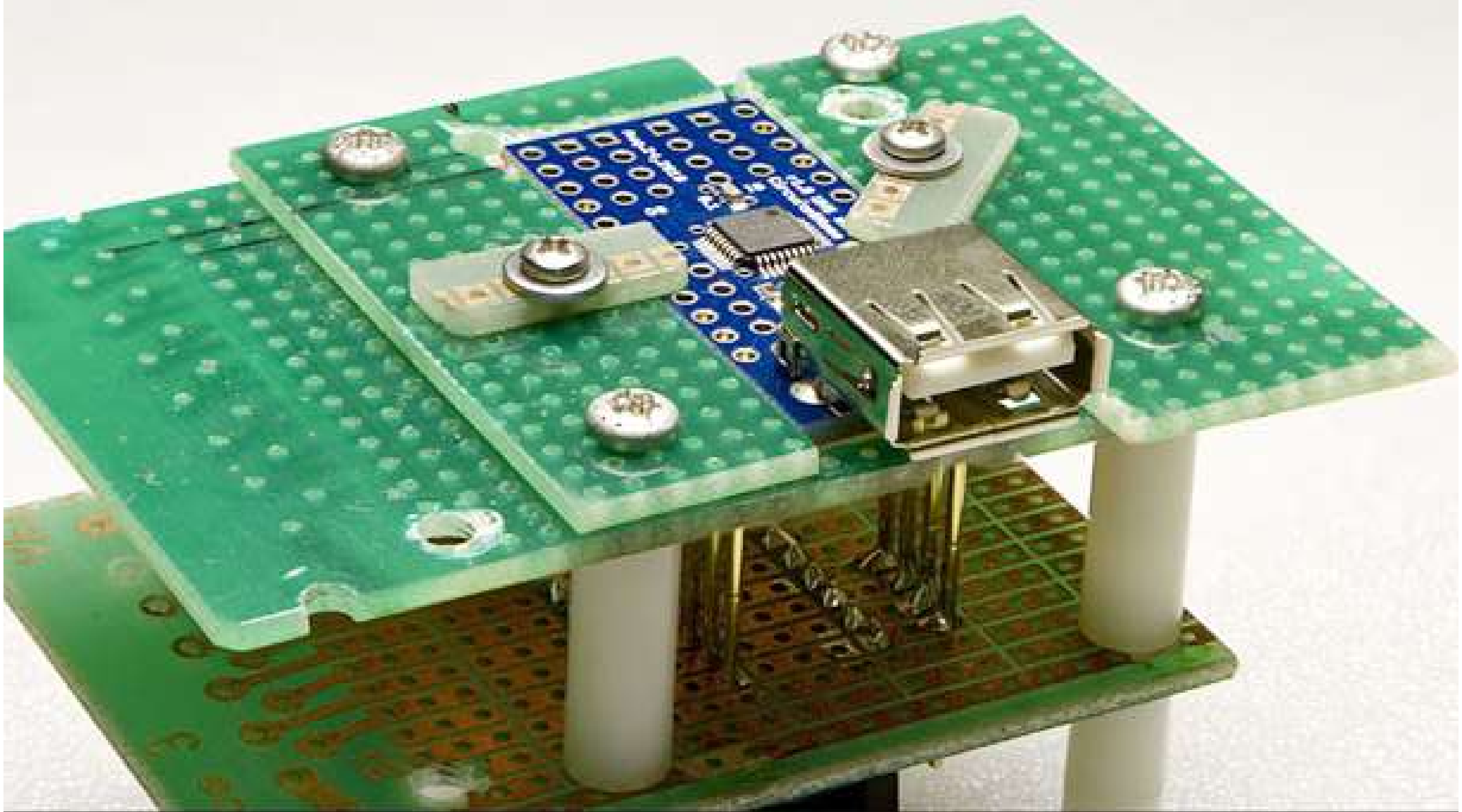
Communication Circuits

Rheba Shipley



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Chapter- 1

Frequency Multiplier

In electronics, a **frequency multiplier** is an electronic circuit that generates an output signal whose output frequency is a harmonic of its input frequency. Frequency multipliers consist of a nonlinear circuit that distorts the input signal and consequently generates harmonics of the input signal. A subsequent bandpass filter selects the desired harmonic frequency and removes the unwanted fundamental and other harmonics from the output.

Frequency multipliers are often used in frequency synthesizers and communications circuits. It can be more economic to develop a lower frequency signal with lower power and less expensive devices, and then use a frequency multiplier chain to generate an output frequency in the microwave or millimeter wave range. Some modulation schemes, such as frequency modulation, survive the nonlinear distortion without ill effect (but schemes such as amplitude modulation do not).

Frequency multiplication is also used in nonlinear optics. The nonlinear distortion in crystals can be used to generate harmonics of laser light.

Theory

A pure sinewave at frequency f has no harmonics. If it goes through a linear amplifier, the result continues to be pure (but may acquire a phase shift).

If the sinewave is run through a stateless nonlinear circuit (transcribing function), the resulting distortion creates harmonics. The distorted signal can be described by a Fourier series in f.

$$x(t) = \sum_{k=-\infty}^{\infty} c_k e^{i2\pi kft}.$$

The nonzero c_k represent the generated harmonics. The Fourier coefficients are given by integrating over the fundamental period T:

$$c_k = \frac{1}{2\pi} \int_0^T x(t)\, e^{-i2\pi kt/T}\, dt$$

These harmonics can be selected by a bandpass filter.

The power in the distorted signal is spread across all the resulting harmonics. An ideal halfwave rectifier, for example, has all nonzero coefficients. An approximate circuit could use a diode.

From a conversion efficiency standpoint, the nonlinear circuit should maximize the coefficient for the desired harmonic and minimize the others. Consequently, the transcribing function is often specially chosen. Easy choices are to use an even function to generate even harmonics or an odd function to for odd harmonics. A full wave rectifier, for example, is good for making a doubler. On the other hand, a tripler may over drive an amplifier to symmetrically distort the positive and negative peaks.

YIG multipliers often want to select an arbitrary harmonic, so they use a stateful distortion circuit that converts the input sine wave into an approximate impulse train. The ideal (but impractical) impulse train generates an infinite number of (weak) harmonics. In practice, an impulse train generated by a monostable circuit will have many usable harmonics. YIG multipliers using step recovery diodes may, for example, take an input frequency of 1 to 2 GHz and produce outputs up to 18 GHz. Sometimes the frequency multiplier circuit will adjust the width of the impulses to improve conversion efficiency for a specific harmonic.

Circuits

Spark generator

Before amplifiers, frequency multipliers were the way to generate radio frequencies.

- Spark gap transmitter
- Arc converter

Diode

Clipping circuits. Full wave bridge doubler.

Class C amplifier and multiplier

Efficiently generating power becomes more important at high power levels. Linear Class A amplifiers are at best 25 percent efficient. Push-pull Class B amplifiers are at best 50 percent efficient. The basic problem is the amplifying element is dissipating power.

Switching Class C amplifiers are nonlinear, but they can be better than 50 percent efficient because an ideal switch does not dissipate any power.

A clever design can use the nonlinear Class C amplifier for both gain and as a frequency multipler.

Step recovery diode

Generating a large number of useful harmonics requires a fast nonlinear device.

Step recovery diodes.

Varactor diode

Resistive loaded varactors. Regenerative varactors. Penfield.

Frequency multipliers have much in common with frequency mixers, and some of the same nonlinear devices are used for both: transistors operated in Class C and diodes. In transmitting circuits many of the amplifying devices (vacuum tubes or transistors) operate nonlinearly and create harmonics, so an amplifier stage can be made a multiplier by tuning the tuned circuit at the output to a multiple of the input frequency. Usually the power (gain) produced by the nonlinear device drops off rapidly at the higher harmonics, so most frequency multipliers just double or triple the frequency, and multiplication by higher factors is accomplished by cascading doubler and tripler stages.

Previous

Frequency multipliers use circuits tuned to a harmonic of the input frequency. Non-linear elements such as diodes may be added to enhance the production of harmonic frequencies. Since the power in the harmonics declines rapidly, usually a frequency multiplier is tuned to only a small multiple (twice, three times, or five times) of the input frequency. Usually amplifiers are inserted in a chain of frequency multipliers to ensure adequate signal level at the final frequency.

Since the tuned circuits have a limited bandwidth, if the base frequency is changed significantly (more than one percent or so), the multiplier stages may have to be adjusted; this can take significant time if there are many stages.

PLLs with frequency dividers

In **digital electronics**, frequency multipliers are often used along with frequency dividers and phase-locked loops to generate any desired frequency from an external reference frequency. The frequency multiplication is carried out in the phase-locked loop's feedback loop, by using a frequency divider on the output of the voltage controlled oscillator (VCO). This **divided-down output** is fed-back to the input comparator and compared to the reference frequency. Since the divided down frequency is smaller than the reference frequency, the comparator generates a voltage signal to the VCO, telling it to increase the output frequency. It continues to do this via the feedback loop, raising the VCO output frequency, until the divided-down frequency from the VCO output is equal to the reference frequency. At this point the comparator stabilizes and generates no more signals to the VCO, or only minor changes to maintain stability. The output frequency from the VCO will be stable at the input reference frequency multiplied by the value of the feedback divider.

A PLL with a frequency divider in its feedback loop acts as a frequency multiplier and is a type of frequency synthesizer.

Integer-N synthesizer

In a configuration with an integer-N divider, its VCO's output frequency is N times its reference, or input, frequency.

Fractional-N synthesizer

Periodic changes in the integer value of an integer-N frequency divider will effectively result in a multiplier with both whole number and fractional component. Such a multiplier is called a fractional-N synthesier after its fractional component. Fractional-N synthesizers provide an effective means of achieving fine frequency resolution with lower values of N, allowing loop architectures with tens of thousands of times less phase noise than alternative designs with lower reference frequencies and higher integer N values. They also allow a faster settling time because of their higher reference frequencies, allowing wider closed and open loop bandwidths.

Delta sigma synthesizer

A delta sigma synthesizer adds a randomization to programmable-N frequency divider of the fractional-N synthesizer. This is done to shrink sidebands created by periodic changes of an integer-N frequency divider.

Chapter- 2

Frequency Synthesizer and Current Loop

Frequency synthesizer

A **frequency synthesizer** is an electronic system for generating any of a range of frequencies from a single fixed timebase or oscillator. They are found in many modern devices, including radio receivers, mobile telephones, radiotelephones, walkie-talkies, CB radios, satellite receivers, GPS systems, etc. A frequency synthesizer can combine frequency multiplication, frequency division, and frequency mixing (the frequency mixing process generates sum and difference frequencies) operations to produce the desired output signal.

Types

Three types of synthesizer can be distinguished. The first and second type are routinely found as stand-alone architecture: **Direct Analog Synthesis** (also called a **mix-filter-divide** architecture as found in the 1960s HP 5100A) and by comparison the more modern **Direct Digital Synthesizer** (DDS) (Table-Look-Up). The third type are routinely used as communication system IC building-blocks: indirect digital (PLL) synthesizers including integer-N and fractional-N.

Digiphase Synthesizer

It is in some ways similar to a DDS, but it has architectural differences. One of its big advantages is to allow a much finer resolution than other types of synthesizers with a given reference frequency.

History

Although frequency as the inverse of a wave period is a relatively recent idea, the origins of frequency synthesis can be found in the much older concept of angular velocity. The wheel trains of timekeeping devices have gear ratio relationships that were well-studied at least as far back as the time of Christian Huygens, who died in 1695.

Prior to widespread use of synthesizers, radio and television receivers relied on manual tuning of a local oscillator, such as with the turret tuner commonly used in television receivers prior to the 1980s. Variations in temperature and aging of components caused frequency drift. Automatic frequency control (AFC) solves some of the drift problem, but manual retuning was often necessary. Since transmitter frequencies are well known and very stable, an accurate means of generating fixed, stable frequencies would solve the problem.

A simple and effective solution employs the use of many stable resonators or oscillators, one for each tuning frequency. Quartz crystals offer good stability and are often used for this purpose. This "brute force" technique is practical when only a handful of frequencies are required, but quickly becomes costly and impractical in many applications. For example, the FM radio band in many countries supports 100 individual frequencies from about 88 MHz to 108 MHz. Cable television can support even more frequencies or channels over a much wider band. A large number of crystals increases cost and requires greater space.

Many coherent and incoherent techniques have been devised over the years. Some approaches include phase locked loops, double mix, triple mix, harmonic, double mix divide, and direct digital synthesis (DDS). The choice of approach depends on several factors, such as cost, complexity, frequency step size, switching rate, phase noise, and spurious output.

Coherent techniques generate frequencies derived from a single, stable master oscillator. In most applications, crystal oscillator are common, but other resonators and frequency sources can be used. Incoherent techniques derive frequencies from a set of several stable oscillators. The vast majority of synthesizers in commercial applications use coherent techniques due to simplicity and low cost.

Synthesizers used in commercial radio receivers are largely based on phase-locked loops or PLLs. Many types of frequency synthesiser are available as integrated circuits, reducing cost and size. High end receivers and electronic test equipment use more sophisticated techniques, often in combination.

System analysis and design

A well-thought-out *design procedure* is considered to be the first significant step to a successful synthesizer project. In the system design of a frequency synthesizer, states Manassewitsch, there are as many "best" design procedures as there are experienced synthesizer designers. System analysis of a frequency synthesizer involves output frequency range (or frequency bandwidth or tuning range), frequency increments (or resolution or frequency tuning), frequency stability (or phase stability, compare spurious outputs), phase noise performance (e.g., spectral purity), switching time (compare settling time and rise time), and size, power consumption, and cost. James A. Crawford says that these are mutually contradictive requirements

Trial-and-error superseded by calculation and control theory

The trial and error method was once the work-horse for designers of frequency synthesizers.

This began to change with the works of Floyd M. Gardner (his 1966 *Phaselock techniques*) and Venceslav F. Kroupa (his 1973 *Frequency Synthesis*). Manassewitsch calls this the Brute-force approach. Techniques and formulae have been provided by Dean Banerjee.

Gearbox approach

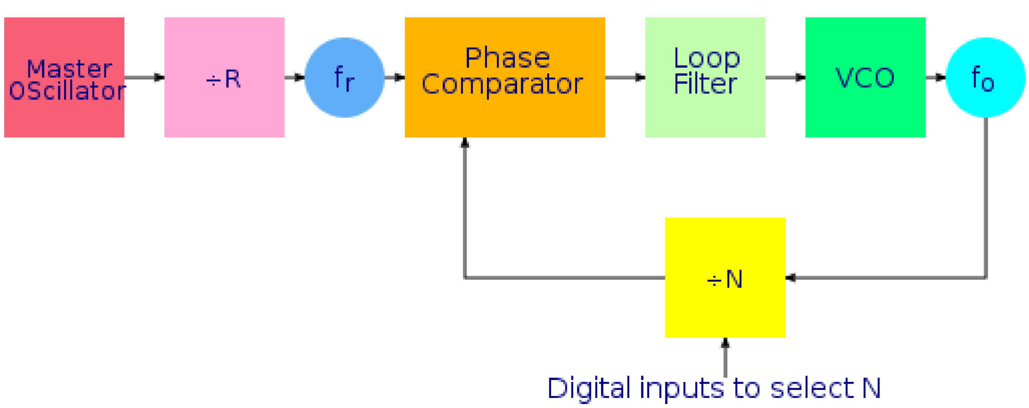
Surprisingly sophisticated mathematical techniques analogous to mechanical gear ratio relationships can be employed in frequency synthesis when the frequency synthesis factor is composed of multiplicative integers in the numerator and denominator. This method allows for effective planning of distribution and suppression of spectral spurs.

Modulo-N approach

Variable frequency synthesizers including DDS are routinely designed using this method.

Principle of PLL synthesizers

A phase locked loop is a feedback control system. It compares the phases of two input signals and produces an error signal that is proportional to the difference between their phases. The error signal is then low pass filtered and used to drive a voltage-controlled oscillator (VCO) which creates an output frequency. The output frequency is fed through a frequency divider back to the input of the system, producing a negative feedback loop. If the output frequency drifts, the phase error signal will increase, driving the frequency in the opposite direction so as to reduce the error. Thus the output is *locked* to the frequency at the other input. This other input is called the **reference** and is usually derived from a crystal oscillator, which is very stable in frequency. The block diagram below shows the basic elements and arrangement of a PLL based frequency synthesizer.



The key to the ability of a frequency synthesizer to generate multiple frequencies is the divider placed between the output and the feedback input. This is usually in the form of a digital counter, with the output signal acting as a clock signal. The counter is preset to some initial count value, and counts down at each cycle of the clock signal. When it reaches zero, the counter output changes state and the count value is reloaded. This circuit is straightforward to implement using flip-flops, and because it is digital in nature, is very easy to interface to other digital components or a microprocessor. This allows the frequency output by the synthesizer to be easily controlled by a digital system.

Example

Suppose the reference signal is 100 kHz, and the divider can be preset to any value between 1 and 100. The error signal produced by the comparator will only be zero when the output of the divider is also 100 kHz. For this to be the case, the VCO must run at a frequency which is 100 kHz x the divider count value. Thus it will produce an output of 100 kHz for a count of 1, 200 kHz for a count of 2, 1 MHz for a count of 10 and so on. Note that only whole multiples of the reference frequency can be obtained with the simplest integer N dividers. Fractional N dividers are readily available .

Practical considerations

In practice this type of frequency synthesiser cannot operate over a very wide range of frequencies, because the comparator will have a limited bandwidth and may suffer from aliasing problems. This would lead to false locking situations, or an inability to lock at all. In addition, it is hard to make a high frequency VCO that operates over a very wide range. This is due to several factors, but the primary restriction is the limited capacitance range of varactor diodes. However, in most systems where a synthesiser is used, we are not after a huge range, but rather a finite number over some defined range, such as a number of radio channels in a specific band.

Many radio applications require frequencies that are higher than can be directly input to the digital counter. To overcome this, the entire counter could be constructed using high-speed logic such as ECL, or more commonly, using a fast initial division stage called a *prescaler* which reduces the frequency to a manageable level. Since the prescaler is part of the overall division ratio, a fixed prescaler can cause problems designing a system with narrow channel spacings - typically encountered in radio applications. This can be overcome using a dual-modulus prescaler.

Further practical aspects concern the amount of time the system can switch from channel to channel, time to lock when first switched on, and how much noise there is in the output. All of these are a function of the *loop filter* of the system, which is a low-pass filter placed between the output of the frequency comparator and the input of the VCO. Usually the output of a frequency comparator is in the form of short error pulses, but the input of the VCO must be a smooth noise-free DC voltage. (Any noise on this signal naturally causes frequency modulation of the VCO.). Heavy filtering will make the VCO slow to respond to changes, causing drift and slow response time, but light filtering will produce noise and other problems with harmonics. Thus the design of the filter is critical to the performance of the system and in fact the main area that a designer will concentrate on when building a synthesiser system.

Current loop

A **current loop** describes two different electrical signalling schemes.

Digital

RS-232 / Current loop converter

For digital serial communications, a current loop is a communication interface that uses current instead of voltage for signaling. Current loops can be used over moderately long distances (tens of kilometres), and can be interfaced with optically isolated links.

Long before the RS-232 standard, current loops were used to send digital data in serial form for teleprinters. More than two teletypes could be connected on a single circuit allowing a simple form of networking. Older teletypes used a 60 mA current loop. Later machines, such as the ASR33 teleprinter, operated on a lower 20 mA current level and most early minicomputers featured a 20 mA current loop interface, with an RS-232 port generally available as a more expensive option. The original IBM PC serial port card had provisions for a 20 mA current loop. A digital current loop uses the absence of current for high (space or break), and the presence of current in the loop for low (mark).

The maximum resistance for a current loop is limited by the available voltage. Current loop interfaces usually use voltages much higher than those found on an RS-232 interface, and cannot be interconnected with voltage-type inputs without some form of level translator circuit.

MIDI (Musical Instrument Digital Interface) is a digital current loop interface.

Analog

Analog current loops are used where a device must be monitored or controlled remotely over a pair of conductors. Only one current level can be present at any time.

Given its analog nature, current loops are easier to understand and debug than more complicated digital fieldbuses, requiring only a handheld digital multimeter in most situations. Using fieldbuses and solving related problems usually requires much more education and understanding than required by simple current loop systems.

Additional digital communication to the device can be added to current loop using HART Protocol. Digital process buses such as FOUNDATION Fieldbus and Profibus may replace analog current loops.

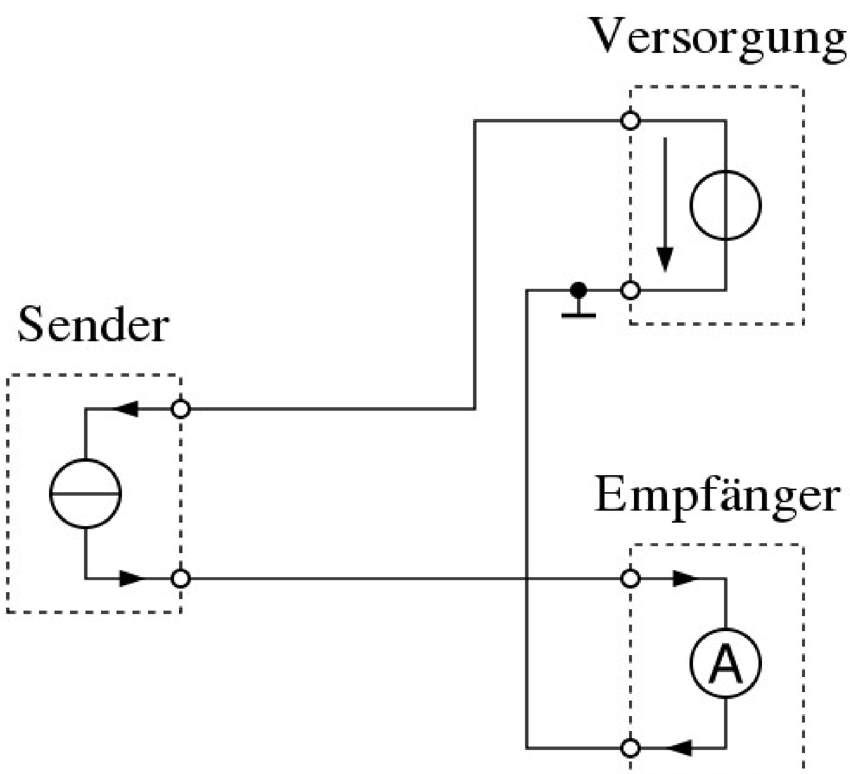
Process-control use

For industrial process control instruments, analog **4–20 mA** and 10–50 mA current loops are commonly used for analog signaling, with 4 mA representing the lowest end of the range and 20 mA the highest. The key advantages of the current loop are that the accuracy of the signal is not affected by voltage drop in the interconnecting wiring, and that the loop can supply operating power to the device. Even if there is significant electrical resistance in the line, the current loop transmitter will maintain the proper current, up to its maximum voltage capability. The *live-zero* represented by 4 mA allows the receiving instrument to detect some failures of the loop, and also allows transmitter devices to be powered by the same current loop (called *two-wire* transmitters). Such instruments are used to measure pressure, temperature, flow, pH or other process variables. A current loop can also be used to control a valve positioner or other output actuator. An analog current loop can be converted to a voltage input with a precision resistor. Since input terminals of instruments may have one side of the current loop input tied to the chassis ground (earth), analog isolators may be required when connecting several instruments in series.

Depending on the source of current for the loop, devices may be classified as *active* (supplying power) or *passive* (relying on loop power). For example, a chart recorder may provide loop power to a pressure transmitter. The pressure transmitter modulates the current on the loop to send the signal to the strip chart recorder, but does not in itself supply power to the loop and so is passive. (A *4-wire* instrument has a power supply input separate from the current loop.) Another loop may contain two passive chart recorders, a passive pressure transmitter, and a 24 V battery. (The battery is the active device).

Panel mount displays and chart recorders are commonly termed 'indicator devices' or 'process monitors'. Several passive indicator devices may be connected in series, but a loop must have only one transmitter device and only one power source (active device).

The relationship between current value and process variable measurement is set by calibration, which assigns different ranges of engineering units to the span between 4 and 20 mA. The mapping between engineering units and current can be inverted, so that 4 mA represents the maximum and 20 mA the minimum.



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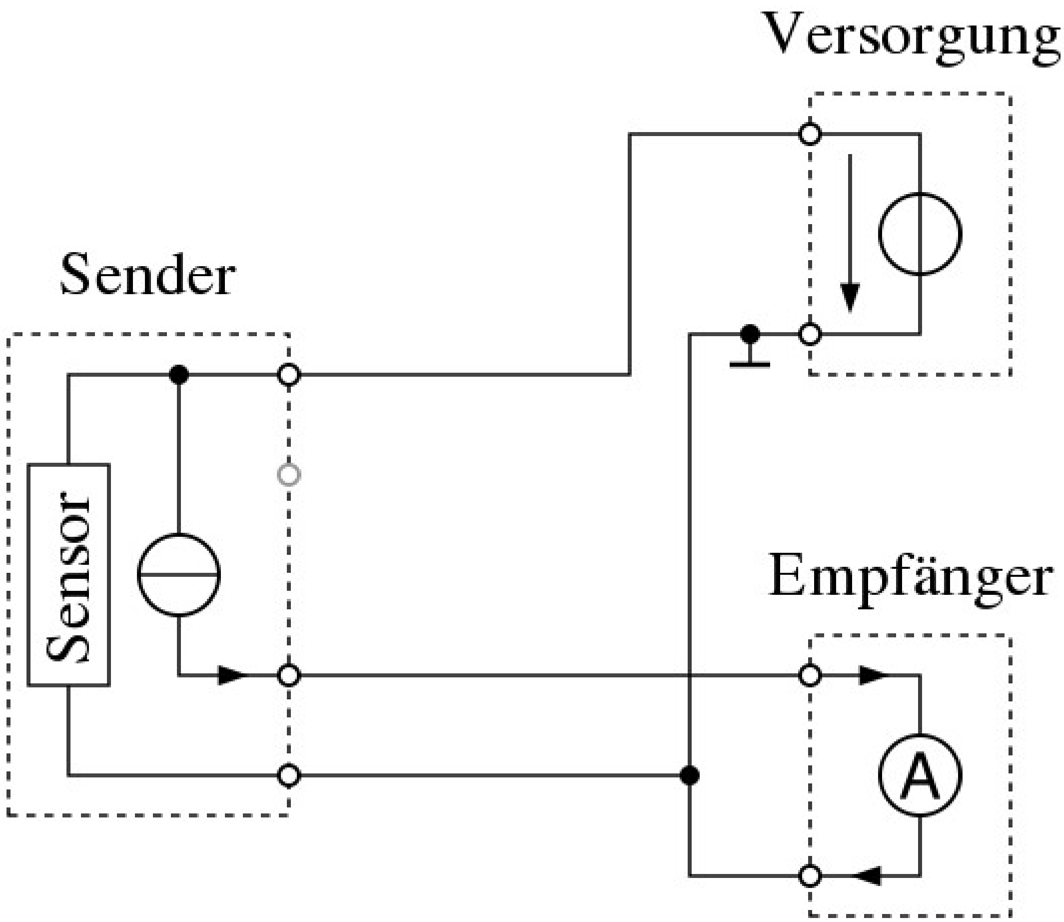

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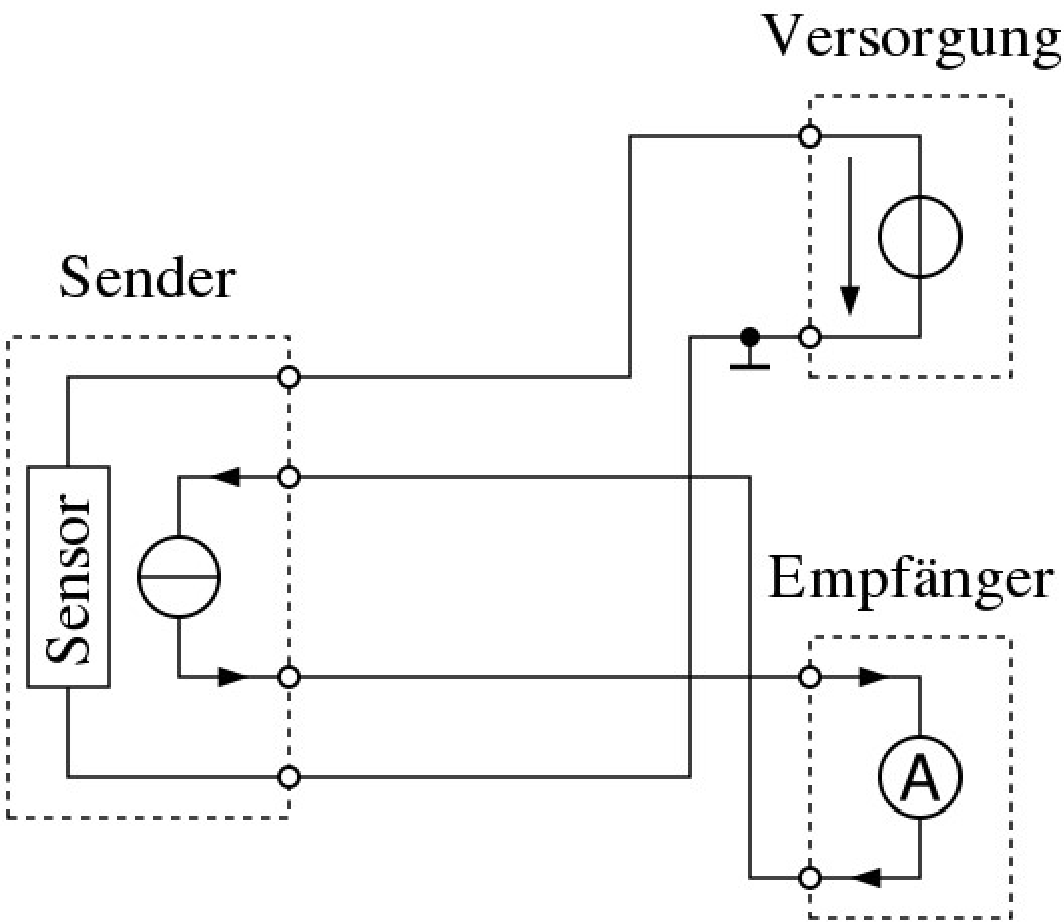


Typ 4

Long circuits

Analog current loops were occasionally carried between buildings by dry pairs in telephone cables leased from the local telephone company. 4–20 mA loops were more common in the days of analog telephony. These circuits require end-to-end direct current (DC) continuity. DC continuity is not available over a microwave radio, optical fiber, or a multiplexed telephone circuit connection.

Basic DC circuit theory shows that the current is the same all along the line. It was common to see 4–20 mA circuits that had loop lengths in miles or circuits working over telephone cable pairs that were longer than ten thousand feet end-to-end. There are still legacy systems in place using this technology. In Bell System circuits, voltages up to 125 VDC were employed.

Discrete control

Discrete control functions can be represented by discrete levels of current sent over a loop. This would allow multiple control functions to be operated over a single pair of wires. Currents required for a specific function vary from one application or manufacturer to another. There is no specific current that is tied to a single meaning. It is almost universal that 0 mA indicates the circuit has failed. In the case of a fire alarm, 6 mA could be normal, 15 mA could mean a fire has been detected, and 0 mA would produce a trouble indication, telling the monitoring site the alarm circuit had failed. Some devices, such as two-way radio remote control consoles, can reverse the polarity of currents and can multiplex audio onto a DC current.

These devices can be employed for any remote control need a designer might imagine. For example, a current loop could actuate an evacuation siren or command synchronized traffic signals.

Two-way radio use

A Motorola T-1300 series remote control is built in a telephone housing. The dial is replaced with a speaker and volume control. This remote control uses a two-wire circuit to control a base station.

Current loop circuits are one possible way used to control radio base stations at distant sites. The two-way radio industry calls this type of remote control **DC remote**. This name comes from the need for DC circuit continuity between the control point and the radio base station. The purpose current loop remote control is to save the cost of extra pairs of wires between the operating point and the radio transceiver. Some equipment, such as the Motorola MSF-5000 base station, uses currents below 4 mA for some functions. An alternative type, the Tone remote, is more complex but requires only an audio path between control point and base station. The patent does not describe this tone remote but confirms the use of the phrase to describe this system of signaling.

For example, a taxi dispatch base station might be physically located on the rooftop of an eight-story building. The taxi company office might be in the basement of a different building nearby. The office would have a remote control unit that would operate the taxi company base station over a current loop circuit. The circuit would normally be over a telephone line or similar wiring. Control function currents come from the remote control console at the dispatch office end of a circuit. In two-way radio use, an idle circuit would normally have no current present.

In two-way radio use, radio manufacturers use different currents for specific functions. Polarities are changed to get more possible functions over a single circuit. For example, imagine one possible scheme where the presence of these currents cause the base station to change state:

- no current means *receive on channel 1*, (the default).
- +6 mA might mean *transmit on channel 1*
- −6 mA might mean *stay in receive mode but switch to channel 2*. So long as the −6 mA current were present, the remote base station would continue to receive on channel 2.
- −12 mA might command the base station to *transmit on channel 2*.

Note that this circuit is polarity-sensitive. If a telephone company cable splicer accidentally reversed the conductors, selecting channel 2 would lock the transmitter on.

Each current level could close a set of contacts, or operate solid-state logic, at the other end of the circuit. That contact closure caused a change of state on the controlled device. Some remote control equipment could have options set to allow compatibility between manufacturers. That is, a base station that was configured to transmit with a +18 mA current could have options changed to (instead) make it transmit when +6 mA was present.

In two-way radio use, AC signals were also present on the circuit pair. If the base station were idle, receive audio would be sent over the line from the base station to the dispatch office. In the presence of a transmit command current, the remote control console would send audio to be transmitted. The voice of the user in the dispatch office would be superimposed over the DC current that caused the transmitter to operate.

Chapter- 3

Leased Line

A **leased line** is a service contract between a provider and a customer, whereby the provider agrees to deliver a symmetric telecommunications line connecting two or more locations in exchange for a monthly rent (hence the term lease). It is sometimes known as a 'Private Circuit' or 'Data Line' in the UK or as CDN (Circuito Diretto Numerico) in Italy. Unlike traditional PSTN lines it does not have a telephone number, each side of the line being permanently connected to the other. Leased lines can be used for telephone, data or Internet services. Some are ringdown services, and some connect two PBXes.

Typically, leased lines are used by businesses to connect geographically distant offices. Unlike dial-up connections, a leased line is always active. The fee for the connection is a fixed monthly rate. The primary factors affecting the monthly fee are distance between end points and the speed of the circuit. Because the connection doesn't carry anybody else's communications, the carrier can assure a given level of quality.

An internet leased line is a premium internet connectivity product, delivered over fiber normally, which is dedicated and provides uncontended, symmetrical speeds, Full Duplex. It is also known as an ethernet leased line, DIA line, data circuit or private circuit.

For example, a T-1 channel can be leased, and provides a maximum transmission speed of 1.544 Mbps. The user can divide the connection into different lines for multiplexing data and voice communication, or use the channel for one high speed data circuit. Increasingly, leased lines are being used by companies, and even individuals, for Internet access because they afford faster data transfer rates and are cost-effective for heavy users of the Internet.

History

Leased lines services (or private line services) became digital in the 1970s with the conversion of the Bell backbone network from analog to digital circuits . This conversion allowed AT&T to offer Dataphone Digital Services (later re-branded digital data services) that started the deployment of ISDN and T1 lines to customer premises to connect .

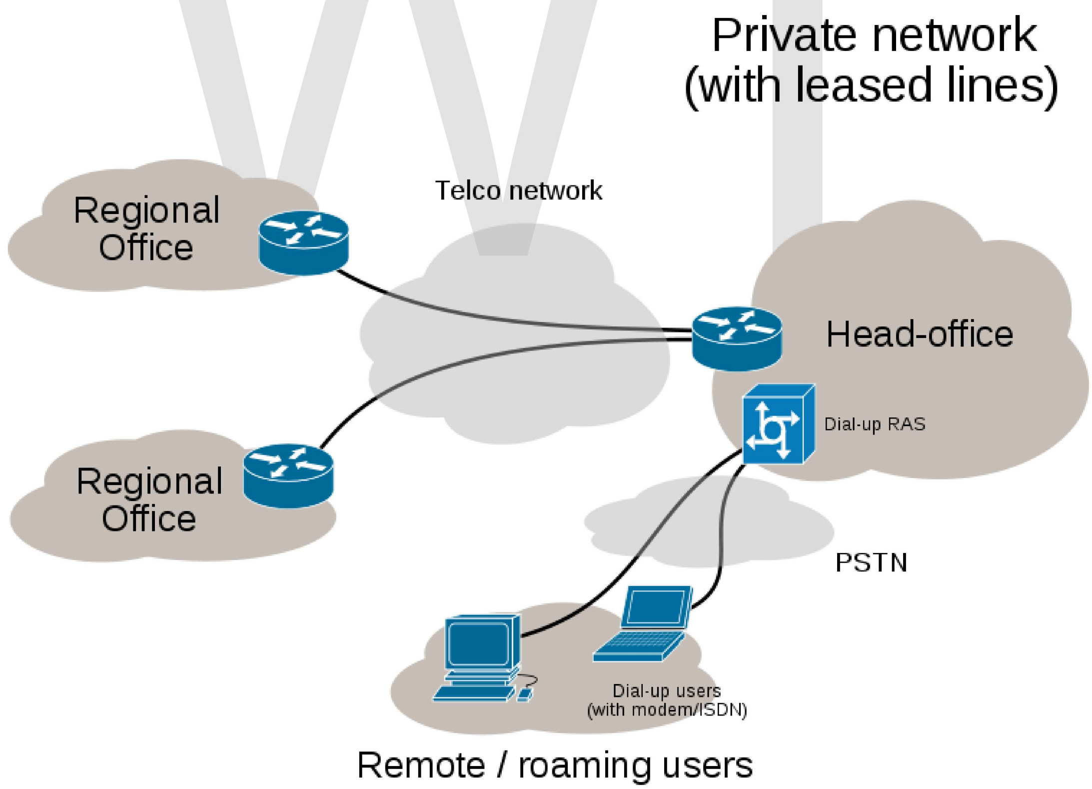
Leased lines were used to connect mainframe computers with terminals and remote sites, via IBM Systems Network Architecture (created in 1974) or DECnet (created in 1975).

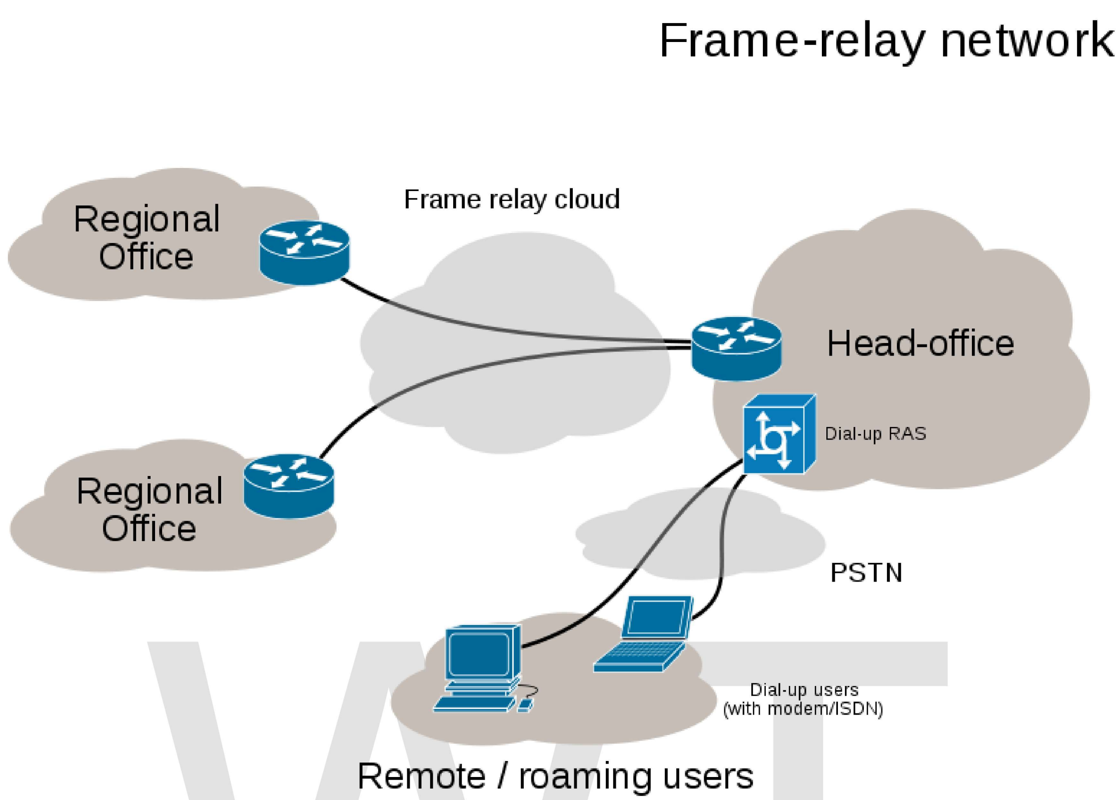
With the extension of digital services in the 1980s leased lines were used to connect customer premises to Frame Relay or ATM networks. Access data rates increased from the original T1 option up to T3 circuits.

In the 1990s with the advances of the Internet, leased lines were also used to connect customer premises to ISP Point of Presence's whilst the following decade saw a convergence of the aforementioned services (frame relay, ATM, Internet for businesses) with the MPLS integrated offerings.

Access data rates also evolved dramatically to speeds of up to 10Gb/s in the early 21st century with the Internet boom and increased offering in long-haul optical networks or Metropolitan Area Networks.

Applications





Leased lines are used to build up private networks, private telephone networks (by interconnecting PBX's) or access the internet or a partner network (extranet).

Here's is a review of the leased line applications in Network designs over time:

Site to site data connectivity

Terminating a leased line with two routers can extend network capabilities across sites. Leased lines were first used in the 1970s by enterprise with proprietary protocols such as IBM System Network Architecture and Digital Equipment DECnet, and with TCP/IP in University and Research networks before the Internet became widely available. Note that other Layer 3 protocols were used such as Novell IPX on enterprise networks until TCP/IP became ubiquitous in the 2000s. Today, point to point data circuits are typically provisioned as either TDM, Ethernet, or Layer 3 MPLS.

Site to site PBX connectivity

Terminating a leased line with two PBX allowed customers to by-pass PSTN for inter-site telephony. This allowed the customers to manage their own dial plan (and to use short extensions for internal telephone number) as well as to make significant savings if enough voice traffic was carried across the line (specially when the savings on the telephone bill exceeded the fixed cost of the leased line).

Site to network connectivity

As demand grew on data network telcos started to build more advanced network using packet switching on top of their infrastructure. Thus number of telecommunication companies added ATM, Frame-relay or ISDN offerings to their services portfolio. Leased lines were used to connect the customer site to the telco network access point.

International Private Lease Circuit

An IPLC is an International Private Leased Circuit that functions as a point-to-point private line. IPLCs are usually Time-division multiplexing (TDM) circuits that utilize the same circuit amongst many customers. The nature of TDM requires the use of a CSU/DSU and a router. Usually the router will include the CSU/DSU.

Then came the Internet (in the mid-1990s) and since the most common application for leased line is to connect a customer to its ISP Point of presence. With the changes that Internet brought in the networking world other technologies were developed to propose alternative to Frame-relay or ATM networks such as VPN's (hardware and software) and MPLS networks (that are in effect an upgrade to TCP/IP of existing ATM/Frame-relay infrastructures).

Availability

In the United Kingdom

In the U.K., leased lines are available at speeds from 64Kb/s increasing in 64Kb/s increments to 2.048Mb/s over a channelised E1 tail circuit and at speeds between 2.048Mb/s to 34.368Mb/s via channelised E3 tail circuits. The NTE will terminate the circuit and provide the requested presentation most frequently X.21 however higher speed interfaces are available such as G.703 or 10baseT. Some ISPs however use the term more loosely, defining a leased line as “any dedicated bandwidth service delivered over a leased fibre connection".

In the United States

In the U.S., low-speed leased lines (56 kbit/s and below) are usually provided using analog modems. Higher-speed leased lines are usually presented using **FT1** (**Fractional T1**): a **T1** bearer circuit with 1 to 24, 56k or 64k timeslots. Customers must manage their own network termination equipment—Channel Service Unit and Data Service Unit (CSU/DSU).

In Hong Kong

In Hong Kong, leased lines are usually available at speeds of 64k, 128k, 256k, 512k, T1 (channelized or not) or E1 (less common). Whatever the speed, telcos usually provide the CSU/DSU and present to the customer on V.35 interface.

In India

In India, leased lines are available at speeds of 64k, 128k, 256k, 512k,1mbps, 2mbps, 4mbps,8mbps,16mbps T1(1.544mbps) or E1(2.048mbps). Customers are connected either through OFC, telephone lines ADSL, or through Wifi. Customers would have to manage their own network termination equipment, namely the Channel service unit and Data service unit. All service providers gives a 99% uptime guarantee.

In Italy

In Italy, leased lines are available at speeds of 64k (terminated by DCE2 or DCE2plus modem) or multiple of 64k from 128k up to framed or unframed E1 (DCE3 modem) in digital form (PDH service, known as CDN, Circuito Diretto Numerico). Local TELCOs also may provide CDA (Circuito Diretto Analogico), that are plain copper dry pair between two buildings, without any line termination: in the past (pre-2002) a full analog base band was provided, giving an option to customer to deploy xDSL technology between sites: nowadays everything is limited at 4 kHz of bearer channel, so the service is just a POTS connection without any setup channel.

For many purposes, leased lines are gradually being replaced by DSL and metro Ethernet.

Leased line alternatives

Leased lines are more expensive than alternative connectivity services including (ADSL, SDSL, etc.) because they are reserved exclusively to the leaseholder. Some internet service providers have therefore developed alternative products that aim to deliver leased-line type services (Carrier Ethernet-based, zero contention, guaranteed availability), with more moderate bandwidth, over the standard UK national broadband network. While a leased line is full-duplex, most leased line alternatives provide only half-duplex or in many cases asymmetrical service.

Chapter- 4

Balanced Line and Balanced Circuit

Balanced line

In telecommunications and professional audio, a **balanced line** or **balanced signal pair** is a transmission line consisting of two conductors of the same type, each of which have equal impedances along their lengths and equal impedances to ground and to other circuits. The chief advantage of the balanced line format is good rejection of external noise. Common forms of balanced line are twin-lead, used for radio frequency signals and twisted pair, used for lower frequencies. They are to be contrasted to unbalanced lines, such as coaxial cable, which is designed to have its return conductor connected to ground, or circuits whose return conductor actually is ground. Balanced and unbalanced circuits can be interconnected using a transformer called a balun.

Circuits driving balanced lines must themselves be balanced to maintain the benefits of balance. This may be achieved by differential signaling, transformer coupling or by merely balancing the impedance in each conductor.

Lines carrying symmetrical signals (those with equal but opposite voltages to ground on each leg) are often referred to as balanced, but this is an entirely different meaning. The two conditions are not related.

Explanation

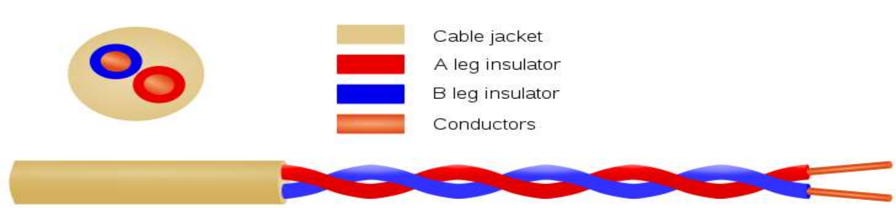


Fig. 1. Balanced line in twisted pair format. This line is intended for use with 2-wire circuits.

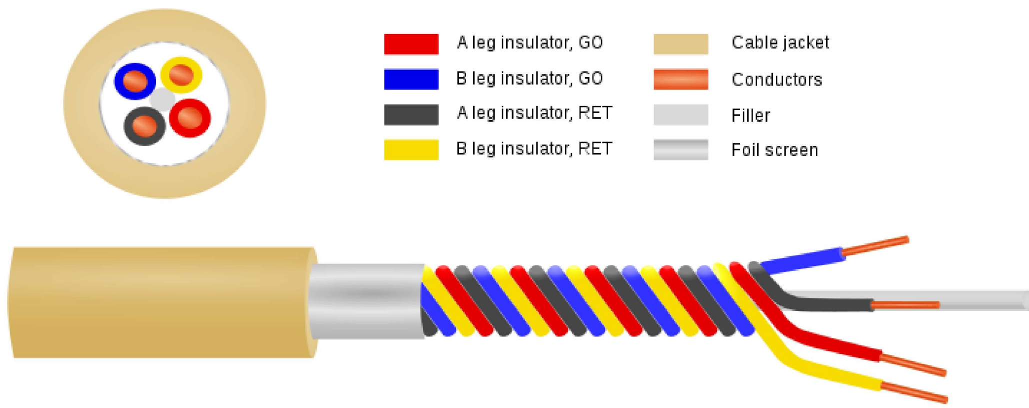


Fig. 2. Balanced line in star quad format. This line is intended for use with 4-wire circuits or two 2-wire circuits. It is also used with microphone signals in professional audio.

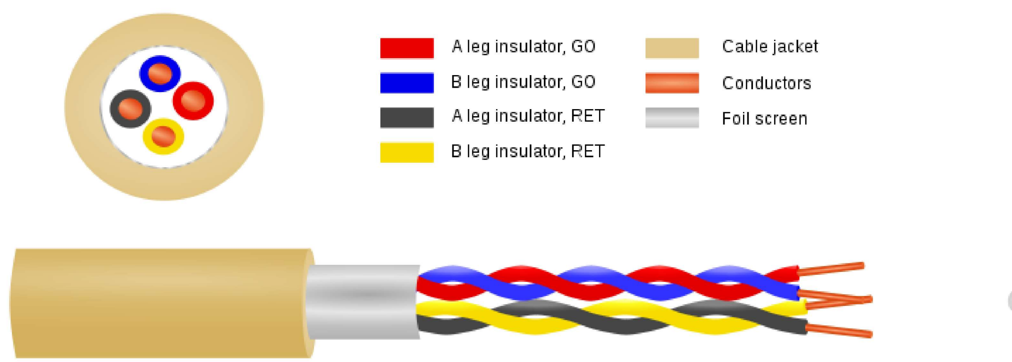


Fig. 3. Balanced line in DM quad format. This line is intended for use with 4-wire circuits or two 2-wire circuits.

Fig. 4. Balanced line in twin lead format. This line is intended for use with RF circuits, particularly antennae.

Transmission of a signal over a balanced line reduces the influence of noise or interference due to external stray electric fields. Any external signal sources tend to induce only a common mode signal on the line and the balanced impedances to ground minimizes differential pickup due to stray electric fields. The conductors are sometimes twisted together to ensure that each conductor is equally exposed to any external magnetic fields that could induce unwanted noise.

Some balanced lines also have electromagnetic shielding to reduce the amount of noise introduced.

A balanced line allows a differential receiver to reduce the noise on a connection by rejecting common-mode interference. The lines have the same impedance to ground, so the interfering fields or currents induce the same voltage in both wires. Since the receiver responds only to the difference between the wires, it is not influenced by the induced noise voltage. If twisted pair becomes unbalanced, for example due to insulation failure, noise will be induced. Examples of twisted pairs include Cat-3 Ethernet cables or telephone wires.

Compared to unbalanced circuits, balanced lines reduce the amount of noise per distance, allowing a longer cable run to be practical. This is because electromagnetic interference will affect both signals the same way. Similarities between the two signals are automatically removed at the end of the transmission path when one signal is subtracted from the other.

Telephone systems

The first application for balanced lines was for telephone lines. Interference that was of little consequence on a telegraph system (which is in essence digital) could be very disturbing for a telephone user. The initial format was to take two single-wire unbalanced telegraph lines and use them as a pair. This proved insufficient, however, with the growth of electric power transmission which tended to use the same routes. A telephone line running alongside a power line for many miles will inevitably have more interference induced in one leg than the other since one of them will be nearer to the power line. This issue was addressed by swapping the positions of the two legs every few hundred yards with a cross-over, thus ensuring that both legs had equal interference induced and allowing common-mode rejection to do its work. As the telephone system grew, it became preferable to use cable rather than open wires to save space, and also to avoid poor performance during bad weather. The format used for balanced telephone cables was twisted pair, however, this did not become widespread until repeater amplifiers became available. On an unamplified line cable could only manage a maximum distance of 30 km. Open wires, on the other hand, with their lower capacitance had been used for enormous distances - the longest was the 1500 km from New York to Chicago built in 1893. Loading coils were used to improve the distance achievable with cable but the problem was not finally overcome until amplifiers started to be installed in 1912. Twisted pair balanced lines are still widely used for the telephone subscribers local end.

Telephone trunk lines, and especially frequency division multiplexing carrier systems, are usually 4-wire circuits rather than 2-wire circuits (or at least they were before fibre-optic became widespread) and require a different kind of cable. This format requires the conductors to be arranged in two pairs, one pair for the sending (go) signal and the other for the return signal. The greatest source of interference on this kind of transmission is usually the crosstalk between the go and return circuits themselves. The most common cable format is star quad, where the diagonally opposite conductors form the pairs. This geometry gives maximum common mode rejection between the two pairs. An alternative format is DM quad which consists of two twisted pairs with the twisting at different pitches.

Audio systems

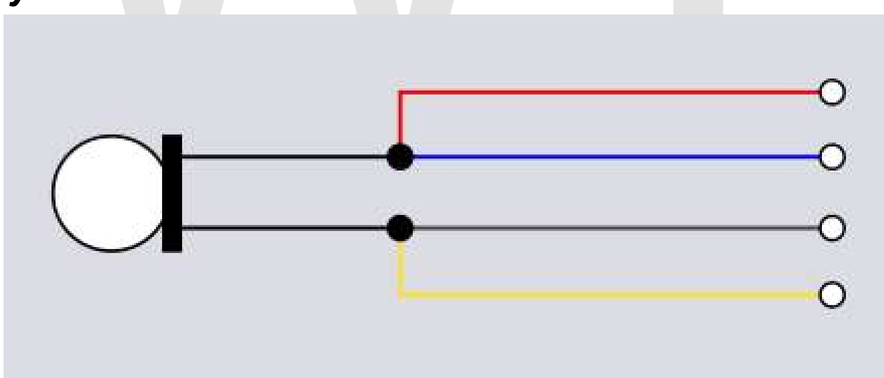


Fig. 5. Microphones connected to star quad cable join together diametrically opposite conductors to maintain balance. This is different from the usage on 4-wire circuits. The colours in this diagram correspond with the colouring in figure 2.

An example of balanced lines is the connection of microphones to a mixer in professional systems. Classically, both dynamic and condenser microphones used transformers to provide a differential-mode signal. While transformers are still used in the large majority of modern dynamic microphones, more recent condenser microphones are more likely to use electronic drive circuitry. Each leg, irrespective of any signal, should have an identical impedance to ground. Pair cable (or a pair-derivative such as star quad) is used to maintain the balanced impedances and close twisting of the cores ensures that any interference is common to both conductors. Providing that the receiving end (usually a mixing console) does not disturb the line balance, and is able to ignore common-mode (noise) signals, and can extract differential ones, then the system will have excellent immunity to induced interference.

Typical professional audio sources, such as microphones, have three-pin XLR connectors. One is the shield or chassis ground, while the other two are signal connections. These signal wires carry two copies of the same signal, but with opposite polarity. (They are often termed "hot" and "cold," and the AES14-1992(r2004) Standard [and EIA Standard RS-297-A] suggest that the pin that carries the positive signal that results from a positive air pressure on a transducer will be deemed 'hot'. Pin 2 has been designated as the 'hot' pin, and that designation serves useful for keeping a consistent polarity in the rest of the system.) Since these conductors travel the same path from source to destination, the assumption is that any interference is induced upon both conductors equally. The appliance receiving the signals compares the difference between the two signals (often with disregard to electrical ground) allowing the appliance to ignore any induced electrical noise. Any induced noise would be present in equal amounts and in identical polarity on each of the balanced signal conductors, so the two signals' difference from each other would be unchanged. The successful rejection of induced noise from the desired signal depends in part on the balanced signal conductors receiving the same amount and type of interference. This typically leads to twisted, braided, or co-jacketed cables for use in balanced signal transmission.

Balanced and differential

Most explanations of balanced lines assume symmetrical (antiphase) signals but this is an unfortunate confusion - signal symmetry and balanced lines are quite independent of each other. Essential in a balanced line is matched impedances in the driver, line and receiver. These conditions assure that external noise affects each leg of the differential line equally and thus appears as a common mode signal that is removed by the receiver. There are balanced drive circuits that have excellent common-mode impedance matching between "legs" but do *not* provide symmetrical signals. Symmetrical differential signals exist to prevent interference with *other* circuits - the electromagnetic fields are canceled out by the equal and opposite currents. But they are not necessary for interference rejection *from* other circuits.

Baluns

To convert a signal from balanced to unbalanced requires a balun. For example, baluns can be used to send line level audio or E-carrier level 1 signals over coaxial cable (which is unbalanced) through 300 feet (91 m) of Category 5 cable by using a pair of baluns at each end of the CAT5 run. The balun takes the unbalanced signal, and creates an inverted copy of that signal. It then sends these 2 signals across the CAT5 cable as a balanced signal. Upon reception at the other end, the balun takes the difference of the two signals, thus removing any noise picked up along the way and recreating the unbalanced signal.

A once common application of a radio frequency balun was found at the antenna terminals of a television receiver. Typically a 300-ohm balanced twin lead antenna input could only be connected to a coaxial cable from a cable TV system through a balun.

Characteristic Impedance

The characteristic impedance Z_0 of a transmission line is an important parameter at higher frequencies of operation. For a parallel 2-wire transmission line,

$$Z_0 = \frac{1}{\pi}\sqrt{\frac{\mu}{\epsilon}}\ln\left(\frac{l}{R}+\sqrt{\left(\frac{l}{R}\right)^2-1}\right),$$

where l is half the distance between the wire centres, R is the wire radius and μ, ε are respectively the permeability and permittivity of the surrounding medium. A commonly used approximation that is valid when the wire separation is much larger than the wire radius and in the absence of magnetic materials is

$$Z_0 = \frac{120}{\sqrt{\epsilon_r}}\ln\left(\frac{2l}{R}\right),$$

where ε_r is the relative permittivity of the surrounding medium.

Electric power lines

In electric power transmission, the three conductors used for three-phase power transmission are referred to as a balanced line since the instantaneous sum of the three line voltages is nominally zero. However, *balance* in this field is referring to the symmetry of the source and load: it has nothing to do with the impedance balance of the line itself, the sense of the meaning in telecommunications.

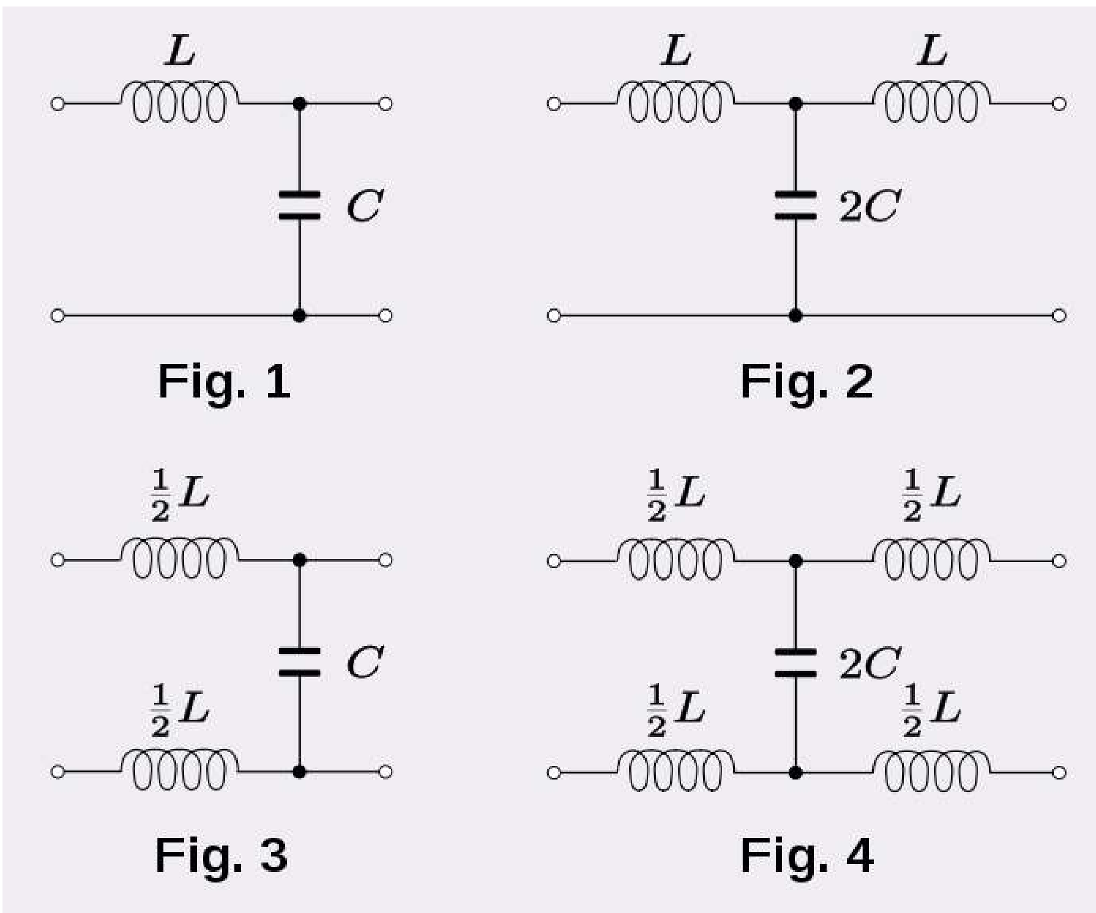
For the transmission of single-phase electric power as used for railway electrification systems, two conductors are used to carry in-phase and out-of-phase voltages such that the line is balanced.

Balanced circuit

A **balanced circuit** is circuitry for use with a balanced line or the balanced line itself. Balanced lines are a common method of transmitting many types of electrical communication signals between two points on two wires. In a balanced line the two signal lines are of a matched impedance to help ensure that interference induced in the line is common-mode and can be removed at the receiving end by circuitry with good common-mode rejection. To maintain the balance, circuit blocks which interface to the line, or are connected in the line, must also be balanced.

Balanced lines work because the interfering noise from the surrounding environment is induced into both wires equally. By measuring the difference between the two wires at the receiving end, the original signal is recovered while the noise is cancelled. Any inequality in the noise induced in each wire is an imbalance and will result in the noise not being fully cancelled. One requirement for balance is that both wires are an equal distance from the noise source. This is often achieved by placing the wires as close together as possible and twisting them together. Another requirement is that the impedance to ground (or to whichever reference point is being used by the difference detector) is the same for both conductors at all points along the length of the line. If one wire has a higher impedance to ground it will tend to have a higher noise induced, destroying the balance.

Balance and symmetry



Examples circuits using a low-pass filter to demonstrate. **Fig. 1.** Unbalanced, asymmetrical circuit. **Fig. 2.** Unbalanced, symmetrical circuit. **Fig. 3.** Balanced, asymmetrical circuit. **Fig. 4.** Balanced, symmetrical circuit.

A balanced circuit will normally show a symmetry of its components about a horizontal line mid-way between the two conductors (example in figure 3). This is different from what is normally meant by a symmetrical circuit which is a circuit showing symmetry of its components about a vertical line at its mid-point. An example of a symmetrical circuit is shown in figure 2. Circuits designed for use with balanced lines will often be designed to be both balanced and symmetrical as shown in figure 4. The advantages of symmetry are that the same impedance is presented at both ports and that the circuit has the same effect on signals travelling in both directions on the line.

Balance and symmetry are usually associated with reflected horizontal and vertical physical symmetry respectively as shown in figures 1 to 4. However, physical symmetry is not a necessary requirement for these conditions. It is only necessary that the electrical impedances are symmetrical. It is possible to design circuits that are not physically symmetrical but which have equivalent impedances which are symmetrical.

Balanced signals and balanced circuits

A balanced signal is one where the voltages on each wire are symmetrical with respect to ground (or some other reference). That is, the signals are inverted with respect to each other. A balanced circuit is a circuit where the two sides have identical transmission characteristics in all respects. A balanced line is a line in which the two wires will carry balanced currents (that is, equal and opposite currents) when balanced (symmetrical) voltages are applied. The condition for balance of lines and circuits will be met, in the case of passive circuitry, if the impedances are balanced. The line and ciruit remain balanced, and the benefits of common-mode noise rejection continue to apply, whether or not the applied signal is itself balanced (symmetrical), always provided that the generator producing that signal maintains the impedance balance of the line.

Driving and receiving circuits

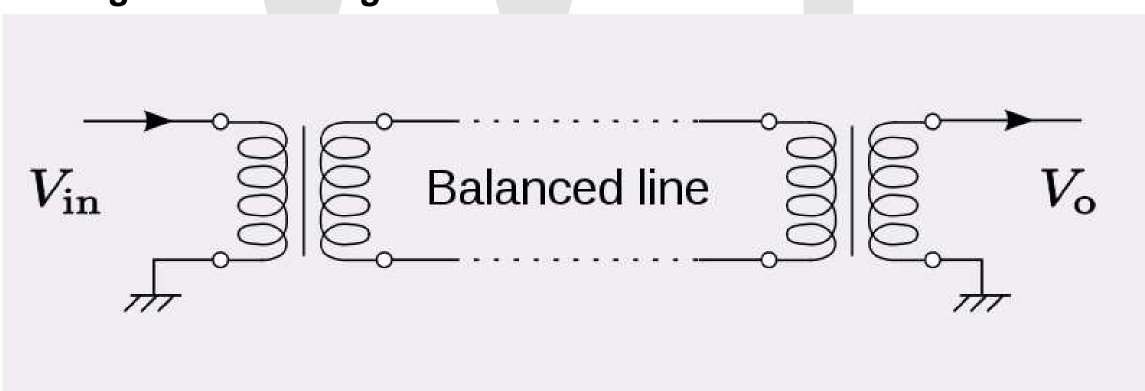


Fig. 5. Balanced line connected by transformers.

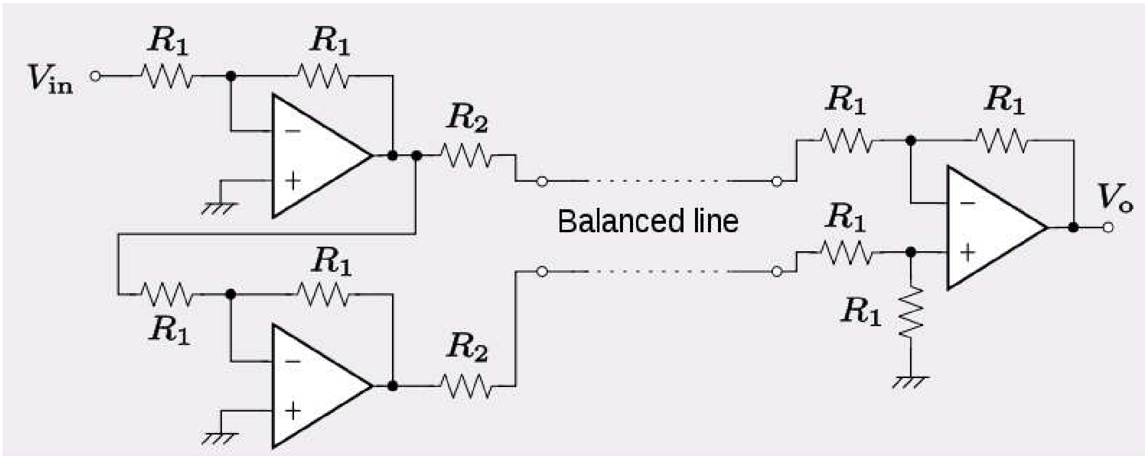


Fig. 6. Balanced line connected to electronically balanced circuitry.

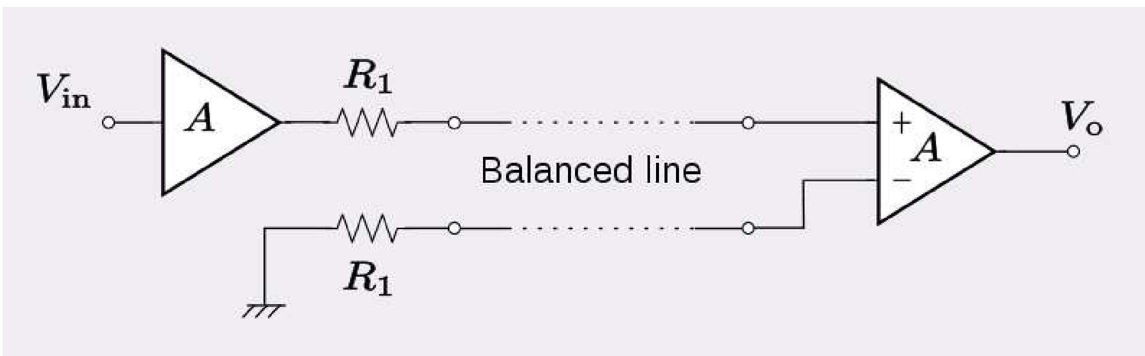


Fig. 7. Balanced line connected to an unbalanced signal, but with balanced impedances.

There are a number of ways that a balanced line can be driven and the signal detected. In all methods, for the continued benefit of good noise immunity, it is essential that the driving and receiving circuit maintain the impedance balance of the line. It is also essential that the receiving circuit detects only differential signals and rejects common-mode signals. It is not essential (although it is often the case) that the transmitted signal is balanced, that is, symmetrical about ground.

Transformer balance

The conceptually simplest way to connect to a balanced line is through transformers at each end shown in figure 5. Transformers were the original method of making such connections in telephony, and before the advent of active circuitry were the only way. In the telephony application they are known as repeating coils. Transformers have the additional advantage of completely isolating (or "floating") the line from earth and earth loop currents, which are an undesirable possibility with other methods, are completely eliminated.

The side of the transformer facing the line, in a good quality design, will have the winding laid in two parts (often with a centre tap provided) which are carefully balanced to maintain the line balance. Line side and equipment side windings are more useful concepts than the more usual primary and secondary windings when discussing these kinds of transformers. At the sending end the line side winding is the secondary, but at the receiving end the line side winding is the primary. When discussing a two-wire circuit primary and secondary cease to have any meaning at all, since signals are flowing in both directions at once.

The equipment side winding of the transformer does not need to be so carefully balanced. In fact,One leg of the equipment side can be earthed without effecting the balance on the line as shown in figure 5. With transformers the sending and receiving circuitry can be entirely unbalanced with the transformer providing the balancing.

Electronic balance

Electronic balance, or active balance, is achieved using differential amplifiers at each end of the line. An op-amp implementation of this is shown in figure 6, other circuitry is possible. Unlike transformer balance, there is no isolation of the circuitry from the line. Each of the two wires is driven by an op amp circuit which are identical except that one is inverting and one is non-inverting. Each one produces an unbalanced signal individually but together they drive the line with a symmetrical balanced signal. Because the currents in the two lines are equal and opposite, this has the further advantage that radiated signals cancel each other except in the near field of the conductors, thereby reducing cross-talk onto other conductors.

While it is not possible to create an isolated drive with op-amp circuitry alone, it is possible to create a floating output. This is important if one leg of the line might become grounded or connected to some other voltage reference. Grounding one leg of the line in the circuit of figure 6 will result in the line voltage being halved since only one op-amp is now providing signal. To achieve a floating output additional feedback paths are required between the two op-amps resulting in a more complex circuit than figure 6, but still avoiding the expense of a transformer. A floating op-amp output can only float within the limits of the op-amp's supply rails. An isolated output can be achieved without transformers with the addition of opto-isolators.

Impedance balance

As noted above, it is possible to drive a balanced line with an unbalanced signal and still maintain the line balance. This is represented in outline in figure 7. Amplifier A is assumed to be an ideal (that is, zero output impedance) unbalanced output amp. This is connected through a resistor to one leg of the line. The other leg is connected through another resistor of the same value. The impedance to ground of both legs is the same and the line remains balanced. The receiving amplifier still rejects any common-mode noise as it has a differential input. On the other hand the line signal is not balanced. The voltages at the input to the two legs, V_+ and V_- are given by;

$$V_+ = V_{\text{in}} \frac{Z_{\text{in}} + R_1}{Z_{\text{in}} + 2R_1}$$

$$V_- = V_{\text{in}} \frac{R_1}{Z_{\text{in}} + 2R_1}$$

Where Z_{in} is the input impedance of the line. These are clearly not symmetrical since V_- is much smaller than V_+. They are not even opposite polarities.

Balanced to unbalanced conversion

A circuit that has the specific purpose of converting between balanced and unbalanced formats is called a balun. A balun could be a transformer with one leg earthed on the unbalanced side as described in the transformer balance section above. Other circuits are possible such as autotransformers or active circuits.

Connectors

Common connectors used with balanced circuits include RJ-11 (telephone instruments), RJ-45 (broadband data) and XLR (professional audio) connectors. Also 1/4" tip ring sleeve (TRS) connectors were once widely used on manual switchboards and other telephone infrastructure. TRS connectors are now more commonly seen in miniature sizes (2.5 and 3.5 mm) being used for unbalanced stereo audio; however, some professional audio equipment such as mixing consoles still commonly use balanced "line-level" connections with 1/4" TRS jacks.

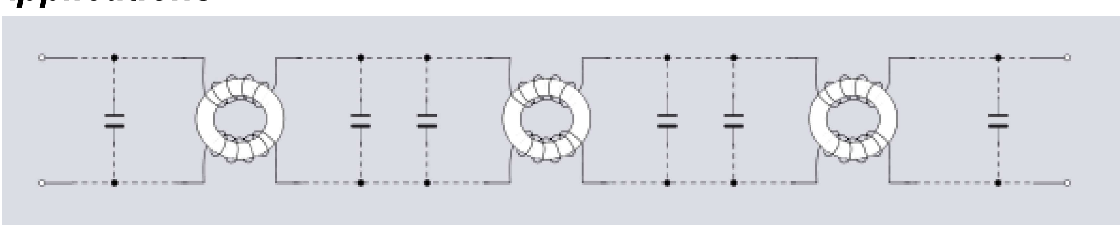
Chapter- 5

Loading Coil

In electronics, a **loading coil** or **load coil** is a coil (inductor) that does not provide coupling to any other circuit, but is inserted in a circuit to increase its inductance. The need was discovered by Oliver Heaviside in studying the disappointing slow speed of the Transatlantic telegraph cable. He concluded additional inductance was required to prevent amplitude and time delay distortion of the transmitted signal. The mathematical condition for distortionless transmission is known as the Heaviside condition. Previous telegraph lines were overland or shorter, hence had less delay and the need for extra inductance was not so great. Submarine communications cables are particularly subject to the problem, but early 20th century ones using balanced pairs were often continuously loaded by iron tape rather than discretely by load coils.

Loading coils are archaically known as **Pupin coils** after Mihajlo Pupin (especially when used for the Heaviside condition), and the process of inserting them is sometimes called *pupinization*.

Applications



Schematic of a balanced loaded line. The capacitors are shown connected with dotted lines to indicate that the capacitance is actually distributed along the line rather than the discrete elements shown. The windings of the loading coil are wound such that the magnetic flux induced in the core is in the same direction for both windings.

Voice circuits

A common application of loading coils is to improve the voice-frequency amplitude response characteristics of the twisted balanced pairs in a telephone cable.

Loading coils inserted periodically in series with a pair of wires reduce the attenuation at the higher voice frequencies up to the cutoff frequency of the low-pass filter formed by the inductance of the coils (plus the distributed inductance of the wires) and the distributed capacitance between the wires. Above the cutoff frequency, attenuation increases rapidly. The shorter the interval between the coils, the higher the cut-off frequency.

It should be emphasised that the cutoff effect is an artifact of using lumped inductors. With loading methods using continuous distributed inductance there is no cutoff.

Without loading coils, the line response is dominated by the resistance and capacitance of the line with the attenuation gently increasing with frequency. With loading coils of exactly the right inductance, neither capacitance nor inductance dominate: the response is flat, waveforms are undistorted and the characteristic impedance is resistive up to the cutoff frequency. The coincidental formation of an audio frequency filter is also beneficial in that noise is reduced.

DSL

When loading coils are in place, signal attenuation remains low for signals within the passband of the transmission line but increases rapidly for frequencies above the audio cutoff frequency. Thus, if the pair is subsequently reused to support applications that require higher frequencies (such as analog or digital carrier systems or DSL), any loading coils that were present on the line must be removed or replaced with one which is transparent to DSL. Using coils with parallel capacitors will form a filter with the topology of an m-derived filter and a band of frequencies above the cut-off will also be passed.

If they are not removed, as when the subscriber is an extended distance (e.g. over 4 miles) from the Central Office, DSL can not be supported. This sometimes happens in dense, growing areas (subject to frequent national numbering scheme repartitioning) such as Southern California in the late 1990s and early 21st century.

Carrier systems

American early and middle 20th Century telephone cables had load coils at intervals of a mile (1.61 km), usually in coil cases holding many. The coils must be removed to pass high frequencies, but the coil cases provided convenient places for repeaters for digital T-carrier systems, which could carry 1.5 Mbit/s across that distance. Due to narrower streets and higher cost of copper, European cables had thinner wires and needed closer intervals. Intervals of a kilometer allowed European systems to carry 2 Mbit/s.

Radio antennae

A (mobile) radio antenna, shorter than a quarter wavelength for practical reasons, presents capacitive reactance to a transmission line. This can be canceled by inserting an equal and opposite (inductive) reactance in series, by means of a loading coil typically at the base or center of the antenna. Consequently the antenna presents a resistance (desirable) to the transmission line.

Campbell equation

The Campbell equation is a relationship due to George Ashley Campbell for predicting the propagation constant of a loaded line. It is stated as;

$$\cosh(\gamma' d) = \cosh(\gamma d) + \frac{Z}{2Z_0}\sinh(\gamma d)$$

where,
γ is the propagation constant of the unloaded line
γ' is the propagation constant of the loaded line
d is the interval between coils on the loaded line
Z is the impedance of a loading coil and
Z_0 is the characteristic impedance of the unloaded line.

A more engineer friendly rule of thumb is that the approximate requirement for spacing loading coils is ten coils per wavelength of the maximum frequency being transmitted. This approximation can be arrived at by treating the loaded line as a constant k filter and applying image filter theory to it. From basic image filter theory the angular cutoff frequency and the characteristic impedance of a low-pass constant k filter are given by;

$$\omega_c = \frac{1}{\sqrt{L_{\frac{1}{2}}C_{\frac{1}{2}}}} \text{ and, } Z_0 = \sqrt{\frac{L_{\frac{1}{2}}}{C_{\frac{1}{2}}}}$$

where $L_{\frac{1}{2}}$ and $C_{\frac{1}{2}}$ are the half section element values.

From these basic equations the necessary loading coil inductance and coil spacing can be found;

$$L = \frac{Z_0}{\omega_c} \text{ and, } d = \frac{2}{\omega_c Z_0 C}$$

where C is the capacitance per unit length of the line.

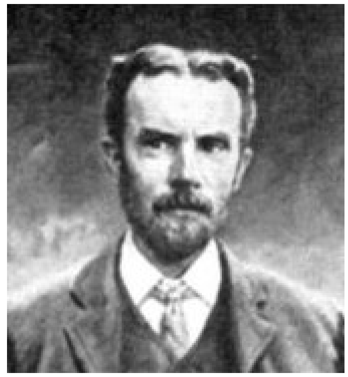
Expressing this in terms of number of coils per cutoff wavelength yields;

$$\frac{\lambda_c}{d} = \pi v Z_0 C$$

where *v* is the velocity of propagation of the cable in question.

History

Oliver Heaviside



Oliver Heaviside

The origin of the loading coil can be found in the work of Oliver Heaviside on the theory of transmission lines. Heaviside (1881) represented the line as a network of infinitesimally small circuit elements. By applying his operational calculus to the analysis of this network he discovered (1887) what has become known as the Heaviside condition. This is the condition that must be fulfilled in order for a transmission down a line to be free from distortion. The Heaviside condition is that the line series impedance, Z, must be proportional to the line shunt admittance, Y, at all frequencies. In terms of the primary line coefficients this is the condition;

$$\frac{R}{G} = \frac{L}{C}$$

where;
R is the series resistance of the line per unit length
L is the series self-inductance of the line per unit length
G is the shunt leakage conductance of the line insulator per unit length
C is the shunt capacitance between the line conductors per unit length

Heaviside was aware that this condition was not met in the practical telegraph cables in use in his day. In general, a real cable would have,

$$\frac{R}{G} \gg \frac{L}{C}$$

This is mainly due to the low value of leakage through the cable insulator, which is even more pronounced in modern cables which have better insulators than in Heaviside's day. In order to meet the condition, the choices are therefore to try and increase G or L or to decrease R or C. Decreasing R requires larger conductors. Copper was already in use in telegraph cables and this is the very best conductor available short of using silver. Decreasing R means using more copper and a more expensive cable. Decreasing C would also mean a larger cable (although not necessarily more copper). Increasing G is highly undesirable, while it would reduce distortion, it would at the same time increase the signal loss. Heaviside considered, but rejected, this possibility which left him with the strategy of increasing L as the way to reduce distortion.

Heaviside immediately (1887) proposed several methods of increasing the inductance, including spacing the conductors further apart and loading the insulator with iron dust. Finally, Heaviside made the proposal (1893) to use discrete inductors at intervals along the line. However, he never succeeded in persuading the British GPO to take up the idea. Brittain attributes this to Heaviside's failure to provide engineering details on the size and spacing of the coils for particular cable parameters. Heaviside's eccentric character and setting himself apart from the establishment may also have played a part in their ignoring of him.

John Stone

John S. Stone worked for the American Telephone & Telegraph Company (AT&T) and was the first to attempt to apply Heaviside's ideas to real telecommunications. Stone's idea (1896) was to use a bimetallic iron-copper cable which he had patented. This cable of Stone's would increase the line inductance due to the iron content and had the potential to meet the Heaviside condition. However, Stone left the company in 1899 and the idea was never implemented.

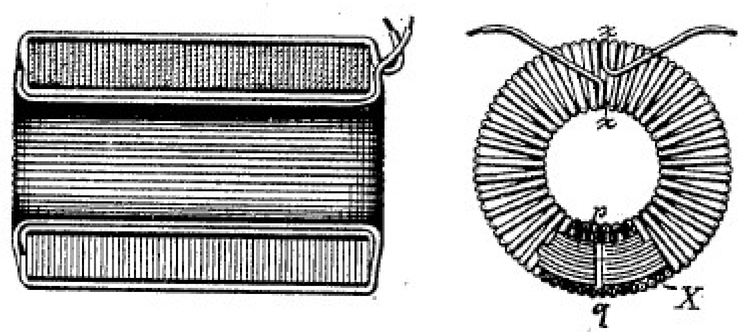
George Campbell

George Campbell was another AT&T engineer working for them in their Boston facility. Campbell was tasked with continuing the investigation into Stone's bimetallic cable, but soon abandoned this in favour of the loading coils idea. This was an independent discovery, Campbell being aware of Heaviside's work in discovering the Heaviside condition, but apparently not aware of Heaviside's suggestion of using loading coils to force a line to meet it. The motivation for the change of direction was Campbell's limited budget.

Campbell was struggling to set up a practical demonstration over a real telephone route with the budget he had been allocated. After considering that his artificial line simulators used lumped components rather than the distributed quantities found in a real line, he wondered if he could not insert the inductance with lumped components instead of using Stone's distributed line. When his calculations showed that the manholes on telephone routes were sufficiently close together to be able to insert the loading coils without the expense of either having to dig up the route or lay in new cables he changed to this new plan. The very first demonstration of loading coils on a telephone cable was on a 46-mile length of the so-called Pittsburgh cable (the test was actually in Boston, the cable had previously been used for testing in Pittsburgh) on September 6, 1899 carried out by Campbell himself and his assistant. The first telephone cable using loaded lines put into public service was between Jamaica Plain and West Newton in Boston on May 18, 1900.

Campbell's work on loading coils provided the theoretical basis for his subsequent work on filters which proved to be so important for frequency-division multiplexing. The cut-off phenomena of loading coils, an undesirable side-effect, can be exploited to produce a desirable filter frequency response.

Michael Pupin



Pupin's design of loading coil

Michael Pupin, inventor and Serbian immigrant to the USA, also played a part in the story of loading coils. Pupin filed a rival patent to the one of Campbell's. This patent of Pupin's dates from 1899. There is an earlier patent (1894, filed December 1893) which is sometimes cited as Pupin's loading coil patent but is, in fact, something different. The confusion is easy to understand, Pupin himself claims that he first thought of the idea of loading coils while climbing a mountain in 1894, although there is nothing from him published at that time.

Pupin's 1894 patent "loads" the line with capacitors rather than inductors, a scheme that has been criticised as being theoretically flawed and never put into practice. To add to the confusion, one variant of the capacitor scheme proposed by Pupin does indeed have coils. However, these are not intended to compensate the line in any way. They are there merely to restore DC continuity to the line so that it may be tested with regular equipment. Pupin states that the inductance is to be so large that it will block all AC signals above 50 Hz. Consequently, only the capacitor is adding any significant impedance to the line and "the coils will not exercise any material influence on the results before noted".

Legal battle

Heaviside never patented his idea; indeed, he took no commercial advantage of any of his work. Despite the legal disputes surrounding this invention, it is unquestionable that Campbell was the first to actually construct a telephone circuit using loading coils. There also can be little doubt that Heaviside was the first to publish and many would dispute Pupin's priority.

AT&T fought a legal battle with Pupin over his claim. Pupin was first to patent but Campbell had already conducted practical demonstrations before Pupin had even filed his patent (December 1899). Campbell's delay in filing was due to the slow internal machinations of AT&T.

However, AT&T foolishly deleted from Campbell's proposed patent application all the tables and graphs detailing the exact value of inductance that would be required before the patent was submitted. Since Pupin's patent contained a (less accurate) formula, AT&T was open to claims of incomplete disclosure. Fearing that there was a risk that the battle would end with the invention being declared unpatentable due to Heaviside's prior publication, they decided to desist from the challenge and buy an option on Pupin's patent for a yearly fee so that AT&T would control both patents. By January 1901 Pupin had been paid $200,000 and by 1917, when the AT&T monopoly ended and payments ceased, he had received a total of $455,000.

Benefit to AT&T

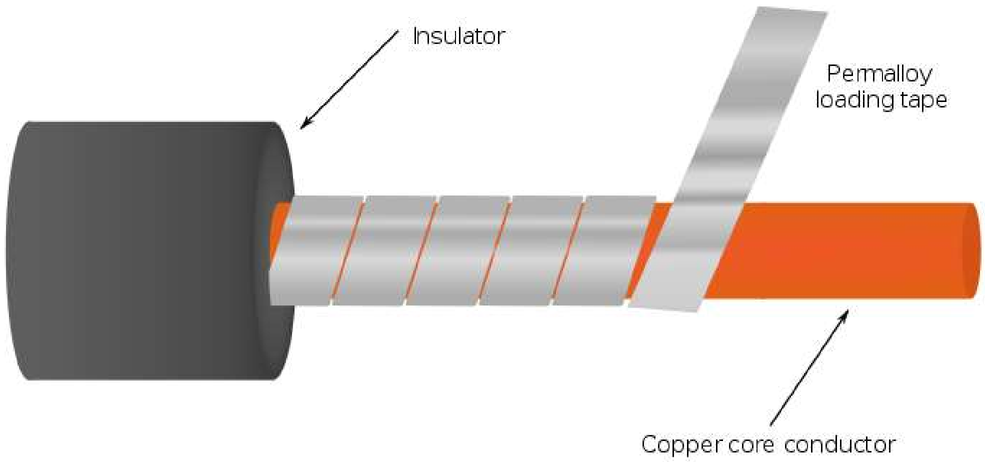
The invention was of enormous value to AT&T. Telephone cables could now be used to twice the distance previously possible, or alternatively, a cable of half the previous quality (and cost) could be used over the same distance. When considering whether to allow Campbell to go ahead with the demonstration, their engineers had estimated that they stood to save $700,000 in new installation costs in New York and New Jersey alone. It has been estimated that AT&T saved $100 million in the first quarter of the 20th century. Heaviside, who began it all, came away with nothing. He was offered a token payment but would not accept, wanting the credit for his work. He remarked ironically that if his prior publication had been admitted it would "interfere . . . with the flow of dollars in the proper direction . . .".

Krarup cable

Loading coils were not without their problems. For submarine cables where they were of most benefit, they were difficult to lay. The cable needed to be heavier and both this and the discontinuities in the profile where the coils occurred caused stresses in the cable during laying. Without great care, the cable might part and would be enormously expensive, possibly impossible, to fix. A second problem was that the material science of the time had difficulties sealing the joint between coil and cable against ingress of seawater. When this occurred, of course, the cable was ruined.

A Danish engineer, Carl Emil Krarup, invented a form of continuously loaded cable which solved these problems and the cable is named for him. Krarup cable has iron wires continuously wound around the central copper conductor with adjacent turns in contact with each other. This cable was the first use of continuous loading on any telecommunication cable. In 1902 Krarup both wrote his paper on this subject and saw the installation of the first cable between Helsingør (Denmark) and Helsingborg (Sweden).

Permalloy cable

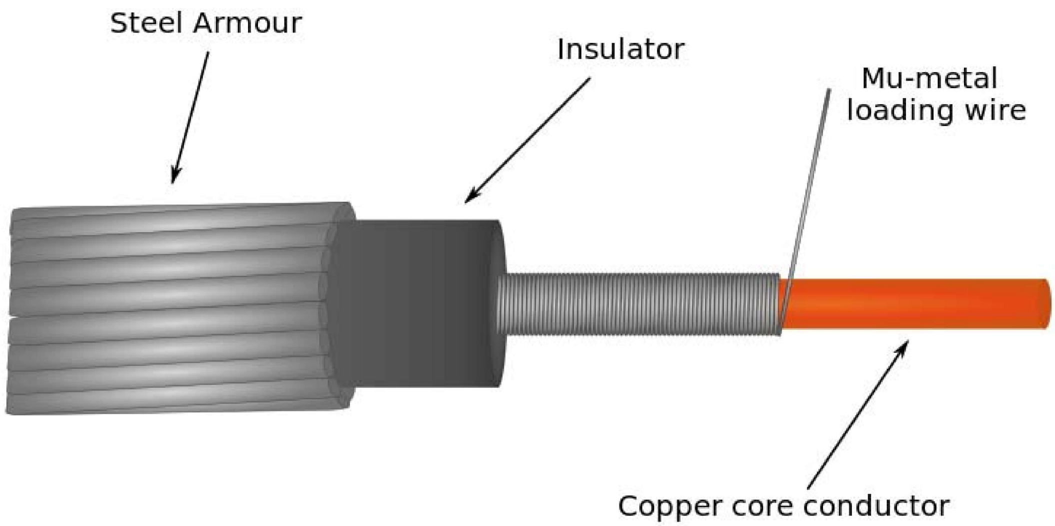


Permalloy cable construction

Even though Krarup cable added inductance to the line, it did not add enough to meet the Heaviside condition. AT&T searched for a better material with higher magnetic permeability. In 1914 Gustav Elmen discovered permalloy, a magnetic nickel-iron annealed alloy. Oliver E. Buckley, along with his colleagues at Bell Labs, H. D. Arnold and Elmen, c.1915 proposed a method of constructing submarine cable using permalloy tape wrapped around the copper conductors. This construction greatly improved the performance of the cable.

The cable was tested in a trial in Bermuda in 1923. The first permalloy cable to be put into service was between New York and Horta (Azores) in September 1924.

Mu-metal cable



Mu-metal cable construction

Mu-metal has similar magnetic properties to permalloy but the addition of copper to the alloy increases the ductility and allows the metal to be drawn into wire. Mu-metal cable is easier to construct than permalloy cable, the mu-metal being wound around the core copper conductor in much the same way as the iron wire in Krarup cable. A further advantage with mu-metal cable is that the construction lends itself to a variable loading profile whereby the loading is tapered towards the ends.

Mu-metal was invented (1923) by The Telegraph Construction and Maintenance Company Ltd., London, who made the cable, initially, for the Western Union Telegraph Co. Western Union were in competition with AT&T and the Western Electric Company who were using permalloy (the patent for permalloy was held by Western Electric).

Current practice

Loaded cable is no longer a useful technology for submarine communication cables, having first been superseded by co-axial cable using electrically powered in-line repeaters and then by fibre-optic cable. Manufacture of loaded cable declined in the 1930s and was then superseded by other technologies post-war. Loading coils can still be found in some telephone landlines today but new installations would use more modern technology.

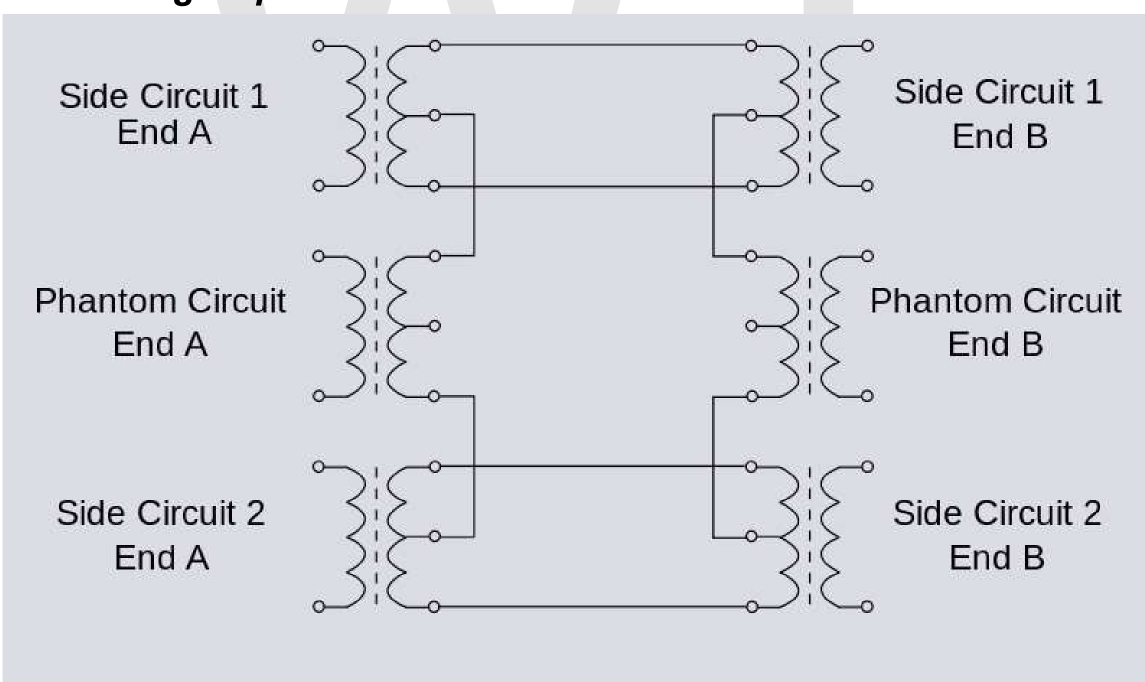
Chapter- 6

Phantom Circuit and Frequency Mixer

Phantom circuit

In telecommunication and electrical engineering, a **phantom circuit** is an electrical circuit derived from suitably arranged wires with one or more conductive paths being a circuit in itself and at the same time acting as one conductor of another circuit.

Phantom group



Phantom circuit derived from two subscriber circuits

A **phantom group** is composed of three circuits that are derived from two single-channel circuits to form a *phantom circuit*. Here the phantom circuit is a third circuit derived from two suitably arranged pairs of wires, called side circuits, with each pair of wires being a circuit in itself and at the same time acting as one conductor of the third circuit. The "side circuits" within phantom circuits can be coupled to their respective voltage drops by center-tapped transformers, usually called "repeating coils". The center taps are on the line side of the side circuits. Current from the phantom circuit is split evenly by the center taps. This cancels crosstalk from the phantom circuit to the side circuits.

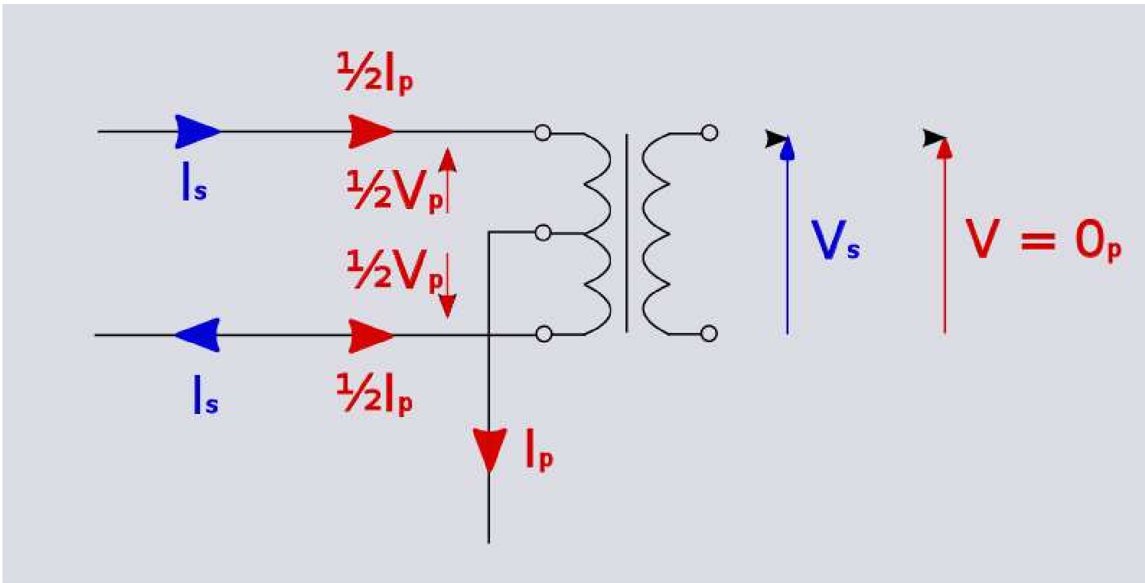


Diagram showing how the phantom currents (red) cancel in the transformer. Side circuit currents (blue) do not cancel and are transmitted through the transformer.

Phantom working increased the number of circuits on long distance routes in the early 20th century without putting up more wires. Phantoming declined with the adoption of carrier systems.

It is theoretically possible to create a phantom circuit from two other phantom circuits and so on up in a pyramid with a maximum 2n-1 circuits being derived from n original circuits. However, more than one level of phantoming is usually impractical. Isolation between the phantom circuit and the side circuits relies on accurate balance of the line and transformers. Imperfect balance results in crosstalk between the phantom and side circuits and this effect accumulates as each level of phantoms is added. Even small levels of crosstalk are unacceptable on analogue telecommunications circuits since speech crosstalk is still intelligible down to quite low levels.

Phantom microphone powering

Recording and broadcast studios commonly use phantom powering as a means to provide power to microphones. Power may be needed either for a device that requires power such as a pre-amp on an electret microphone or because the microphone is a type that intrinsically requires powering such as a condenser microphone. Since the microphone has only one pair of wires the return path for the power has to be provided elsewhere. This is usually done via the microphone cable screen.

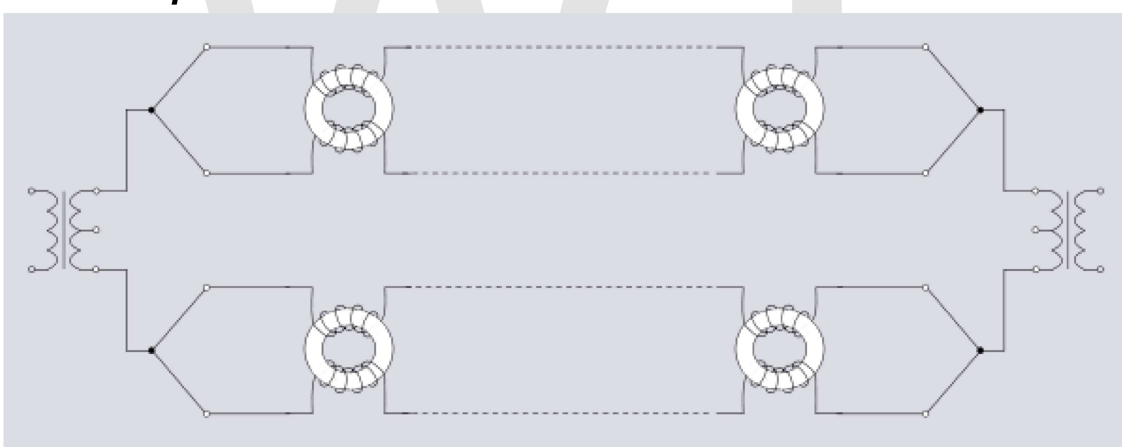
DC phantom

Simple DC signalling can be achieved on a telecommunications line in a similar way to phantom powering of microphones. A switch connected to the transformer centre-tap at one end of the line can operate a similarly connected relay at the other end. The return path is through the ground connection. This arrangement can be used for remotely controlling equipment.

Carrier circuit phantoms

From the 1950s to around the 1980s, using phantoms on star-quad trunk carrier circuits was a popular method of deriving a high quality broadcast audio circuit. The multiplexed FDM telecommunications carrier system usually did not use the baseband of the cable because it was inconvenient to separate low frequencies with filters. On the other hand, a one-way audio phantom could be formed from the two pairs (go and return signals) making up the star-quad cable.

Unloaded phantom



Unloaded phantom configuration. The windings of the loading coil are wound such that the magnetic flux induced in the core is normally in the same direction for both windings. However, in the phantom configuration the flux cancels.

Unloaded phantom is a phantom configuration of loaded lines (a circuit fitted with loading coils). The idea here is not to create additional circuits. Rather, the purpose is to cancel or greatly reduce the effect of the loading coils fitted to a line. The reason for doing this is that loaded lines have a definite cut-off frequency and it may be desired to equalise the line to a frequency which is higher than this, for example to make a circuit suitable for use by a broadcaster. Ideally, the loading would be removed or reduced for a permanent connection, but this is not feasible for temporary arrangements such as a requirement for outside broadcast. Instead, two circuits in a phantom configuration can be used to greatly reduce the inductance being inserted by the loading coils, and hence the loading effect.

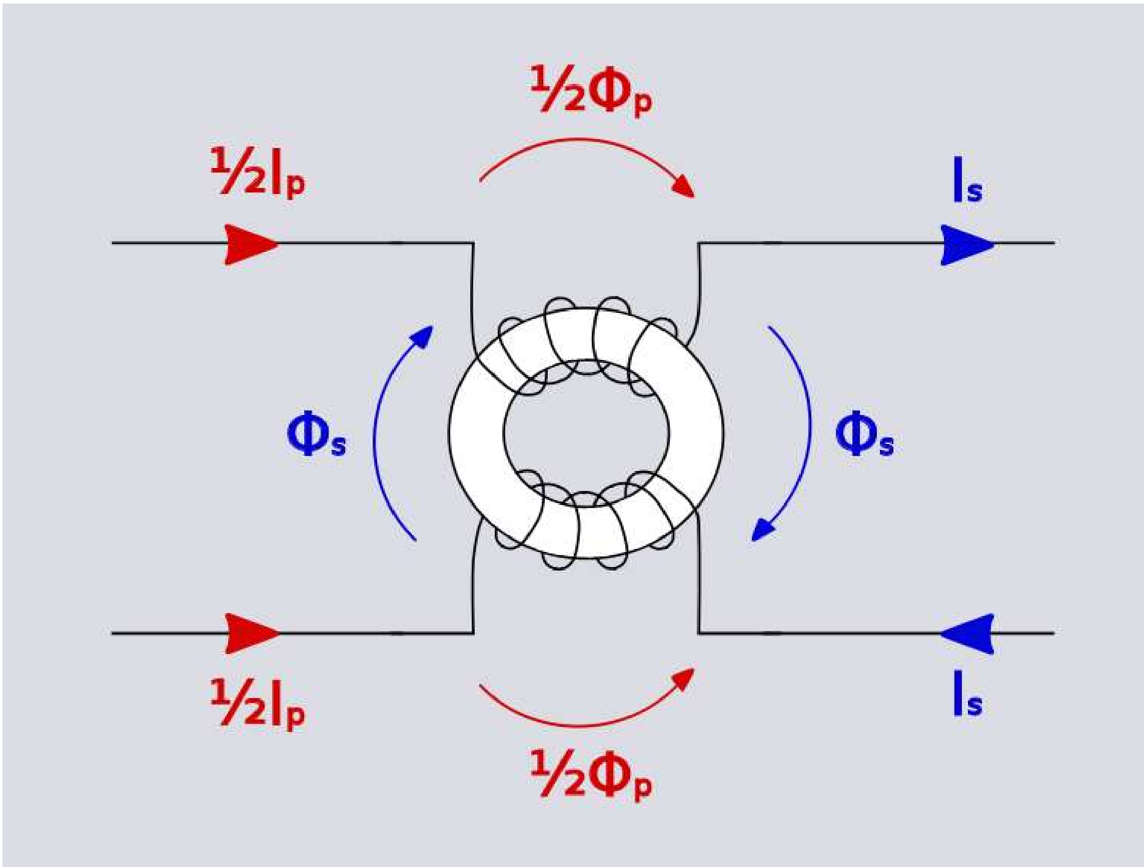


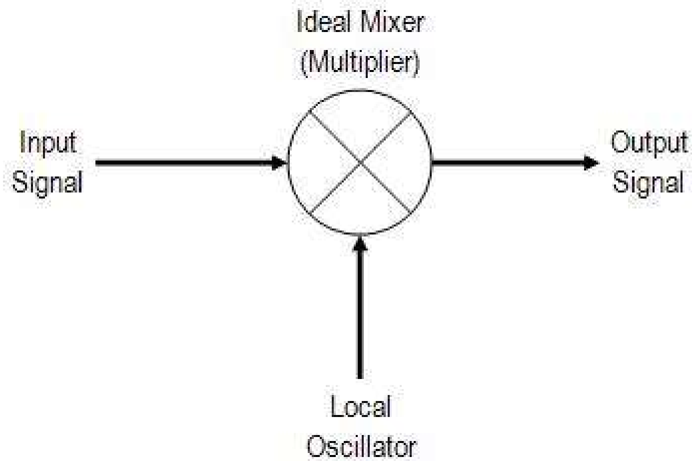
Diagram showing how the flux due to the phantom currents (red) is cancelled in the load coil. Flux due to normal line currents (blue) is additive.

It works because the loading coils used on balanced lines have two windings, one for each leg of the circuit. They are both wound on a common core and the windings are so arranged that the magnetic flux induced by both of them is in the same direction. Both windings induce an emf in each other as well as their own self-induction. This effect greatly increases the inductance of the coil and hence its loading effectiveness. By contrast, when the circuit is in the phantom configuration the currents in the two wires of each pair are in the same direction and the magnetic flux is being cancelled. This has precisely the opposite effect and the inductance is greatly reduced.

This configuration is most commonly used on the two pairs of a star-quad cable. It is not so successful with other pairs of wires. The difference in the path of the two pairs can easily destroy the balance and results in crosstalk and interference.

This configuration can also be called "bunched pairs". However, "bunched pairs" can also refer to the straightforward connection of two lines in parallel which is not a phantom circuit and will not reduce the loading.

Frequency mixer



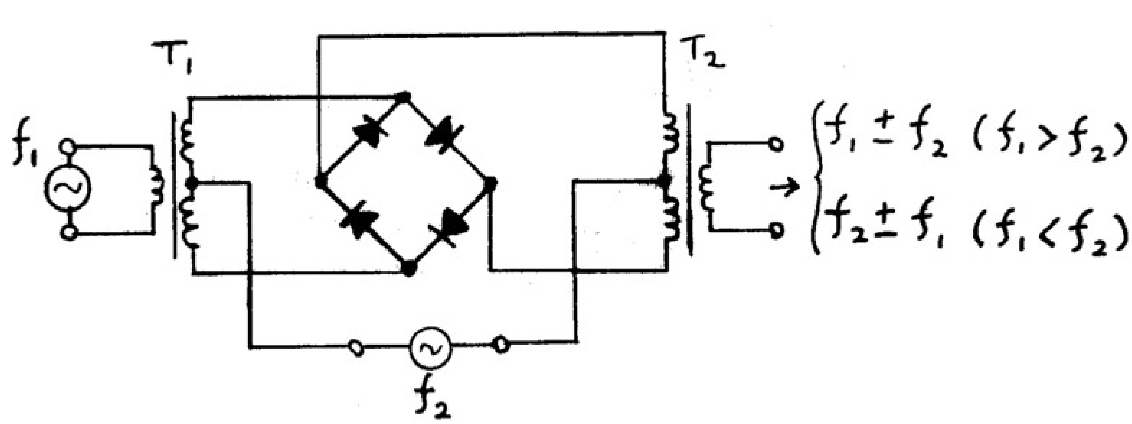
Frequency Mixer Symbol.

In electronics a **mixer** or **frequency mixer** is a nonlinear electrical circuit that creates new frequencies from two signals applied to it. In its most common application, two signals at frequencies f_1 and f_2 are applied to a mixer, and it produces new signals at the sum $f_1 + f_2$ and difference $f_1 - f_2$ of the original frequencies. Other frequency components may also be produced in a practical frequency mixer.

Mixers are widely used to shift signals from one frequency range to another, a process known as heterodyning, for convenience in transmission or further signal processing. For example, a key component of a superheterodyne receiver is a mixer used to move received signals to a common intermediate frequency. Frequency mixers are also used to modulate a carrier frequency in radio transmitters.

Types

Passive mixers use one or more diodes and rely on the non-linear relation between voltage and current to provide the multiplying element. In a passive mixer, the desired output signal is always of lower power than the input signals. Active mixers can increase the strength of the product signal. Active types improve isolation between the ports, but may have higher noise and more power consumption and be less tolerant of overload. Mixers may be built of discrete components, may be part of integrated circuits, or can be delivered as hybrid modules. Mixers may also be classified by their topology. Unbalanced mixers allow some of the input signal power to pass through to the output. A single-balanced mixer is arranged so that the local oscillator (or RF) signal port cancels. A doubly-balanced mixer has symmetrical paths for both inputs, and will have no output if either input signal is not present.



Schematic diagram of a double-balanced passive diode mixer. There is no output unless both f1 and f2 inputs are present.

Selection of a mixer type is a trade off for a particular application. Mixer circuits are characterized by conversion gain, and noise figure. Balanced and double-balanced designs allow less of the input signals to feed through to the output.

Nonlinear electronic components that are used as mixers include diodes, transistors biased near cutoff, and at lower frequencies, analog multipliers. Ferromagnetic-core inductors driven into saturation have also been used. In nonlinear optics, crystals with nonlinear characteristics are used to mix two frequencies of laser light to create optical heterodynes.

Diode

A diode can be used to create a simple mixer. The importance of the diode is that it is non-linear (or non-Ohmic), which means its response (current) is not proportional to its input (voltage). The diode therefore does not reproduce the frequencies of its driving voltage in the current through it, which allows the desired frequency manipulation. Certain other non-linear devices could be utilized similarly.

The current *I* through an ideal diode as a function of the voltage *V* across it is given by

$$I = I_S \left(e^{\frac{qV_D}{nkT}} - 1 \right)$$

where what is important is that *V* appears in *e*'s exponent. The exponential can be expanded as

$$e^x = \sum_{n=0}^{\infty} \frac{x^n}{n!}$$

and can be approximated for small x (that is, small voltages) by the first few terms of that series:

$$e^x - 1 \approx x + \frac{x^2}{2}$$

Suppose that the sum of the two input signals $v_1 + v_2$ is applied to a diode, and that an output voltage is generated that is proportional to the current through the diode (perhaps by providing the voltage that is present across a resistor in series with the diode). Then, disregarding the constants in the diode equation, the output voltage will have the form

$$v_o = (v_1 + v_2) + \frac{1}{2}(v_1 + v_2)^2 + \ldots$$

The first term on the right is the original two signals, as expected, followed by the square of the sum, which can be rewritten as $(v_1 + v_2)^2 = v_1^2 + 2v_1v_2 + v_2^2$, where the multiplied signal is obvious. The ellipsis represents all the higher powers of the sum which we assume to be negligible for small signals.

Switching

Another form of mixer operates by switching, with the smaller input signal being passed inverted or uninverted according to the phase of the local oscillator (LO). This would be typical of the normal operating mode of a packaged double balanced mixer module such as an SBL-1, with the local oscillator drive considerably higher than the signal amplitude.

The aim of a switching mixer is to achieve linear operation over the signal level, and hard switching driven by the local oscillator. Mathematically the switching mixer is not much different from a multiplying mixer, just because instead of the LO sine wave term we would use the signum function. In the frequency domain the switching mixer operation leads to the usual sum and difference frequencies, but also to further terms e.g. +-3*fLO, +-5*fLO, etc. The advantage of a switching mixer is that it can achieve - with the same effort - a lower noise figure (NF) and larger conversion gain. This come because the switching diodes or transistors act either like a low resistor (switch closed) or large resistor (switch open) and in both cases only minimum noise is added. From the circuit perspective many multiplying mixers can be used as switching mixers, just by increasing the LO amplitude. So RF engineers simply talk about mixers, and mean switching mixers.

Applications

The mixer circuit can be used not only to shift the frequency of an input signal as in a receiver, but also as a product detector, modulator, phase detector or frequency multiplier. For example a communications receiver might contain two mixer stages for conversion of the input signal to an intermediate frequency, and another mixer employed as a detector for demodulation of the signal.

Chapter- 7

Phase-locked Loop

A **phase-locked loop** or **phase lock loop** (PLL) is a control system that tries to generate an output signal whose phase is related to the phase of the input "reference" signal. It is an electronic circuit consisting of a variable frequency oscillator and a phase detector. This circuit compares the phase of the input signal with the phase of the signal derived from its output oscillator and adjusts the frequency of its oscillator to keep the phases matched. The signal from the phase detector is used to control the oscillator in a feedback loop.

Frequency is the derivative of phase. Keeping the input and output phase in lock step implies keeping the input and output frequencies in lock step. Consequently, a phase-locked loop can track an input frequency, or it can generate a frequency that is a multiple of the input frequency. The former property is used for demodulation, and the latter property is used for indirect frequency synthesis.

Phase-locked loops are widely used in radio, telecommunications, computers and other electronic applications. They may generate stable frequencies, recover a signal from a noisy communication channel, or distribute clock timing pulses in digital logic designs such as microprocessors. Since a single integrated circuit can provide a complete phase-locked-loop building block, the technique is widely used in modern electronic devices, with output frequencies from a fraction of a hertz up to many gigahertz.

Practical analogies

Automobile race analogy

For a practical idea of what is going on, consider an auto race. There are many cars, and each of them wants to go around the track as fast as possible. Each lap corresponds to a complete cycle, and each car will complete dozens of laps per hour. The number of laps per hour (a speed) is a frequency, but the number of laps (a distance) corresponds to a phase. At one instant, car 3 may have gone 37.23 laps.

During most of the race, each car is on its own and is trying to beat every other car on the course. However, if there is an accident, a pace car comes out to set a safe speed. None of the race cars is permitted to pass the pace car (or the race cars in front of them), but each of the race cars wants to stay as close to the pace car as it can. While it is on the track, the pace car is a reference, and the race cars become phase-locked loops. Each driver will measure the phase difference (a distance in laps) between him and the pace car. If the driver is far away, he will increase his engine speed to close the gap. If he's too close to the pace car, he will slow down. The result is all the race cars lock on to the phase of the pace car. The cars travel around the track in a tight group that is a small fraction of a lap.

Clock analogy

Phase can be proportional to time, so a phase difference can be a time difference. Clocks are, with varying degrees of accuracy, phase-locked (time-locked) to a master clock.

Left on its own, each clock will mark time at slightly different rates. A wall clock, for example, might be fast by a few seconds per hour compared to the reference clock at NIST. Over time, that time difference would become substantial.

To keep his clock in synch, each week the owner compares the time on his wall clock to a more accurate clock (a phase comparison), and he resets his clock. Left alone, the wall clock will continue to diverge from the reference clock at the same few seconds per hour rate.

Some clocks have a timing adjustment (a fast-slow control). When the owner compared his wall clock's time to the reference time, he noticed that his clock was too fast. Consequently, he could turn the timing adjust a small amount to make the clock run a little slower. If things work out right, his clock will be more accurate. Over a series of weekly adjustments, the wall clock's notion of a second would agree with the reference time (within the wall clock's stability).

An early mechanical version of a phase-locked loop was used in 1921 in the Short-Synchronome clock.

History

Automatic synchronization of electronic oscillators was described in 1923. Earliest research towards what became known as the phase-locked loop goes back to 1932, when British researchers developed an alternative to Edwin Armstrong's superheterodyne receiver, the Homodyne or direct-conversion receiver. In the homodyne or synchrodyne system, a local oscillator was tuned to the desired input frequency and multiplied with the input signal. The resulting output signal included the original modulation information. The intent was to develop an alternative receiver circuit that required fewer tuned circuits than the superheterodyne receiver. Since the local oscillator would rapidly drift in frequency, an automatic correction signal was applied to the oscillator, maintaining it in the same phase and frequency as the desired signal. The technique was described in 1932, in a paper by Henri de Bellescize, in the French journal *L'Onde Électrique*.

In analog television receivers since at least the late 1930s, phase-locked-loop horizontal and vertical sweep circuits are locked to synchronization pulses in the broadcast signal.

When Signetics introduced a line of monolithic integrated circuits that were complete phase-locked loop systems on a chip in 1969, applications for the technique multiplied. A few years later RCA introduced the "CD4046" CMOS Micropower Phase-Locked Loop, which became a popular integrated circuit.

Structure and function

Phase-locked loop mechanisms may be implemented as either analog or digital circuits. Both implementations use the same basic structure. Both analog and digital PLL circuits include four basic elements:

- Phase detector,
- low-pass filter
- Variable frequency oscillator, and
- feedback path (which may include a frequency divider).

Variations

There are several variations of PLLs. Some terms that are used are analog phase-locked loop (APLL) also referred to as a linear phase-locked loop (LPLL), digital phase-locked loop (DPLL), all digital phase-locked loop (ADPLL), and software phase-locked loop (SPLL).

Analog or Linear PLL (LPLL)
: Phase detector is an analog multiplier. Loop filter is active or passive. Uses a Voltage-controlled oscillator (VCO).

Digital PLL (DPLL)
: An analog PLL with a digital phase detector (such as XOR, edge-trigger JK, phase frequency detector). May have digital divider in the loop.

All digital PLL (ADPLL)
: Phase detector, filter and oscillator are digital. Uses a numerically-controlled oscillator (NCO).

Software PLL (SPLL)
: Functional blocks are implemented by software rather than specialized hardware.

Performance parameters

- Type and order
- Lock range: The frequency range the PLL is able to stay locked. Mainly defined by the VCO range.

- Capture range: The frequency range the PLL is able to lock-in, starting from unlocked condition. This range is usually smaller than the lock range and will depend e.g. on phase detector.
- Loop bandwidth: Defining the speed of the control loop.
- Transient response: Like overshoot and settling time to a certain accuracy (like 50ppm).
- Steady-state errors: Like remaining phase or timing error
- Output spectrum purity: Like sidebands generated from a certain VCO tuning voltage ripple.
- Phase-noise: Defined by noise energy in a certain frequency band (like 10kHz offset from carrier). Highly dependent on VCO phase-noise, PLL bandwidth, etc.
- General parameters: Such as power consumption, supply voltage range, output amplitude, etc.

Applications

Phase-locked loops are widely used for synchronization purposes; in space communications for coherent demodulation and threshold extension, bit synchronization, and symbol synchronization. Phase-locked loops can also be used to demodulate frequency-modulated signals. In radio transmitters, a PLL is used to synthesize new frequencies which are a multiple of a reference frequency, with the same stability as the reference frequency.

Other applications include:

- Demodulation of both FM and AM signals
- Recovery of small signals that otherwise would be lost in noise (lock-in amplifier)
- Recovery of clock timing information from a data stream such as from a disk drive
- Clock multipliers in microprocessors that allow internal processor elements to run faster than external connections, while maintaining precise timing relationships
- DTMF decoders, modems, and other tone decoders, for remote control and telecommunications

Clock recovery

Some data streams, especially high-speed serial data streams (such as the raw stream of data from the magnetic head of a disk drive), are sent without an accompanying clock. The receiver generates a clock from an approximate frequency reference, and then phase-aligns to the transitions in the data stream with a PLL. This process is referred to as clock recovery. In order for this scheme to work, the data stream must have a transition frequently enough to correct any drift in the PLL's oscillator. Typically, some sort of redundant encoding is used; 8B10B is very common.

Deskewing

If a clock is sent in parallel with data, that clock can be used to sample the data. Because the clock must be received and amplified before it can drive the flip-flops which sample the data, there will be a finite, and process-, temperature-, and voltage-dependent delay between the detected clock edge and the received data window. This delay limits the frequency at which data can be sent. One way of eliminating this delay is to include a deskew PLL on the receive side, so that the clock at each data flip-flop is phase-matched to the received clock. In that type of application, a special form of a PLL called a delay-locked loop (DLL) is frequently used.

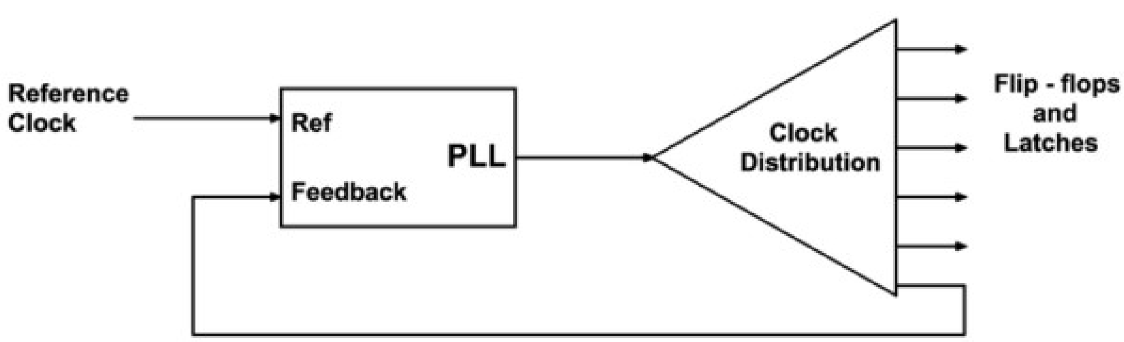
Clock generation

Many electronic systems include processors of various sorts that operate at hundreds of megahertz. Typically, the clocks supplied to these processors come from clock generator PLLs, which multiply a lower-frequency reference clock (usually 50 or 100 MHz) up to the operating frequency of the processor. The multiplication factor can be quite large in cases where the operating frequency is multiple gigahertz and the reference crystal is just tens or hundreds of megahertz.

Spread spectrum

All electronic systems emit some unwanted radio frequency energy. Various regulatory agencies (such as the FCC in the United States) put limits on the emitted energy and any interference caused by it. The emitted noise generally appears at sharp spectral peaks (usually at the operating frequency of the device, and a few harmonics). A system designer can use a spread-spectrum PLL to reduce interference with high-Q receivers by spreading the energy over a larger portion of the spectrum. For example, by changing the operating frequency up and down by a small amount (about 1%), a device running at hundreds of megahertz can spread its interference evenly over a few megahertz of spectrum, which drastically reduces the amount of noise seen on broadcast FM radio channels, which have a bandwidth of several tens of kilohertz.

Clock distribution



Typically, the reference clock enters the chip and drives a phase locked loop (**PLL**), which then drives the system's clock distribution. The clock distribution is usually balanced so that the clock arrives at every endpoint simultaneously. One of those endpoints is the PLL's feedback input. The function of the PLL is to compare the distributed clock to the incoming reference clock, and vary the phase and frequency of its output until the reference and feedback clocks are phase and frequency matched.

PLLs are ubiquitous—they tune clocks in systems several feet across, as well as clocks in small portions of individual chips. Sometimes the reference clock may not actually be a pure clock at all, but rather a data stream with enough transitions that the PLL is able to recover a regular clock from that stream. Sometimes the reference clock is the same frequency as the clock driven through the clock distribution, other times the distributed clock may be some rational multiple of the reference.

Jitter and noise reduction

One desirable property of all PLLs is that the reference and feedback clock edges be brought into very close alignment. The average difference in time between the phases of the two signals when the PLL has achieved lock is called the **static phase offset** (also called the **steady-state phase error**). The variance between these phases is called **tracking jitter**. Ideally, the static phase offset should be zero, and the tracking jitter should be as low as possible.

Phase noise is another type of jitter observed in PLLs, and is caused by the oscillator itself and by elements used in the oscillator's frequency control circuit. Some technologies are known to perform better than others in this regard. The best digital PLLs are constructed with emitter-coupled logic (ECL) elements, at the expense of high power consumption. To keep phase noise low in PLL circuits, it is best to avoid saturating logic families such as transistor-transistor logic (TTL) or CMOS.

Another desirable property of all PLLs is that the phase and frequency of the generated clock be unaffected by rapid changes in the voltages of the power and ground supply lines, as well as the voltage of the substrate on which the PLL circuits are fabricated. This is called substrate and supply noise rejection. The higher the noise rejection, the better.

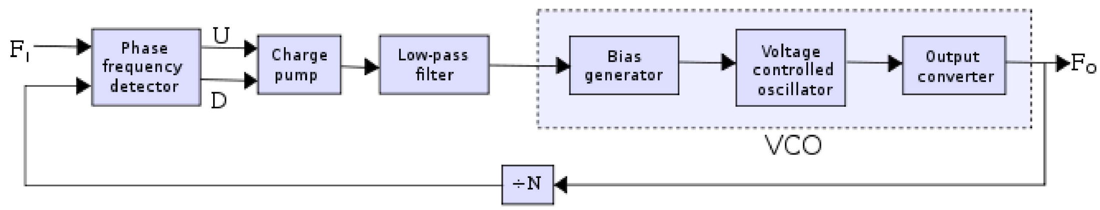
To further improve the phase noise of the output, an injection locked oscillator can be employed following the VCO in the PLL.

Frequency Synthesis

In digital wireless communication systems (GSM, CDMA etc.), PLLs are used to provide the local oscillator for up-conversion during transmission and down-conversion during reception. In most cellular handsets this function has been largely integrated into a single integrated circuit to reduce the cost and size of the handset. However, due to the high performance required of base station terminals, the transmission and reception circuits are built with discrete components to achieve the levels of performance required. GSM local oscillator modules are typically built with a frequency synthesizer integrated circuit and discrete resonator VCOs.

Frequency synthesizer manufacturers include Analog Devices, National Semiconductor and Texas Instruments. VCO manufacturers include Sirenza, Z-Communications, Inc. (Z-COMM).

Phase-locked loop block diagram



Digital phase-locked loop block diagram

A phase detector compares two input signals and produces an error signal which is proportional to their phase difference. The error signal is then low-pass filtered and used to drive a VCO which creates an output phase. The output is fed through an optional divider back to the input of the system, producing a negative feedback loop. If the output phase drifts, the error signal will increase, driving the VCO phase in the opposite direction so as to reduce the error. Thus the output phase is locked to the phase at the other input. This input is called the reference.

Analog phase locked loops are generally built with an analog phase detector, low pass filter and VCO placed in a negative feedback configuration. A digital phase locked loop uses a digital phase detector; it may also have a divider in the feedback path or in the reference path, or both, in order to make the PLL's output signal frequency a rational multiple of the reference frequency. A non-integer multiple of the reference frequency can also be created by replacing the simple divide-by-N counter in the feedback path with a programmable pulse swallowing counter. This technique is usually referred to as a fractional-N synthesizer or fractional-N PLL.

The oscillator generates a periodic output signal. Assume that initially the oscillator is at nearly the same frequency as the reference signal. If the phase from the oscillator falls behind that of the reference, the phase detector changes the control voltage of the oscillator so that it speeds up. Likewise, if the phase creeps ahead of the reference, the phase detector changes the control voltage to slow down the oscillator. Since initially the oscillator may be far from the reference frequency, practical phase detectors may also respond to frequency differences, so as to increase the lock-in range of allowable inputs.

Depending on the application, either the output of the controlled oscillator, or the control signal to the oscillator, provides the useful output of the PLL system.

Elements

Phase detector

The two inputs of the phase detector are the reference input and the feedback from the VCO. The PD output controls the VCO such that the phase difference between the two inputs is held constant, making it a negative feedback system. There are several types of phase detectors in the two main categories of analog and digital.

Different types of phase detectors have different performance characteristics.

For instance, the frequency mixer produces harmonics that adds complexity in applications where spectral purity of the VCO signal is important. The resulting unwanted (spurious) sidebands, also called "reference spurs" can dominate the filter requirements and reduce the capture range and lock time well below the requirements. In these applications the more complex digital phase detectors are used which do not have as severe a reference spur component on their output. Also, when in lock, the steady-state phase difference at the inputs using this type of phase detector is near 90 degrees. The actual difference is determined by the DC loop gain.

A **bang-bang** charge pump phase detector must always have a **dead band** where the phases of inputs are close enough that the detector detects no phase error. For this reason, bang-bang phase detectors are associated with significant minimum peak-to-peak jitter, because of drift within the dead band. However these types, having outputs consisting of very narrow pulses at lock, are very useful for applications requiring very low VCO spurious outputs. The narrow pulses contain very little energy and are easy to filter out of the VCO control voltage. This results in low VCO control line ripple and therefore low FM sidebands on the VCO.

In PLL applications it is frequently required to know when the loop is out of lock. The more complex digital phase-frequency detectors usually have an output that allows a reliable indication of an out of lock condition.

Filter

The block commonly called the PLL loop filter (usually a low pass filter) generally has two distinct functions.

The primary function is to determine loop dynamics, also called stability. This is how the loop responds to disturbances, such as changes in the reference frequency, changes of the feedback divider, or at startup. Common considerations are the range over which the loop can achieve lock (pull-in range, lock range or capture range), how fast the loop achieves lock (lock time, lock-up time or settling time) and damping behavior. Depending on the application, this may require one or more of the following: a simple proportion (gain or attenuation), an integral (low pass filter) and/or derivative (high pass filter). Loop parameters commonly examined for this are the loop's gain margin and phase margin.

Common concepts in control theory including the PID controller are used to design this function.

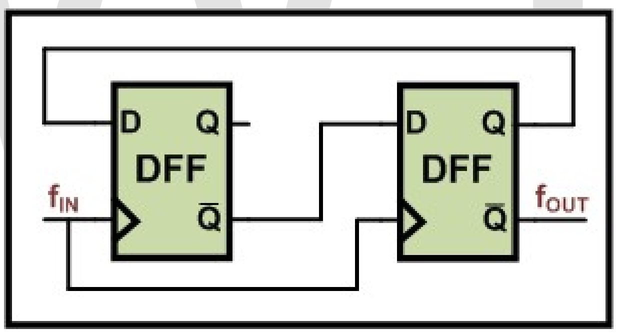
The second common consideration is limiting the amount of reference frequency energy (ripple) appearing at the phase detector output that is then applied to the VCO control input. This frequency modulates the VCO and produces FM sidebands commonly called "reference spurious". The low pass characteristic of this block can be used to attenuate this energy, but at times a band reject "notch" may also be useful.

The design of this block can be dominated by either of these considerations, or can be a complex process juggling the interactions of the two. Typical trade-offs are: increasing the bandwidth usually degrades the stability or too much damping for better stability will reduce the speed and increase settling time. Often also the phase-noise is affected.

Oscillator

All phase-locked loops employ an oscillator element with variable frequency capability. This can be an analog VCO either driven by analog circuitry in the case of an APLL or driven digitally through the use of a digital-to-analog converter as is the case for some DPLL designs. Pure digital oscillators such as a numerically-controlled oscillator are used in ADPLLs.

Feedback path and optional divider



An Example Digital Divider (by 4) for use in the Feedback Path of a Multiplying PLL

PLLs may include a divider between the oscillator and the feedback input to the phase detector to produce a frequency synthesizer. A programmable divider is particularly useful in radio transmitter applications, since a large number of transmit frequencies can be produced from a single stable, accurate, but expensive, quartz crystal–controlled reference oscillator.

Some PLLs also include a divider between the reference clock and the reference input to the phase detector. If the divider in the feedback path divides by *N* and the reference input divider divides by *M*, it allows the PLL to multiply the reference frequency by *N* / *M*. It might seem simpler to just feed the PLL a lower frequency, but in some cases the reference frequency may be constrained by other issues, and then the reference divider is useful.

Frequency multiplication in a sense can also be attained by locking the PLL to the 'N'th harmonic of the signal.

It should also be noted that the feedback is not limited to a frequency divider. This element can be other elements such as a frequency multiplier, or a mixer. The multiplier will make the VCO output a sub-multiple (rather than a multiple) of the reference frequency. A mixer can translate the VCO frequency by a fixed offset. It may also be a combination of these. An example being a divider following a mixer; this allows the divider to operate at a much lower frequency than the VCO without a loss in loop gain.

Modeling

Time domain model

The equations governing a phase-locked loop with an analog multiplier as the phase detector may be derived as follows. Let the input to the phase detector be $x_c(t)$ and the output of the VCO is $x_r(t)$ with frequency $\omega_r(t)$, then the output of the phase detector $x_m(t)$ is given by

$$x_m(t) = x_c(t) \cdot x_r(t)$$

the VCO frequency may be written as a function of the VCO input $y(t)$ as

$$\omega_r(t) = \omega_f + g_v y(t)$$

where g_v is the *sensitivity* of the VCO and is expressed in Hz / V.

Hence the VCO output takes the form

$$x_r(t) = A_r \cos\left(\int_0^t \omega_r(\tau)\,d\tau\right) = A_r \cos(\omega_f t + \varphi(t))$$

where

$$\varphi(t) = \int_0^t g_v y(\tau)\,d\tau$$

The loop filter receives this signal as input and produces an output

$$x_f(t) = F_{\text{filter}}(x_m(t))$$

where F_{Filter} is the operator representing the loop filter transformation.

When the loop is closed, the output from the loop filter becomes the input to the VCO thus

$$y(t) = x_f(t) = F_{\text{filter}}(x_m(t))$$

We can deduce how the PLL reacts to a sinusoidal input signal:

$$x_c(t) = A_c \sin(\omega_c t).$$

The output of the phase detector then is:

$$x_m(t) = A_c \sin(\omega_c t) A_r \cos(\omega_f t + \varphi(t)).$$

This can be rewritten into sum and difference components using trigonometric identities:

$$x_m(t) = \frac{A_c A_f}{2} \sin(\omega_c t - \omega_f t - \varphi(t)) + \frac{A_c A_f}{2} \sin(\omega_c t + \omega_f t + \varphi(t))$$

As an approximation to the behaviour of the loop filter we may consider only the difference frequency being passed with no phase change, which enables us to derive a small-signal model of the phase-locked loop. If we can make $\omega_f \approx \omega_c$, then the $\sin(\cdot)$ can be approximated by its argument resulting in: $y(t) = x_f(t) \simeq -A_c A_f \varphi(t)/2$. The phase-locked loop is said to be *locked* if this is the case.

Linearized phase domain model

Phase locked loops can also be analyzed as control systems by applying the Laplace transform. The loop response can be written as:

$$\frac{\theta_o}{\theta_i} = \frac{K_p K_v F(s)}{s + K_p K_v F(s)}$$

Where

- θ_o is the output phase in radians
- θ_i is the input phase in radians
- K_p is the phase detector gain in volts per radian
- K_v is the VCO gain in radians per volt-second
- $F(s)$ is the loop filter transfer function (dimensionless)

The loop characteristics can be controlled by inserting different types of loop filters. The simplest filter is a one-pole RC circuit. The loop transfer function in this case is:

$$F(s) = \frac{1}{1 + sRC}$$

The loop response becomes:

$$\frac{\theta_o}{\theta_i} = \frac{\frac{K_p K_v}{RC}}{s^2 + \frac{s}{RC} + \frac{K_p K_v}{RC}}$$

This is the form of a classic harmonic oscillator. The denominator can be related to that of a second order system:

$$s^2 + 2s\zeta\omega_n + \omega_n^2$$

Where

- ζ is the damping factor
- ω_n is the natural frequency of the loop

For the one-pole RC filter,

$$\omega_n = \sqrt{\frac{K_p K_v}{RC}}$$

$$\zeta = \frac{1}{2\sqrt{K_p K_v RC}}$$

The loop natural frequency is a measure of the response time of the loop, and the damping factor is a measure of the overshoot and ringing. Ideally, the natural frequency should be high and the damping factor should be near 0.707 (critical damping). With a single pole filter, it is not possible to control the loop frequency and damping factor independently. For the case of critical damping,

$$RC = \frac{1}{2K_p K_v}$$

$$\omega_c = K_p K_v \sqrt{2}$$

A slightly more effective filter, the lag-lead filter includes one pole and one zero. This can be realized with two resistors and one capacitor. The transfer function for this filter is

$$F(s) = \frac{1 + sCR_2}{1 + sC(R_1 + R_2)}$$

This filter has two time constants

$\tau_1 = C(R_1 + R_2)$
$\tau_2 = CR_2$

Substituting above yields the following natural frequency and damping factor

$$\omega_n = \sqrt{\frac{K_p K_v}{\tau_1}}$$
$$\zeta = \frac{1}{2\omega_n \tau_1} + \frac{\omega_n \tau_2}{2}$$

The loop filter components can be calculated independently for a given natural frequency and damping factor

$$\tau_1 = \frac{K_p K_v}{\omega_n^2}$$
$$\tau_2 = \frac{2\zeta}{\omega_n} - \frac{1}{K_p K_v}$$

Real world loop filter design can be much more complex e.g. using higher order filters to reduce various types or source of phase noise.

Chapter- 8

Twisted Pair

25-pair color code Chart

Twisted pair cabling is a type of wiring in which two conductors (the forward and return conductors of a single circuit) are twisted together for the purposes of canceling out electromagnetic interference (EMI) from external sources; for instance, electromagnetic radiation from unshielded twisted pair (UTP) cables, and crosstalk between neighboring pairs. It was invented by Alexander Graham Bell.

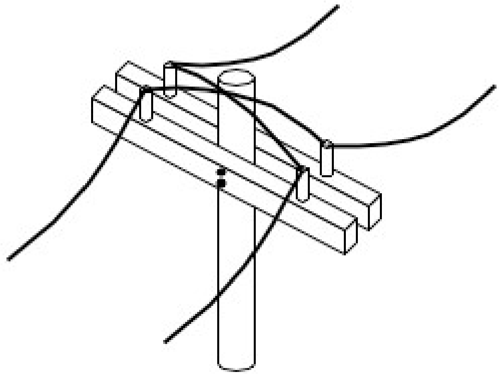
Explanation

In balanced pair operation, the two wires carry equal and opposite signals and the destination detects the difference between the two. This is known as differential mode transmission. Noise sources introduce signals into the wires by coupling of electric or magnetic fields and tend to couple to both wires equally. The noise thus produces a common-mode signal which is cancelled at the receiver when the difference signal is taken. This method starts to fail when the noise source is close to the signal wires; the closer wire will couple with the noise more strongly and the common-mode rejection of the receiver will fail to eliminate it. This problem is especially apparent in telecommunication cables where pairs in the same cable lie next to each other for many miles. One pair can induce crosstalk in another and it is additive along the length of the cable. Twisting the pairs counters this effect as on each half twist the wire nearest to the noise-source is exchanged. Providing the interfering source remains uniform, or nearly so, over the distance of a single twist, the induced noise will remain common-mode. Differential signaling also reduces electromagnetic radiation from the cable, along with the associated attenuation allowing for greater distance between exchanges.

The twist rate (also called *pitch* of the twist, usually defined in twists per meter) makes up part of the specification for a given type of cable. Where nearby pairs have equal twist rates, the same conductors of the different pairs may repeatedly lie next to each other, partially undoing the benefits of differential mode. For this reason it is commonly specified that, at least for cables containing small numbers of pairs, the twist rates must differ.

In contrast to **FTP** (foiled twisted pair) and **STP** (shielded twisted pair) cabling, **UTP** (unshielded twisted pair) cable is not surrounded by any shielding. It is the primary wire type for telephone usage and is very common for computer networking, especially as patch cables or temporary network connections due to the high flexibility of the cables.

History



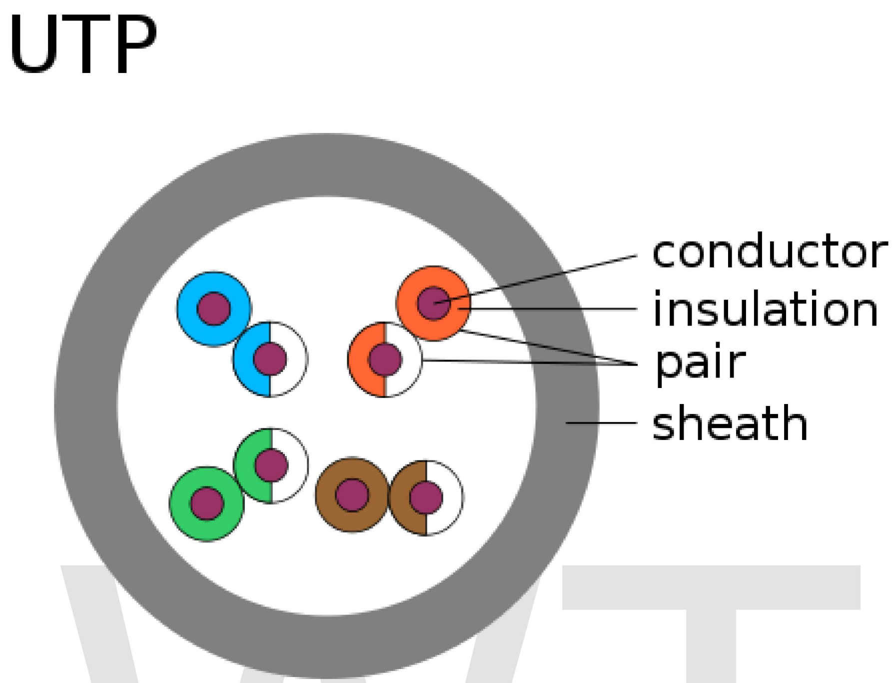
Wire transposition on top of pole

The earliest telephones used telegraph lines, or open-wire single-wire earth return circuits. In the 1880s electric trams were installed in many cities, which induced noise into these circuits. Lawsuits being unavailing, the telephone companies converted to balanced circuits, which had the incidental benefit of reducing attenuation, hence increasing range.

As electrical power distribution became more commonplace, this measure proved inadequate. Two wires, strung on either side of cross bars on utility poles, shared the route with electrical power lines. Within a few years the growing use of electricity again brought an increase of interference, so engineers devised a method called wire transposition, to cancel out the interference. In wire transposition, the wires exchange position once every several poles. In this way, the two wires would receive similar EMI from power lines. This represented an early implementation of twisting, with a twist rate of about four twists per kilometre, or six per mile. Such open-wire balanced lines with periodic transpositions still survives today in some rural areas.

Twisted pair cables were invented by Alexander Graham Bell in 1881. By 1900, the entire American telephone line network was either twisted pair or open wire with transposition to guard against interference. Today, most of the millions of kilometres of twisted pairs in the world are outdoor landlines, owned by telephone companies, used for voice service, and only handled or even seen by telephone workers.

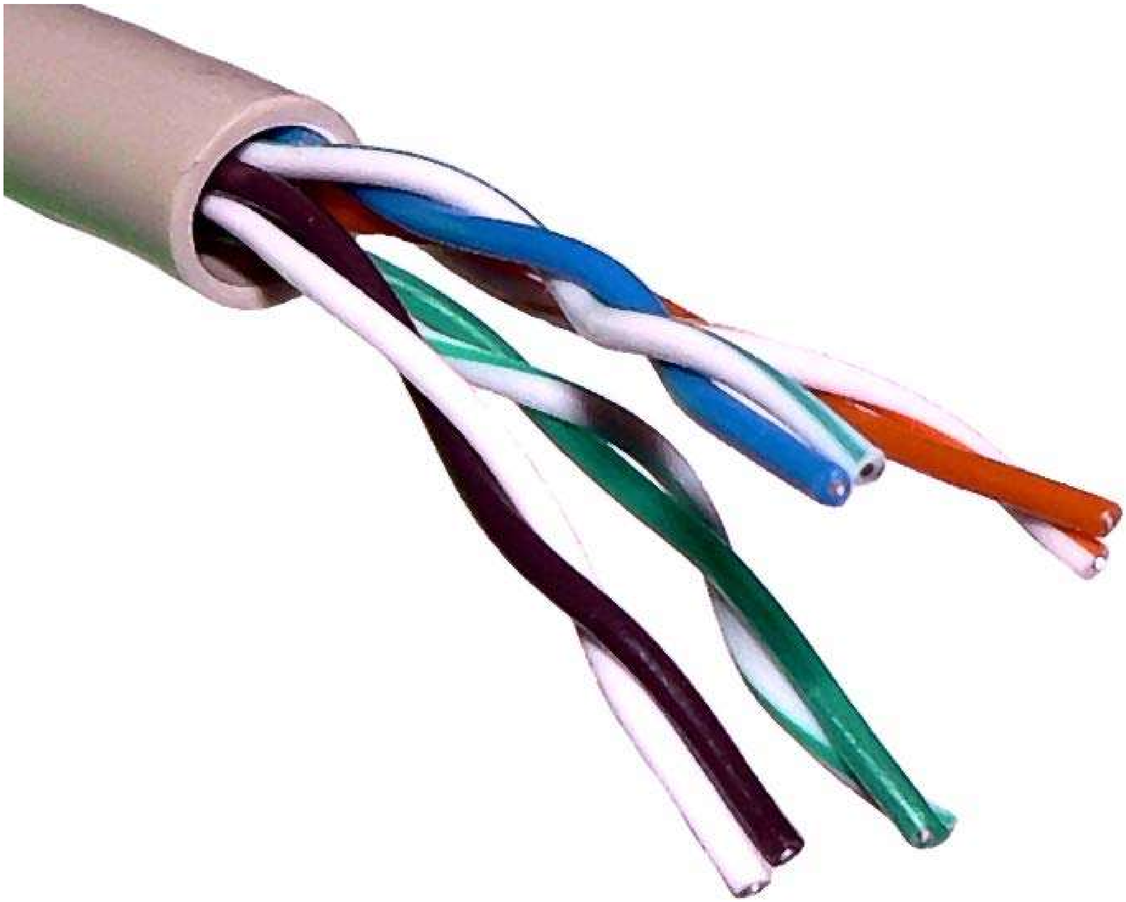
Unshielded twisted pair (UTP)



Unshielded twisted pair

UTP cables are found in many Ethernet networks and telephone systems. For indoor telephone applications, UTP is often grouped into sets of 25 pairs according to a standard 25-pair color code originally developed by AT&T. A typical subset of these colors (white/blue, blue/white, white/orange, orange/white) shows up in most UTP cables.

For urban outdoor telephone cables containing hundreds or thousands of pairs, the cable is divided into smaller but identical bundles. Each bundle consists of twisted pairs that have different twist rates. The bundles are in turn twisted together to make up the cable. Pairs having the same twist rate within the cable can still experience some degree of crosstalk. Wire pairs are selected carefully to minimize crosstalk within a large cable.

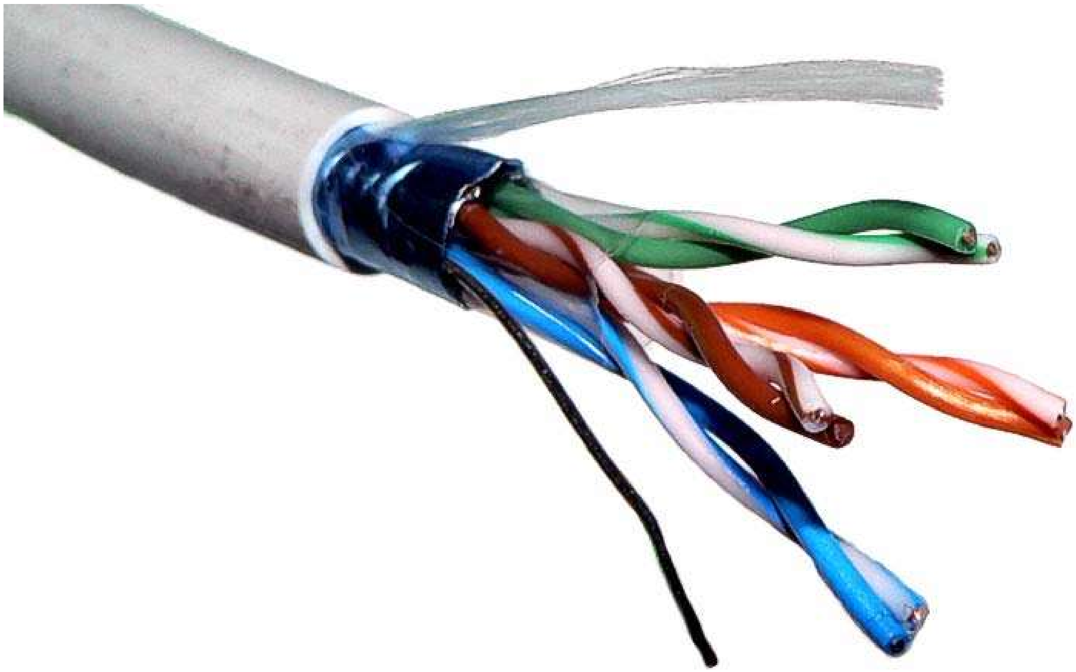


Unshielded twisted pair cable with different twist rates

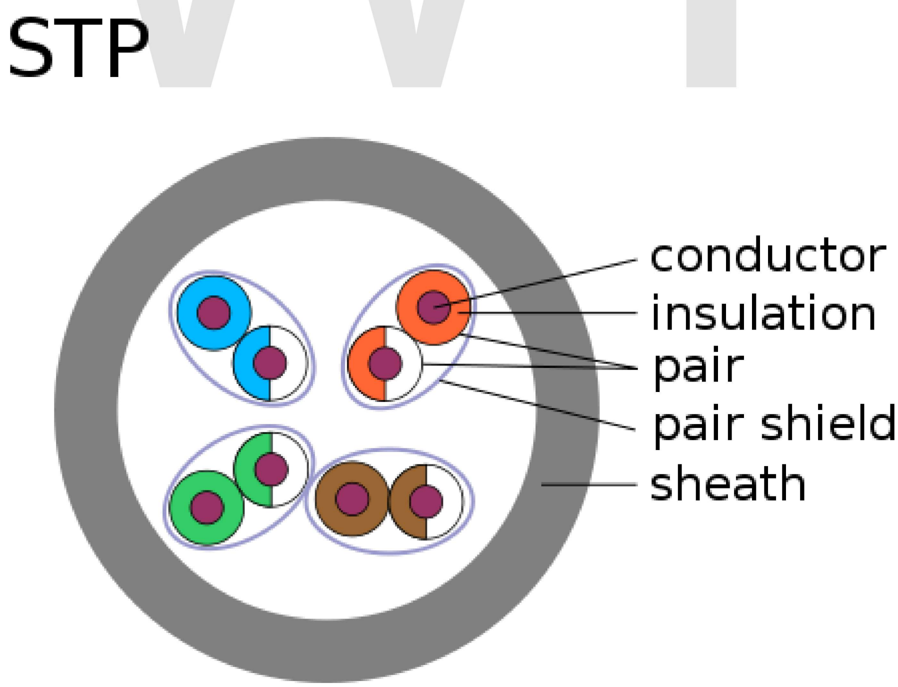
UTP cable is also the most common cable used in computer networking. Modern Ethernet, the most common data networking standard, utilizes UTP cables. Twisted pair cabling is often used in data networks for short and medium length connections because of its relatively lower costs compared to optical fiber and coaxial cable.

UTP is also finding increasing use in video applications, primarily in security cameras. Many middle to high-end cameras include a UTP output with setscrew terminals. This is made possible by the fact that UTP cable bandwidth has improved to match the baseband of television signals. While the video recorder most likely still has unbalanced BNC connectors for standard coaxial cable, a balun is used to convert from 100-ohm balanced UTP to 75-ohm unbalanced. A balun can also be used at the camera end for ones without a UTP output. Only one pair is necessary for each video signal.

Cable shielding

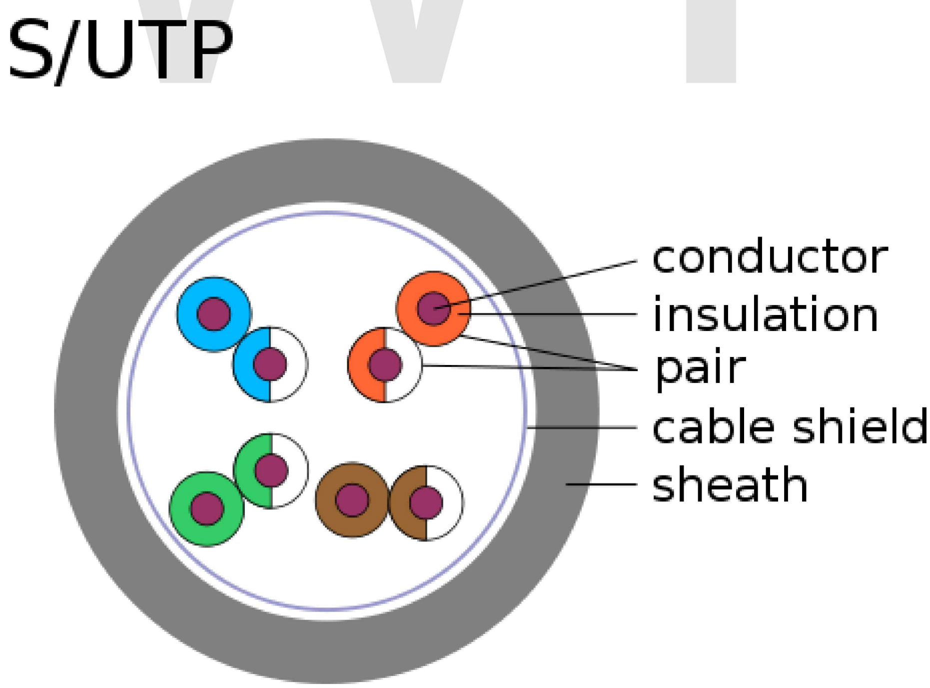


S/UTP, also known as FTP

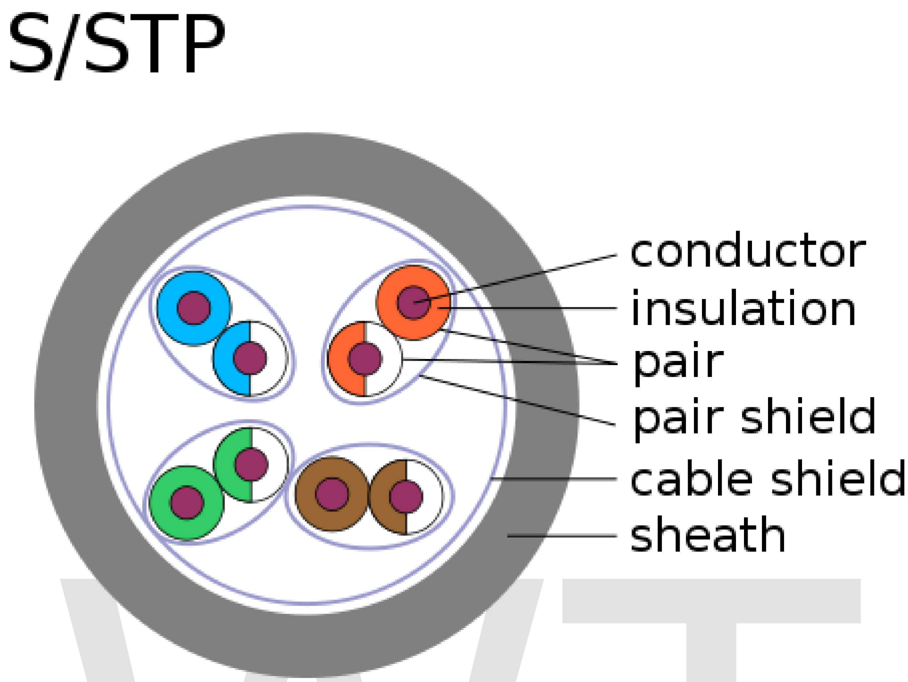


STP cable format

S/STP, also known as S/FTP.



S/UTP cable format



S/STP cable format

Twisted pair cables are often shielded in attempt to prevent electromagnetic interference. Because the shielding is made of metal, it may also serve as a ground. However, usually a shielded or a screened twisted pair cable has a special grounding wire added called a drain wire. This shielding can be applied to individual pairs, or to the collection of pairs. When shielding is applied to the collection of pairs, this is referred to as screening. The shielding must be grounded for the shielding to work.

Shielded twisted pair (STP or STP-A)

STP cabling includes metal shielding over each individual pair of copper wires. This type of shielding protects cable from external EMI. e.g. the 150 ohm shielded twisted pair cables defined by the IBM Cabling System specifications and used with token ring networks.

Screened unshielded twisted pair (S/UTP)

Also known as Foiled Twisted Pair (FTP), is a screened UTP cable (ScTP).

Screened shielded twisted pair (S/STP or S/FTP)

S/STP cabling, also known as Screened Fully shielded Twisted Pair (S/FTP), is both individually shielded (like STP cabling) and also has an outer metal shielding covering the entire group of shielded copper pairs (like S/UTP). This type of cabling offers the best protection from interference from external sources, and also eliminates *alien crosstalk*.

Note that different vendors and authors use different terminology (i.e. STP has been used to denote both STP-A, S/STP, and S/UTP).

Comparison of some old and new abbreviations, according to ISO/IEC 11801:

Old name	New name	cable screening	pair shielding
UTP	U/UTP	none	none
FTP	F/UTP	foil	none
STP	U/FTP	none	foil
S-FTP	SF/UTP	foil, braiding	none
S-STP	S/FTP	braiding	foil

The code before the slash designates the shielding for the cable itself, while the code after the slash determines the shielding for the individual pairs:

TP = twisted pair
U = unshielded
F = foil shielding
S = braided shielding

Most common cable categories

Category	Bandwidth	Applications	Notes
Cat1	0.4 MHz	Telephone and modem lines	Not described in EIA/TIA recommmendations. Unsuitable for modern systems.
Cat2	? MHz	Older terminal systems, e.g. IBM 3270	Not described in EIA/TIA recommmendations. Unsuitable for modern systems.
Cat3	16MHz	10BASE-T and 100BASE-T4 Ethernet	Described in EIA/TIA-568. Unsuitable for speeds above 16 Mbit/s.
Cat4	20MHz	16 Mbit/s Token Ring	
Cat5	100MHz	100BASE-TX & 1000BASE-T Ethernet	
Cat5e	100MHz	100BASE-TX & 1000BASE-T Ethernet	Enhanced Cat5. Practically the same as Cat5, but with better testing standards so Gigabit Ethernet works reliably.
Cat6	250MHz	1000BASE-T Ethernet	Most commonly installed cable in Finland according to the 2002 standard. SFS-EN 50173-1
Cat6e	250MHz (500MHz according to some)	10GBASE-T (under development) Ethernet	Not a standard; a cable maker's own label.

Cat6a	500MHz	10GBASE-T (under development) Ethernet	Standard under development (ISO/IEC 11801:2002 Amendment 2).
Cat7	600MHz	No applications yet.	Four pairs, U/FTP (shielded pairs). Standard under development.
Cat7a	1000MHz	Telephone, CATV, 1000BASE-T in the same cable.	Four pairs, S/FTP (shielded pairs, braid-screened cable). Standard under development.
Cat8	1200MHz	Under development, no applications yet.	Four pairs, S/FTP (shielded pairs, braid-screened cable). Standard under development.

Solid core cable vs stranded cable

Solid core cable is supposed to be used for permanently installed runs. It is less flexible than stranded cable and is more prone to failure if repeatedly flexed. Stranded cable is used for fly leads at patch panel and for connections from wall-ports to end devices, as it resists cracking of the conductors. Stranded core is generally more expensive than solid core.

Connectors need to be designed differently for solid core than for stranded. Use of a connector with the wrong cable type is likely to lead to unreliable cabling. Plugs designed for solid and stranded core are readily available, and some vendors even offer plugs designed for use with both types. The punch-down blocks on patch-panel and wall port jacks are designed for use with solid core cable.

Manufacture

Copper Rod Breakdown

The first step in low voltage cable production is copper rod breakdown. Copper is sent to the factory in 5,000 lb coils. These copper coils are continuously drawn through diamond dies that drastically reduce the diameter of the copper to 10 or 12 gauge. Lubrication is used during this process to reduce the amount of friction and heat on the copper cable. Once completed, the copper is stacked in vertical coils, called *stem packs*. These stem packs are then transferred to another drawing operation that further reduces the gauge of the copper. During this stage, the copper is also charged with an electrical current. This anneals the copper, which is a softening process. Once annealed and cooled off, the copper runs through a laser measurement system, to verify it is within manufacturing specifications.

Copper Insulation Process

The copper insulation process is continually monitored and controlled up to +/- .0001". Once the copper is insulated, it runs through a water cooling trough, allowing the wire jacket to harden properly.

Copper Twisting

Twisting helps reduce crosstalk between the individual pairs of wire. Some Cat6 premises cables include a center spline, or wire separator, to further reduce crosstalk and increase performance. Copper twisting is accomplished by running each individual wire through multiple faceplates. This helps control pair position. Once twisted, this is known as a *cable unit*.

Jacketing

The cable unit then goes through the jacketing process. This step varies, depending on the type of cable being manufactured. OSP cable typically uses a black polyethylene or UV rated polyvinyl chloride (PVC). For Cat3, Cat5e and Cat6 premises cable, varying grades of PVC are used, depending on flame safety rating requirements. This step starts off with molten plastic being extruded at high pressure and formed around the moving cable core. Shielding, ripcords, armoring and water blocking compound may also be applied at this step. Cables that require dual shielding or double armor will need to repeat this process. Once completed, the cable passed through a long cooling bath, then through a laser micrometer to verify the final diameter.

Printing

Printing is done just before the cable is put in its final packaging. For OSP cable, a hot foil printing process is used, that leaves an indented print in the cable jacket. For premises cable, a high speed ink jet printer is used. Some cable manufacturers print distance marking from 1000–0 ft, or from 305–0 m making it very easy to determine how much cable is left in the box, or for measuring out cable runs. Other manufacturers use a six digit distance mark, making the process a little harder.

Coiling

The completed cable is then wound onto a reel or coil. The coiling process requires very precise tension controls to ensure the cable won't tangle when being pulled out of its box.

Final Testing

Once the cable is printed and coiled, it goes through one last set of tests. The manufacturer will test it against a large set of mechanical and electrical performance specifications. Once tested, the cable is ready for shipment.

Advantages

- It is a thin, flexible cable that is easy to string between walls.
- More lines can be run through the same wiring ducts.
- UTP costs less per meter/foot than any other type of LAN cable.

Disadvantages

- Twisted pair’s susceptibility to electromagnetic interference greatly depends on the pair twisting schemes (usually patented by the manufacturers) staying intact during the installation. As a result, twisted pair cables usually have stringent requirements for maximum pulling tension as well as minimum bend radius. This relative fragility of twisted pair cables makes the installation practices an important part of ensuring the cable’s performance.

- In video applications that send information across multiple parallel signal wires, twisted pair cabling can introduce signaling delays known as skew which results in subtle color defects and ghosting due to the image components not aligning correctly when recombined in the display device. The skew occurs because twisted pairs within the same cable often use a different number of twists per meter so as to prevent common-mode crosstalk between pairs with identical numbers of twists. The skew can be compensated by varying the length of pairs in the termination box, so as to introduce delay lines that take up the slack between shorter and longer pairs, though the precise lengths required are difficult to calculate and vary depending on the overall cable length.

Minor twisted pair variants

- **Loaded twisted pair**: A twisted pair that has intentionally added inductance, common practice on telecommunication lines, except those carrying higher than voiceband frequencies. The added inductors are known as load coils and reduce distortion.
- **Unloaded twisted pair**: A twisted pair that has no added load coils.
- **Bonded twisted pair**: A twisted pair variant in which the pairs are individually bonded to increase robustness of the cable. Pioneered by Belden, it means the electrical specifications of the cable are maintained despite rough handling.
- **Twisted ribbon cable**: A variant of standard ribbon cable in which adjacent pairs of conductors are bonded and twisted together. The twisted pairs are then lightly bonded to each other in a ribbon format. Periodically along the ribbon there are short sections with no twisting to enable connectors and pcb headers to be terminated using the usual ribbon cable IDC techniques.

Chapter- 9

Transposition and Trunking

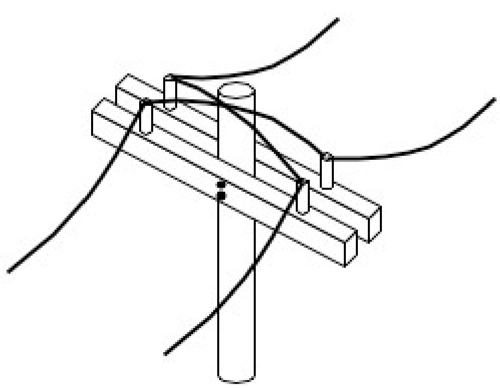
Transposition

Transposition is the periodic swapping of positions of the conductors of a transmission line, in order to reduce crosstalk and otherwise improve transmission. In telecommunications this applies to balanced pairs whilst in power transmission lines three conductors are periodically transposed.

For cables, the swapping is gradual and continuous; that is the two or three conductors are twisted around each other. For communication cables this is called twisted pair. For overhead power lines or open pair communication lines, the conductors are exchanged at pylons, for example at transposing pylons or at utility poles, respectively.

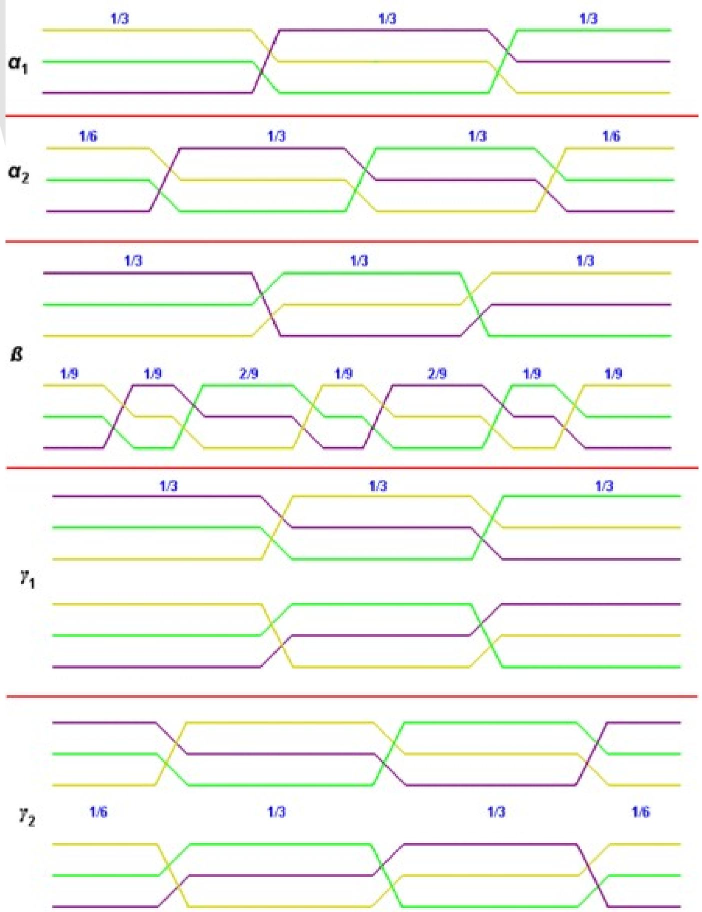
The mutual influence of electrical conductors is reduced by transposition. Transposition also equalizes their impedance relative to ground, thus avoiding one-sided loads in three-phase systems. Transposing is an effective measure for the reduction of inductively linked normal mode interferences.

Power lines



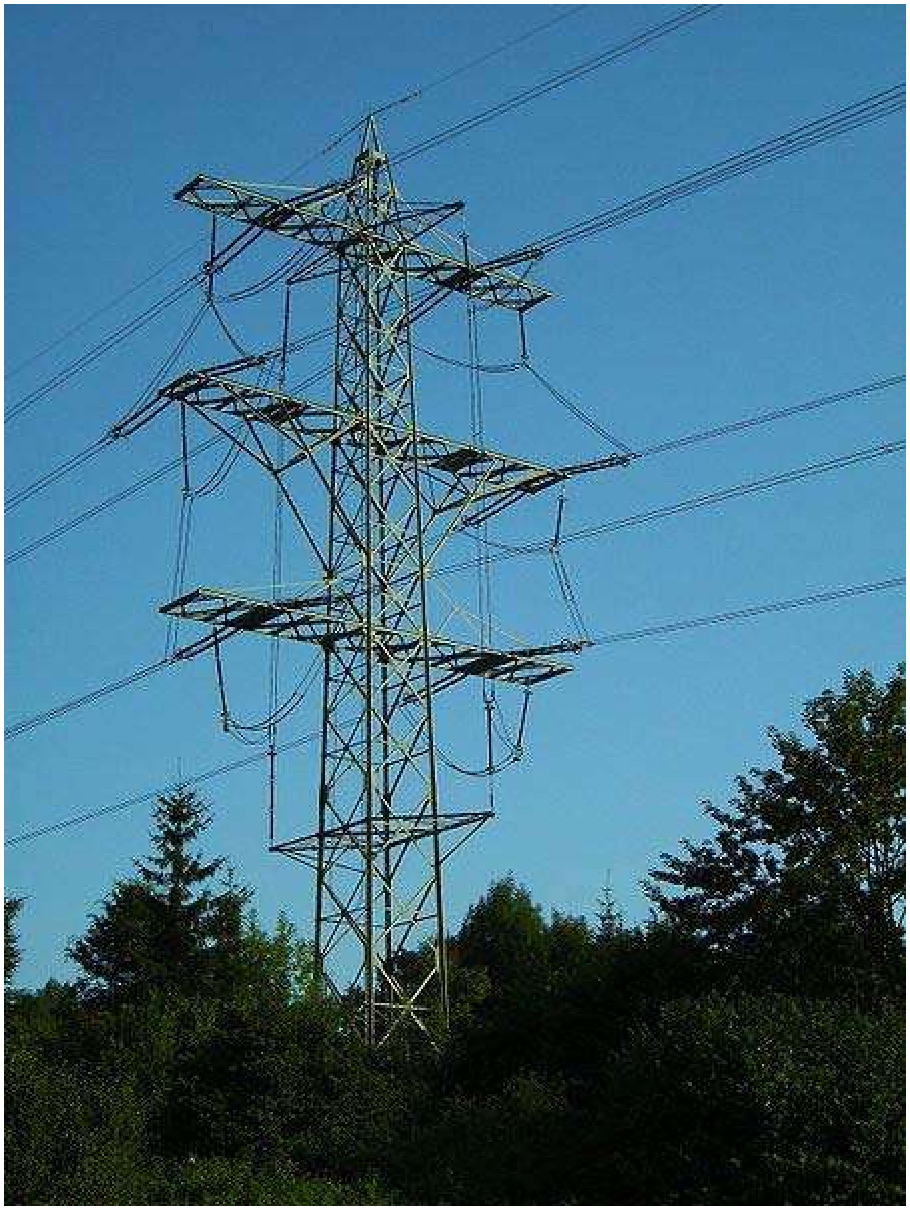
Wire transposition on top of pole

At longer powerlines without branches transposing takes place usually after the so-called transposing scheme, at closely branched grids and at parallel guidance of several electric circuits - in particular when these are line of different voltages levels - on the same pylons the outside unbalance of the line, which is caused by the other electric circuits, dominates. In these cases one finds large deviations from the transposing schemes, in whom for example transpositions are realized, at which only two of the three conductors on the pylons change their place. Also transpositions on pylons directly before the introduction to switchgears are found, in order to get an optimal arrangement of the feeding system without crossing of conductors. As the mutual influence of electric circuits can change after new lines were installed or old lines dismantled, it is possible that certain transpositions disappear or are added after construction measures in electricity mains took place. In the case of a twisted line the individual conductors of an electric circuit exchange either in their whole course (at cables) or at certain points (at overhead lines) their place to each other. The mutual influence of electrical conductors is reduced by transposing. In addition by transposing also the unbalance of the line, which can lead to one-sided loads in three-phase systems, is reduced. Transposing is at overhead lines usually realized at so-called transposing pylons. Transposing is an effective measure for the reduction of inductively linked normal mode interferences.



Three basic patterns, with variants

A **transposing scheme** is a pattern by which the conductors of overhead power lines are transposed at transposing structures. In order to ensure balanced capacitance of a three-phase line, each of the three conductors must hang once at each position of the overhead line.



Pylon 206 of powerline Hoheneck-Herbertingen near Rübgarten.

At a transposing structure, the conductors change their relative places in the line. A transposing structure may be a standard structure with special cross oarms, or may be a dead-end structure. The transposing is necessary as there is capacitance between conductors, as well as between conductors and ground. This is typically not symmetrical across phases. By transposing, the overall capacitance for the whole line is approximately balanced. Transposings also reduce effects to communiciations circuits.

Telecommunication

In communications cables, transposition is used to reduce coupling between circuits in the same cable. The principle measure is the pitch, or the distance over which the pairs of a circuit are twisted. By twisting, the wires become longer than the cable. The stranding factor indicates the relationship of single wire length to cable length; it amounts to with communication cables about 1.02 to 1,04.

Kinds of stranding

In practice the following kinds of stranding occur more frequently:

- Pair stranding: Two single wires are stranded to a conductor (twin wire).
- Three-stranding: Three single wires are stranded to a tripartite group.
- Four-stranding: This group is essentially divided into the two special kinds of stranding:
  -
    - Star quad twisting: Four single wires have the same situation in each place of the rope to each other, whereby the veins of a conductor (twin wire) face each other each other diagonally.

Transmission technique

The kinds of stranding have different transmission characteristics. The capacity of a stranding affects itself, for example the two conductors of a quadruple run over the entire cable length parallel during star quad twisting. Capacitance between the conductors is thus substantially higher than with Dieselhorst Martin (DHM) - stranding in which the situation of the conductors to each other in the cable changes repeatedly. Because of the smaller work capacity of the DHM stranding it is possible to form additional electric circuits with the help of a phantom circuit. Since the phantom transducers are turned on to in the middle of the master transducers, the currents of the phantom circuit on the two coming Rome circles compensate themselves.

Trunking

In modern communications, **trunking** is a concept by which a communications system can provide network access to many clients by sharing a set of lines or frequencies instead of providing them individually. This is analogous to the structure of a tree with one trunk and many branches. Examples of this include telephone systems and the VHF radios commonly used by police agencies. More recently port trunking has been applied in computer networking as well.

A **trunk** is a single transmission channel between two points, each point being either the switching center or the node.

Etymology

How the term came to apply to communications is unclear, but its previous use in railway track terminology (e.g., India's Grand Trunk Road, Canada's Grand Trunk Railway) was based on the natural model of a tree trunk and its branches. It is likely that the same analogy drove the communications usage.

An alternative explanation is that, from an early stage in the development of telephony, the need was found for thick cables (up to around 10 cm diameter) containing many pairs of wires. These were usually covered in lead. Thus, both in colour and size they resembled an elephant's trunk. This leaves open the question of what term was applied to connections among exchanges during the years when only open wire was used.

Radio communications

In two-way radio communications, trunking refers to the ability of transmissions to be served by free channels whose availability is determined by algorithmic protocols. In conventional (i.e., not trunked) radio, users of a single service share one or more exclusive radio channels and must wait their turn to use them, analogous to the operation of a group of cashiers in a grocery store, where each cashier serves his/her own line of customers. The cashier represents each radio channel, and each customer represents a radio user transmitting on their radio.

Trunked radio systems (TRS) pool all of the cashiers (channels) into one group and use a store manager (site controller) that assigns incoming shoppers to free cashiers as determined by the store's policies (TRS protocols).

In a TRS, individual transmissions in any conversation may take place on several different channels, much as if a family of shoppers checked out all at once, they may be assigned different cashiers by the traffic manager. Similarly, if a single shopper checks out more than once, they may be assigned a different cashier each time.

Trunked radio systems provide greater efficiency at the cost of greater management overhead. The store manager's orders must be conveyed to all the shoppers. This is done by assigning one or more radio channels as the "control channel". The control channel transmits data from the site controller that runs the TRS, and is continuously monitored by all of the field radios in the system so that they know how to follow the various conversations between members of their talkgroups (families) and other talkgroups as they hop from radio channel to radio channel.

TRS's have grown massively in their complexity since their introduction, and now include multi-site systems that can cover entire states or groups of states. This is similar to the idea of a chain of grocery stores. The shopper generally goes to the nearest grocery store, but if there are complications or congestion, the shopper may opt to go to a neighboring store. Each store in the chain can talk to each other and pass messages between shoppers at different stores if necessary, and they provide backup to each other: if a store has to be closed for repair, then other stores pick up the customers.

TRS's have greater risks to overcome than conventional radio systems in that a loss of the store manager (site controller) would cause the system's traffic to no longer be managed. In this case, most of the time the TRS automatically reverts to conventional operation. In spite of these risks, TRS's usually maintain reasonable uptime.

TRS's are more difficult to monitor via radio scanner than conventional systems; however, larger manufacturers of radio scanners have introduced models that, with a little extra programming, are able to follow TRS's quite efficiently.

Telecommunications

Trunk line

A **trunk line** is a circuit connecting telephone switchboards (or other switching equipment), as distinguished from local loop circuit which extends from telephone exchange switching equipment to individual telephones or information origination/termination equipment.

When dealing with a private branch exchange (PBX), trunk lines are the phone lines coming into the PBX from the telephone provider . This differentiates these incoming lines from extension lines that connect the PBX to (usually) individual phone sets. Trunking saves cost, because there are usually fewer trunk lines than extension lines, since it is unusual in most offices to have all extension lines in use for external calls at once. Trunk lines transmit voice and data in formats such as analog, T1, E1, ISDN or PRI. The dial tone lines for outgoing calls are called DDCO (Direct Dial Central Office) trunks.

Trunk call

In the UK and the Commonwealth countries, a *trunk call* was a long distance one as opposed to a *local call*.

Telephone exchange

Trunking also refers to the connection of switches and circuits within a telephone exchange. Trunking is closely related to the concept of grading. Trunking allows a group of inlet switches at the same time. Thus the service provider can provide a lesser number of circuits than might otherwise be required, allowing many users to "share" a smaller number of connections and achieve capacity savings.

Computer networks

Link aggregation

In computer networking, **trunking** is a slang term referring to the use of multiple network cables or ports in parallel to increase the link speed beyond the limits of any one single cable or port. This is called link aggregation. These aggregated links may be used to interconnect switches.

VLANs

In the context of VLANs, Avaya and Cisco use the term "trunking" to mean "VLAN multiplexing" - carrying multiple VLANs through a single network link through the use of a "trunking protocol". To allow for multiple VLANs on one link, frames from individual VLANs must be identified. The most common and preferred method, IEEE 802.1Q adds a tag to the Ethernet frame header, labeling it as belonging to a certain VLAN. Since 802.1Q is an open standard, it is the only option in an environment with multiple-vendor equipment.

Cisco also has a proprietary trunking protocol called Inter-Switch Link which encapsulates the Ethernet frame with its own container, which labels the frame as belonging to a specific VLAN.

Chapter- 10

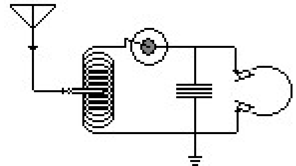
Radio Receiver Design and Variable-Frequency Oscillator

Radio receiver design

Radio receiver design includes the electronic design of different components of a radio receiver which processes the radio frequency signal from an antenna in order to produce usable information such as audio. Here we only concentrates on the historical configurations leading up to and including the modern superheterodyne receiver design. The complexity of a modern receiver and the possible range of circuitry and methods employed are more generally covered in electronics and communications engineering. The term *radio receiver* is understood here to mean any device which is intended to receive a radio signal in order to generate useful information from the signal, most notably a recreation of the so-called baseband signal (such as audio) which modulated the radio signal at the time of transmission in a communications or broadcast system.

Crystal radio

A crystal radio uses no active parts: it is powered solely by the radio signal itself, whose detected power feeds headphones in order to be audible at all. In order to achieve even a minimal sensitivity, a crystal radio is limited to low frequencies using a large antenna (usually a long wire). It relies on detection using some sort of semiconductor diode such as the original cat's-whisker diode discovered long before the development of modern semiconductors.



A *crystal set receiver* consisting of an antenna, a variable inductor, a cat's whisker, and a filter capacitor.

- Advantages
    - Simple, easy to make. Here we see a classic design for a clandestine receiver in a POW camp.
- Disadvantages
    - Insensitive, it needs a very strong RF signal and/or a long-wire antenna to operate.
    - Poor selectivity since it only has one tuned circuit.

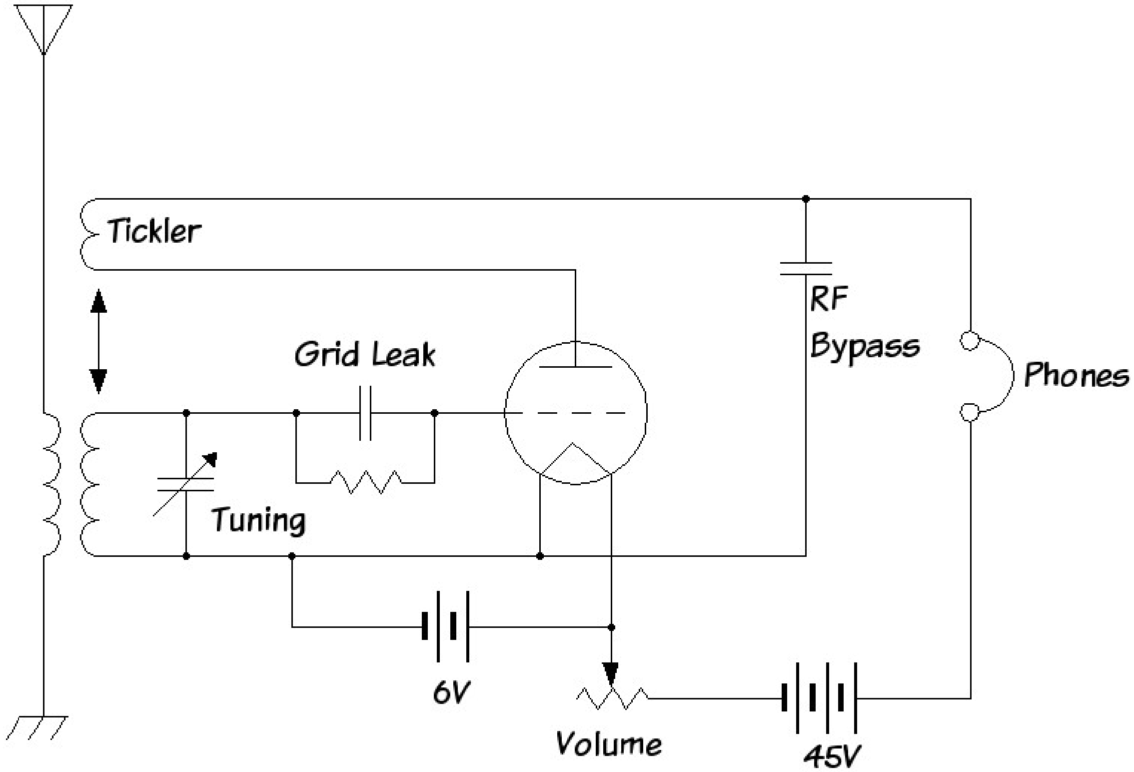
Tuned radio frequency

The tuned radio frequency receiver (TRF) consists of a radio frequency amplifier having one or more stages all tuned to the desired reception frequency. This is followed by a detector, typically an envelope detector using a diode, followed by audio amplification. This was developed after the invention of the triode vacuum tube, greatly improving the reception of radio signals using electronic amplification which had not previously been available. The greatly improved selectivity of the superheterodyne receiver overtook the TRF design in almost all applications, however the TRF design was still used as late as the 1960's among the cheaper "transistor radios" of that era.

Reflectional

The reflectional receiver was a design from the early 20th century which consists of a single stage TRF receiver but which used the same amplifying tube to also amplify the audio signal after it had been detected. This was in an era where each tube was seen as a major cost (and recipient of electrical power) so that a substantial increase in the number of passive elements would be seen as preferable to including an additional tube. The design tends to be rather unstable, and is obsolete.

Regenerative



Classical regenerative receiver using a single triode vacuum tube. The orientation of the "tickler" coil was carefully adjusted by the operator in order to vary the amount of positive feedback.

The regenerative receiver also had its heyday at the time where adding an active element (vacuum tube) was considered costly. In order to increase the gain of the receiver, positive feedback was used in its single RF amplifier stage; this also increased the selectivity of the receiver well beyond what would be expected from a single tuned circuit. The amount of feedback was quite critical in determining the resulting gain and had to be carefully adjusted by the radio operator. Increasing the feedback beyond a point caused the stage to oscillate at the frequency it was tuned to.

Self-oscillation reduced the qualitity of its reception of an AM (voice) radio signal but did ironically make it useful as a CW (morse code) receiver inasmuch as the beat signal between the oscillation and the radio signal would produce an audio "beeping" sound. The oscillation of the regenerative receiver could also be an annoying source of local interference. An improved design known as the super-regenerative receiver improved the performance by allowing an oscillation to build up which was then "quenched," with that cycle repeating at a rapid (ultrasonic) rate. From the accompanying schematic for a practical regenerative receiver, one can appreciate its simplicity in relation to a multi-stage TRF receiver, while able to achieve the same level of amplification through the use of positive feedback.

Direct conversion

In the Direct conversion receiver, the signals from the antenna are only tuned by a single tuned circuit before entering a mixer where they are mixed with a signal from a local oscillator which is tuned to the carrier wave frequency of the transmitted signal (unlike the superheterodyne design, where the local oscillator is at an offset frequency). The output of this mixer is thus audio frequency, which is passed through a low pass filter into an audio amplifier which may drive a speaker.

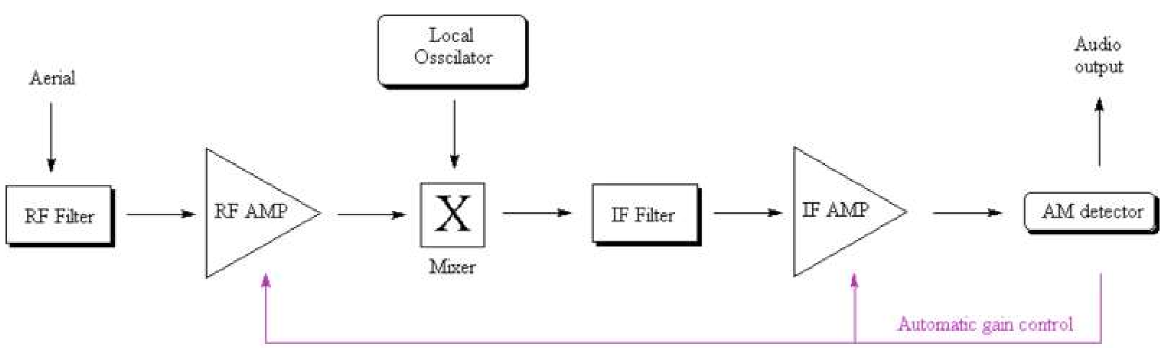
For receiving CW (morse code) the local oscillator is tuned to a frequency slightly different from that of the transmitter in order to turn the received signal into an audible "beep."

- Advantages
    - Simpler than a superheterodyne receiver

- Disadvantages
    - Poor rejection of strong signals at adjacent frequencies compared to a superheterodyne receiver.
    - Increased noise or interference when receiving a SSB signal since there is no selectivity against the undesired sideband.

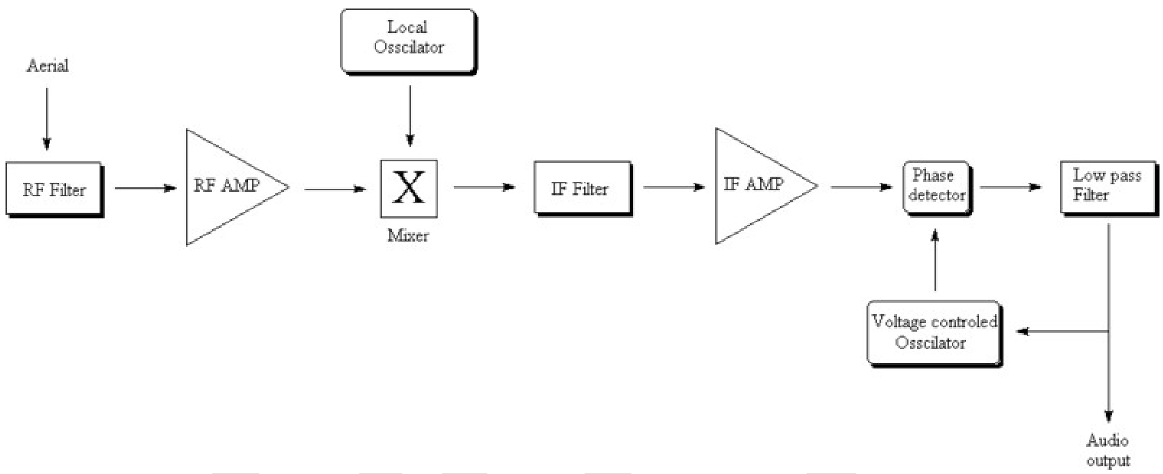
Superheterodyne

Practically all modern receivers are of the superheterodyne design. The RF signal from the antenna may have one stage of amplification to improve the receiver's noise figure, although at lower frequencies this is typically omitted. The RF signal enters a mixer, along with the output of the local oscillator, in order to produce a so-called intermediate frequency (IF) signal. The local oscillator is tuned to a frequency somewhat higher (or lower) than the intended reception frequency so that the IF signal will be at a particular frequency where it is further amplified in a narrow-band multistage amplifier. Tuning the receiver involves changing the frequency of the local oscillator, with further processing of the signal (especially in relation to increasing the selectivity of the receiver) conveniently done at a single frequency (the IF frequency) thus requiring no further tuning for different stations.



A schematic of a superhet AM receiver. Note that the radio includes an AGC loop in order to maintain the RF and IF stages in their linear region, and to produce an audio output not dependent on the signal power received.

Here we show block diagrams for typical superheterodyne receivers for AM and FM broadcast respectively. This particular FM design uses a modern phase locked loop detector, unlike the frequency *discriminator* or ratio detector used in earlier FM receivers.



A schematic of a simple superhet broadcast FM receiver. Note that there is no AGC loop, but simply uses a high-gain IF amplifier which is intentionally driven into saturation (or *limiting*).

For single conversion superheterodyne AM receivers designed for medium wave (AM broadcast) the IF is commonly 455 kHz. Most superheterodyne receivers designed for broadcast FM (88 - 108 MHz) use an IF of 10.7 MHz. TV receivers often use intermediate frequencies of about 40 MHz. Some modern multiband receivers actually convert lower frequency bands first to a much higher frequency (VHF) after which a second mixer with a tunable local oscillator and a second IF stage process the signal as above.

Variable-frequency oscillator

A **variable frequency oscillator (VFO)** in electronics is a oscillator whose frequency can be tuned (i.e. varied) over some range. It is a necessary component in any tunable radio receiver or transmitter that works by the superheterodyne principle, and controls the frequency to which the apparatus is tuned.

Purpose

In a simple superhet radio receiver, the incoming radio frequency signal (at frequency f_{IN}) from the antenna is *mixed* with the VFO output signal tuned to f_{LO}, producing an intermediate frequency (*IF*) signal that can be processed downstream to extract the modulated information. The IF signal frequency is chosen to be either the sum of the two frequencies at the mixer inputs (up-conversion), $f_{IN} + f_{LO}$ or more commonly, the difference frequency (down-conversion), $f_{IN} - f_{LO}$, depending on the receiver design.

In addition to the desired *IF* signal and its unwanted image (the mixing product of opposite sign above), the mixer output will also contain the two original frequencies, f_{IN} and f_{LO} and various harmonic combinations of the input signals. These undesired signals are rejected by the IF filter. If a double balanced mixer is employed, the input signals appearing at the mixer outputs are greatly attenuated, reducing the required complexity of the IF filter.

The advantage of using a VFO as a hetrodyning oscillator is that only a small portion of the radio receiver (all sections before the mixer such as the preamplifier) need to have a wide bandwidth. The rest of the receiver can be finely tuned to the IF frequency.

In a direct-conversion receiver, the VFO is tuned to the same frequency as the incoming radio frequency and $f_{IF} = 0$ Hz. Demodulation takes place at baseband using low-pass filters and amplifiers.

In a radio frequency (RF) transmitter, VFOs are often used to tune the frequency of the output signal, often indirectly through a hetrodyning process similar to that described above. Other uses include chirp generators for radar systems where the VFO is swept rapidly through a range of frequencies, timing signal generation for oscilloscopes and time domain reflectometers, and variable frequency audio generators used in musical instruments and audio test equipment.

There are two main types of VFO in use: analog and digital.

Analog VFO

An analog VFO is an electronic oscillator where the value of at least one of the passive components is adjustable under user control so as to alter its output frequency. The passive component whose value is adjustable is usually a capacitor, but could be a variable inductor.

Tuning Capacitor

The variable capacitor is a mechanical device in which the separation of a series of interleaved metal plates is physically altered to vary its capacitance. Adjustment of this capacitor is sometimes facilitated by a mechanical step-down gearbox to achieve fine tuning.

Varactor

A reversed-biased semiconductor diode exhibits capacitance. Since the width of its non-conducting depletion region depends on the magnitude of the reverse bias voltage, this voltage can be used to control the junction capacitance. The varactor bias voltage may be generated in a number of ways and there may need to be no significant moving parts in the final design. Varactors have a number of disadvantages including temperature drift and aging, electronic noise, low Q factor and non-linearity.

Digital Crystal VFO

Modern radio receivers and transmitters usually use some form of digital frequency synthesis to generate their VFO signal. The advantages including smaller designs, lack of moving parts, and the ease with which preset frequencies can be stored and manipulated in the digital computer that is usually embedded in the design for other purposes.

It is also possible for the radio to become extremely frequency-agile in that the control computer could alter the radio's tuned frequency many tens, thousands or even millions of times a second. This capability allows communications receivers effectively to monitor many channels at once, perhaps using digital selective calling (DSC) techniques to decide when to open an audio output channel and alert users to incoming communications. Pre-programmed frequency agility also forms the basis of some military radio encryption and stealth techniques. Extreme frequency agility lies at the heart of spread spectrum techniques that are currently gaining mainstream acceptance in computer wireless networking such as Wi-Fi.

There are disadvantages to digital synthesis such as the inability of a digital synthesiser to tune smoothly through all frequencies, but with the channelisation of many radio bands, this can also be seen as an advantage in that it prevents radios from operating in between two recognised channels.

Digital frequency synthesis relies on stable crystal controlled reference frequency sources. Crystal controlled oscillators are more stable than inductively and capacitively controlled oscillators. Their disadvantage is that changing frequency (more than a small amount) requires changing the crystal, but frequency synthesizer techniques have made this unnecessary in modern designs.

Digital Frequency Synthesis

The electronic and digital techniques involved in this include:

Direct Digital Synthesis (DDS)

> Enough data points for a mathematical sine function are stored in digital memory. These are recalled at the right speed and fed to a digital to analog converter where the required sine wave is built up.

Direct Frequency Synthesis

Early channelized communication radios had multiple crystals - one for each channel on which they could operate. After a while this thinking was combined with the basic ideas of heterodyning and mixing described under purpose above. Multiple crystals can be mixed in various combinations to produce various output frequencies.

Phase Locked Loop (PLL)

Using a varactor-controlled or voltage-controlled oscillator (VCO) (described above in varactor under analog VFO techniques) and a phase detector, a control-loop can be set up so that the VCO's output is frequency-locked to a crystal controlled reference oscillator. The phase detector's comparison is made between the outputs of the two oscillators after frequency division by different divisors. Then by altering the frequency-division divisor(s) under computer control, a variety of actual (undivided) VCO output frequencies can be generated.

The PLL technique dominates most radio VFO designs today.

Performance

The quality metrics for a VFO include frequency stability, phase noise and spectral purity. All of these factors tend to be inversely proportional to the tuning circuit's Q factor. Since in general, the tuning range is also inversely proportional to Q, these performance factors generally degrade as the VFO's frequency range is increased.

Stability

Stability is the measure of how far a VFO's output frequency drifts with time and temperature. To mitigate this problem, VFOs are generally "phase locked" to a stable reference oscillator. PLLs use negative feedback to correct for the frequency drift of the VFO allowing for both wide tuning range and good frequency stability.

Repeatability

Ideally, for the same control input to the VFO, the oscillator should generate exactly the same frequency. A change in the calibration of the VFO can change receiver tuning calibration; periodic re-alignment of a receiver may be needed. VFO's used as part of a phase-locked loop frequency synthesizer have less stringent requirements since the system is as stable as the crystal-controlled reference frequency.

Purity

A plot of a VFO's amplitude vs. frequency may show several peaks, probably harmonically related. Each of these peaks can potentially mix with some other incoming signal and produce a *spurious* response. These *spurii* (sometimes spelled *spuriae*) can result in increased noise or two signals detected where there should only be one. Additional components can be added to a VFO to suppress high-frequency parasitic oscillations, should these be present.

In a transmitter, these spurious signals are generated along with the one desired signal. Filtering may be required to ensure the transmitted signal meets regulations for bandwidth and spurious emissions.

Phase noise

When examined with very sensitive equipment, the pure sine-wave peak in a VFO's frequency graph will most likely turn out not to be sitting on a flat noise-floor. Slight random 'jitters' in the signal's timing will mean that the peak is sitting on 'skirts' of phase noise at frequencies either side of the desired one.

These are also troublesome in crowded bands. They allow through unwanted signals that are fairly close to the expected one, but because of the random quality of these phase-noise 'skirts', the signals are usually unintelligible, appearing just as extra noise in the received signal. The effect is that what should be a clean signal in a crowded band can appear to be a very noisy signal, because of the effects of strong signals nearby.

The effect of VFO phase noise on a transmitter is that random noise is actually transmitted either side of the required signal. Again, this must be avoided for legal reasons in many cases.

Crystal control

In all performances cases, crystal controlled oscillators are better behaved than the semiconductor- and LC-based alternatives. They tend to be more stable, more repeatable, have fewer and lower harmonics and lower noise than all the alternatives in their cost-band. This in part explains their huge popularity in low-cost and computer-controlled (i.e. PLL and synthesizer-based) VFOs.