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Analog filters: Filter classification 1. Perfect answers. The ideal filter has this Av0 amplification for the signals it transmits, and zero amplification for others. Depending on voltage fluctuations depending on the frequency, the filters are divided into four categories: ideal filter reactions are characterized by - horizontal in bandwidth - and vertical in a faded band. In fact, - bandwidth is not flat and can even ripple - filters, whose slopes in the faded strip are very high, have quite complex structures. 2. Different types of real filters. 2.1. Passive filter filters of the first order A are said to be of the first order, as the denominator of its function of transmission F(p) is a polynom of maximum power 1. Passive filters of the first order contain only one element of inductive or capacitor type, and their slope in the sloping strip is 20 dB/decade (or 6 dB/octave). Note: The load of the passive filter should be infinite or very large, otherwise it brings a high cut rate. The filter can be downloaded with an adapter, such as a follower, followed by a cargo intended for it. 2.2. Active filters of the first order 2.2.1. Low aisle filters. Active second-order filters. A second-order filter if the denominator of its F(p) transmission function is 2p polynom. Low pass filter: Its transmission function is shaped: Gm is the maximum gain, w0 is a pure pulsation, close to the pulse of the break, m - damping factor. The low pass transfer function is permanent. High Pass Filter: Its transmission function is a form: Previous definitions remain unchanged. The numerator in p2. Tape Pass Filter: Its form transition function: the numerator here at p. The selectivity of the lane aisle is represented by the q-f0/B factor, called the splash factor; Also equal to 1/2 m. B bandwidth per - 3 dB filter. Strip cutting filter (or strip ejector): Its transmission function is a form: The numerator here is in constant p2. Note: Transfer function banners are written in the same form. For articles of the same name, see filter and filter (audio). The filter is a quadripolis. An electronic filter is a circuit that performs a voluntary operation to form an electrical size (current or voltage). The filter converts the history of this login size (i.e. its sequential values over a period of time) into a output size. Reason on filters they are considered quadripolises whose electrical entry and exit sizes will be a signal, even if they are not used to transmit information (as in the case of power filters). This approach takes advantage of significant mathematical efforts in signal processing. Analog electronic filters are used, perhaps in conjunction with active components (amplifiers), resistances that are components indifferent to variations over time, with other dipoles that store energy to restore it later, called characteristics: electrostatic form for capacitors whose power resists voltage changes at their terminals, absorbing or returning current; electromagnetic form for coils, the induction of which is contrasted with fluctuations in the current that passes through them by increasing or reducing voltage. Some designs use components that cause signal delay or delay. The classification of components between resistances and reactions reflects their main characteristic, but it is never pure: any electrical or electronic element has resistance and self-induction effect in all its conductors, as well as capacitive effects in its insulation parts. On the other hand, the current spreads at a limited speed, causing internal delay. As a result, all circuits behave at least slightly like filters. At high frequency, these effects are particularly pronounced and should be taken into account. Without further explanation, however, the term filter is reserved for cases where output sizes are voluntarily converted according to the previous state of input sizes. Typically, the filter is a linear filter or digested to it. The term filter applied to electronics is metonichim from a rhetorical point of view. The new concept is called analogy. At the beginning of the 20th century, telegraph and telephone companies sought to make multiple messages on the same cable. Upon arrival, the channels had to be separated. The analogy between electric shock and hydraulics was well established; Engineers described the device as an electric wave filter (certified in 1915). The electronic filter should separate the useful part of the unwanted part, such as an air filter separated from dust. In the power filter, the residual ripples from the straighteners must be discarded, and only the direct current should be retained. However, the similarities are imperfect. While liquid filters retain a solid part, the filter later restores all the electrical energy he had absorbed. To address the area where the filter is studied and calculated, it is best to put aside the memorable value of the term filter. Mathematical studies of these patterns led to overall results, and the term filter took on the general meaning of the chain combining resistances and reactions to convert the signal. The filter category includes bass and sharp controls, as well as accent chains that can amplify rather than just remove, and change the signal in a more subtle way. Even so, there are still uses very close to the original meaning; when we talk about the ideal filter, for example, in the context of conversions between analog and digital signals, we talk about the device, the way completely and without distorting the useful part and rejecting the completely and completely undesirable part of the signal. Studies of filters defined as circuits combining resistance and reactions to convert a signal have limited the duration of their designs to the operating range where they are, precisely or roughly linear. Chains that combine non-linear elements with resistance and reactions, such as diodes in radio and television detection circuits, or level detectors of audiodynamic compressors, are rarely called filters. Thus, the meaning of the term electronic filter has become clearer to denote, in particular, a linear signal processing device. Study and calculation of filters Detailed articles: Linear filter and linear filter synthesis. To study the filters, the component network is transposed to a transmission function that expresses a dynamic relationship between the size of the input and the size of the quadrifield output. The filter can be identified and studied by its impulse reaction or its frequency response; these two approaches are equivalent. Taken in terms of frequency response, the filter enhances or reduces parts of the signal in different ways. For example, in a radio receiver, you can find tone settings to increase or decrease low sounds (low frequencies) or high-frequency beeps. These commands correspond to filters. Considered in terms of its pulse response, the filter spreads a short pulse over a longer period of time by reducing their amplitude, then called an integrative filter, or increases the size of variations without affecting the level of a stable signal, then called the filter In this approach, we also study the delay of the output signal in relation to the input signal. The mathematical tools that correspond to these studies are the transformation of Laplace (converted into digital signal) for the temporal domain and for the pulse response, and for the frequency domain to be transformed by Fourier. We go from one to the other by roll-up. Mathematics filter classes have offered formulas for filters: Bessel filter or Thompson filter, offers constant bandwidth delay. Butterworth filter - get as consistent as possible in the bandwidth filter - is better selective than the Butterworth filter, but wavy either in bandwidth, or in the deflected lane, or in both (elliptical filters). The Legendre filter is designed for strictly monotonous mitigation (without ripples) and maximum stiffness in the immediate vicinity of the cutout frequency. Methods avoid or simplify calculations to get closer to filters: Bode chart to measure frequency response, Nyquist chart to test stability. The order of filter filters can be described in terms of the order of the equation of their transmission function, as well as, in analog electronics, the number of independent reactive elements that make up them. The higher the order, the steeper the slope of the transition between mitigation and reinforcement regions. The first-order filter, consisting of a single reaction resistance cell, has a maximum tilt of 6 dB/octave. Quality Factor - Detailed articles: The quality factor and depreciation factor. The quality factor describes the filter's ability to choose frequency. It is the ratio of bandwidth to the central frequency. The bandwidth is that between frequencies for which the signal power is halved, relative to the maximum. Either the central frequency filter f't'displaystyle f_, its cut-off frequencies (defined as those where the level differs by 3 dB from the central frequency level) f 0 'displaystyle f_{0}' and f 1 'displaystyle f_{1} In terms of impulse response, the quality factor expresses f_{0} f_{1} f_s hydration response in terms of impulse response. The higher q, the more hesitation the output will have after the momentum at the entrance. Residual ripples These are frequency reaction defects compared to the ideal pattern. Stability Detailed Articles: Stability of Linear Filters and Nyquist Chart. A filter that includes a power source using an electronic amplifier can become unstable, i.e. turn into an oscillator, making it unable to transmit a signal. Learning the stability of the filter is an important part of the filter design. Types of filters Result from applying a low pass filter to the image Result of applying a high pass filter to the image The result of applying a tape-pass filter to the image Click on the thumbnail to enlarge it. Filters can be categorized by an effect that is expected to be produced. Low Pass Filters (integrators) Detailed article: Low Pass Filter. A low-frequency filter intensifies frequencies below a certain frequency called cutout frequency, or outsifies others (high frequencies). This could also be called a high reduction. We distinguish between those who have a plateau reaction that have a gain for low frequencies and other gain, less for trebles, with a transition zone between the two transition frequencies, and those who have an endless response for which the response shows an increase in direct current at the cutout frequency and continuously decreases for higher frequencies. In terms of impulse response, low-aisle filters include signal changes; output is a kind of average signal history. Low-aisle filters are used to remove signal parts from useful bandwidth, which can lead to distortions (intermodulation, folding of the spectrum) later. For the beep, the low pass reduces highs and strengthens the bass. DC Power is a low-aisle filter that removes ripple residues from the straightener. High Pass Filters (derivatives or jubilees) Detailed article: High Pass Filter. The high-pass filter enhances frequencies more at a certain frequency called cutout frequency, or atst facilitates others (low frequencies). It can also be called a low neckline. We distinguish between those who have a plateau reaction, who have a gain for low frequencies and another gain, higher, for trebles, with a transition zone between the two transition frequencies, and those who have an endless response, for which the response to continuous current, the low frequency limit is zero (-∞ dB), and increase for the high frequencies, to the limits of the system. In terms of impulse reaction, high-aisle filters amplify signal variations. They can be used to detect a signal jog in the trigger chain or the front of the watch signal. When processing an image, the differentiated player emphasizes the outline. For the beep it's and a sharp amplifier. High pass filters eliminate the continuous component of the signal. Tape pass and tape cutter filter filter tape pass has a higher payout for a certain frequency band. The ejection filter, also known as a trap, bell or strip cutter filter, is an addition to the tape pass. This reduces the range of frequencies. In terms of impulse response, the tape pass filters reflect the degree of similarity of the input to the typical pulse. They can detect a signal in an environment that includes noise. Filters tape and strip-cutters are necessarily second order or higher order. Tape pass filters are fundamental to the radio. Tape pass and tape cutter make up the bulk of audio equalizers. Switch Filters A pass-all filter, also known as phase filter or corrective cell, has the same benefit across the entire frequency range used, but the relative frequency phase that makes up the signal varies depending on the frequency. Filter comb Detailed article: Filter in the crest. The comb filter is applied to get or fade for one frequency and all of its multiples. It is obtained by mixing a deferred copy of this signal multiplied by coefficient C with input. The frequency of 1/2-1 and all of its multiples will be added to the phase opposition and will fade, while the frequency of 1/9 and all its multiples will be added in stages and strengthened. In terms of impulse reaction, the filter converts momentum into a series of pulses. Analog Electronics Passive Filter Common Passive Filter Low Passage Passive Filter is characterized by exclusive use of passive components (resistance, capacitors, coils combined or not). Thus, their profit (power ratio between exit and entrance) can not exceed 1. In other words, they lighten the signal in different ways depending on the frequency. The simplest filters are based on RC, RL diagrams that determine the time constant and first-order transmission function. LC or Circuit RLC schemes allow second-order filters, tape or tape cutters and resonators (configured circuits). More complex configurations may be required. Computer design software can be used to determine their frequency and phase response or impulse reaction. Passive filters can handle importantcuts. They are rarely subject to saturation unless they have coils with nuclei. Acoustic enclosures in most low-frequency circuits, filters using coils have become rare. Active filters are used, which in any order can only use resistance and capacitors. The coils remain dominant for the crossover networks of passive speakers, where they distribute energy depending on the distribution of frequencies between sharp and bass speakers. These filters get modulated power at two input poles, while they have four or six exit poles, depending on the number of speakers they serve. All components interact, including speakers (and their acoustic charge). Calculating acoustic acoustic filters is a real specialty. Complexity can be reduced by separating the chain as soon as the amplifier. With a speaker amplifier rather than a speaker amplifier, the crossover filter processes the signal without significant power, until gain, and the tracks no longer interact electrically; only acoustic problems remain. Detailed articles: Crossover (audio) and Pregnant (audio). Frequency Reject Detailed article: Stop the reels. Passive, RC and often LC filters are used in the input of low-frequency circuits to deflect high frequencies before sending a signal to the amplifier circuit, in which high frequencies can generate disturbances in a useful frequency range, by detecting or intermodulating. These filters usually consist of a serial stop coil (two if the line is symmetrical) and low capacity in parallel. High-frequency passive filters, especially resonators, are often used. From a few hundred MHz the filter concept dissolves because all components, including wires or circuit boards, have significant inductions and capabilities at these frequencies; all parts of the chain are a kind of filter part. To complete, you should mention quartz filters, surface wave filters (Surface Acoustic Waves or SAW filters), ceramic filters and mechanical filters, which are also part of passive filters. Piezoelectric filter Piezoelectric properties of some materials, such as quartz, can be used in filter design. The quartz filters have a high quality factor and very good temperature stability. SAW A SAW (Surface Acoustic Wave) is an electromechanical system commonly used in radio wave applications. Electrical signals are converted into a mechanical wave by crystal This wave is delayed when it spreads through the crystal and then is converted into an electrical signal. Delays in exits are recombined to form the ultimate pulse filter reaction. Ceramic Filter Detailed article: Ceramic filter. Active Filter General Detailed article: Active filter. Active low-aisle filters use at least one active component (electronic tube, transistor, operating amplifier, or other analog integrated circuit). In fact, it is a scheme of amplifiers, the frequency of which is regulated in the oil by pheasant elements both in the direct chain and in the response. As a result, they can have a total profit of more than 1. They can amplify certain frequencies and soften them. These schemes make it possible to do without coils, expensive components, difficult to miniaturize and imperfect (loss angles, own resonances, sensitivity to parasites). Active filters are good for low amplitude and low power signals. Therefore, they are widely used in audio amplifiers and electronic instruments of all kinds. On the other hand, unlike passive filters, they require power. When the peak voltage of the signal reaches the power voltage, or when the voltage change exceeds the chain's capabilities, the active filter quickly produces significant distortions. The active filter amplifier element inevitably leads to some noise and harmonic distortions. It should be noted, however, that passive filters reduce signal, and that they usually need to be accompanied by an amplifier to compensate for this loss. This amplifier has the same drawbacks. The design of active filters requires precautionary measures to ensure their stability. Filter with the capabilities of the set Detailed article: Filter with recruitment capabilities. You can use typed circuits to use analog filters. They allow better integration and easy adjustment of cutout frequencies. Digital Signal Processing Filters Detailed Article: Digital Filter. Digital signal processing allows you to create a wide range of digital filters. Filtering is done by an algorithm run either by a microprocessor or by a specialized scheme (Digital Signal Processor, DSP). The principle of treatment is bundle. Each output sample represents amounts such as the product of input samples of different times stored in the buffer by coefficients stored in a different buffer. Digital filters exist as soon as analog-digital conversion phase (CAN) as long as it uses overwork technique and vice versa to digital analog transformation, for the same reasons. Similarly, anti-replication filters are important for all sampling frequency conversions. Odds are calculated according to the same mathematical principles as for all filters. However, they lead to a series that theoretically should be endless, and will go very far if someone wants to go as far as the smallest number represented by the system. So we have to compromise. Ultimate Impulse Reaction Filters (RIF) Detailed article: Ultimate Impulse Reaction Filter. End impulse reaction filters use signal bundles using an array of coefficients. They should hold up the signal enough to have all the samples that contribute to the current sample output. Digital filters with a finite pulse can change the relative level of different frequencies without affecting the relative phase. Endless Pulse Response Filters (RII) Detailed article: Infinite Pulse Response Filter. Endless impulse response filters use recursion using output samples, perhaps delayed, for calculation. Most often, they are numerical interpretations of analog filter formulas. See also the Bibliography international electrical commission, IEC 60050 International Electric Vocabular, 2015 (1st ad 1982) (read online), including 151-13-55 filter, on www.electropedia.org 131-15-38 Perfect Filter, on www.electropedia.org Annals of Telecommunications, Springer Paris Edition (French) at link.springer.com, 1946 - Insufficient Source Related Articles Related Chart Of Bandwidth Bode Cut Frequency CTCSS Mechanical Filter Notes - With Two Rare Exceptions: Superconductors, and Resistance Mubius consisting of a hard film deposited on the surface of the dielectric tape connected to the chain by two terminals located on either side of the same tape point, which have no reaction and are therefore purely resistant (Richard Taillet and Loic villain, Dictionary of Physics, Brussels, De Boeck, 2013, p. 602). George Ashley Campbell, Electric Wave Filter, U.S. Patent 1227113, filed july 15, 1915 (Passive analog filter development); in France: 1935, Louis Cohen, Heaviside electrical circuit theory. Use of electric filters, underwater cables, power lines and artificial lines. Translated from Frederick Sarrath. Paris: Eyrolles - drums present particular problems and scheme designers are reluctant to use them for a long time. The coils have little pure resistance once they have a slightly significant induction as is necessary in low frequency; this resistance complicates the calculations, but it makes sense to reduce the effect of the capacity between the spires. They have losses from the incision in the nucleus, proportional frequency, and loss on Foucault's current in the nucleus, proportional to the square frequency. Metal coils create a magnetic distortion of the saturation of the nucleus. Air coils capture interference. It's hard to get components of exact value if you're using a dive core that requires manual adjustment. See the coils (electricity). Francis Brouchier, Speakers and Acoustic Speakers - Theory and Practice, filters for acoustic speakers portal physics portal electricity and electronics technology portal This document comes from (electronic) -oldid-175456114. (electronic) -oldid-175456114. android auto apk google play services not working