GTE LENKURT

DEMODULATOR

MARCH/APRIL 1975

light-route radio systems

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Light-route radio systems are a specialized form of communications which enable industrial and common carrier users to cut costs and still retain high quality of transmission.

The 2-GHz band is ideal for light-route communications because in this frequency range, atmospheric fading is minimal, and long transmission paths are practical. Before the advent of light-route systems, a 300-channel radio might have had to be used to carry the small number of channels often required by an industrial user, thus increasing the per-channel cost of the microwave radio system and wasting rf spectrum.

Some of the users of narrow-band, operational fixed, light-route systems include petroleum companies, railroads, and independent telephone companies. An "operational fixed" station is an FCC (Federal Communications Commission) designation used to describe a radio station at a permanent location which is not open to public correspondence, but operated by and for the sole use of those agencies operating their own radio communication facilities in the Public Safety, Industrial, Land Transportation, and Aviation services. The GTE Lenkurt 70F1 radio, for example, has been designed for use in this category of service. It is a 36-channel system which operates in the 2.11 – 2.20 GHz domestic industrial and common carrier frequency allocations. In geographical areas under the territorial jurisdiction of the FCC, this radio has been type-accepted for the types of services shown in Figure 1.

Light-Route Uses

A petroleum company can use a light-route system to control the flow of oil in a pipeline between drilling sites and refineries. The majority of the channels would probably be used for administrative functions, but the main function of the system would be in those channels used for remote and supervisory control, telemetering (the transmission of measurement data over a distance), and on-line control and switching of pipelines or pumping facilities. A typical light-route system might appear as shown in Figure 2.

Complex pipeline control networks are used in the petroleum industry to control both production and petroleum flow. Drilling and pumping operations can be remotely controlled through the use of microwave systems. The results of remote chemical analysis of production samples is used to control drilling. When the percentage

FCC RULE PART	TYPE OF SERVICE
21	DOMESTIC PUBLIC FOR COMMON CARRIER USE
87	OPERATIONAL FIXED FOR AVIATION SERVICES
89	OPERATIONAL FIXED FOR PUBLIC SAFETY RADIO SERVICES
91	OPERATIONAL FIXED FOR INDUSTRIAL RADIO SERVICES
93	OPERATIONAL FIXED FOR LAND TRANSPORTATION RADIO SERVICES

Figure 1. Example of FCC radio type acceptance for specified services.

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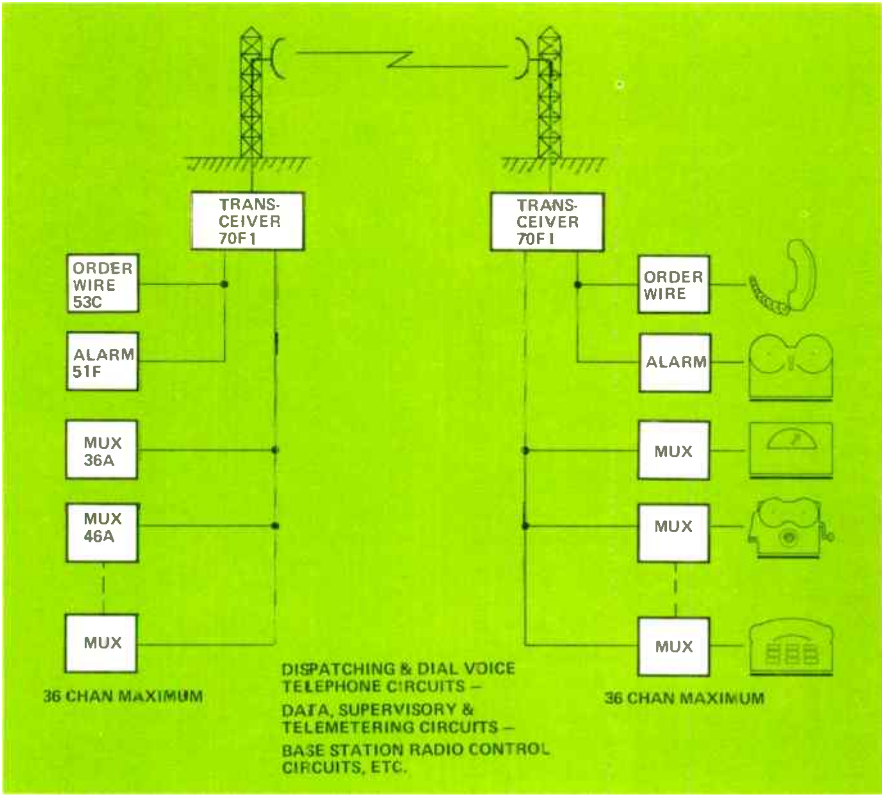

Figure 2. Typical layout of a light-route radio system.

of petroleum in a sample drops to a predetermined value, production is halted at that well head, permanently or until further drilling locates a new source of fuel.

Along the pipeline itself, status monitoring and remote supervisory control contribute significantly to the total operation, since through these functions volumetric output, petroleum flow rate, pressure, and temperature are monitored constantly. In the remotest of areas, valves are opened and closed, and pumps are automatically activated or stopped with the assistance of the microwave link.

Light-route systems may also be used in conjunction with mobile radios. Police dispatchers, for example, can relay traffic information (via operational fixed, light-route radio) from mobile units to a master station or another substation. Power companies use light-route circuits for dispatching and administrative purposes, along with such functions as remote alarming, supervisory control, relaying, and telemetering.

The operation of today's modern railroads requires that information such as telemetering, CTC (Centralized Traffic Control), and hot box indications, as well as administrative and teletype circuits, be brought in from low-density areas by the most reliable and economic means possible. Railroad companies extensively use microwave systems to this end. The status of

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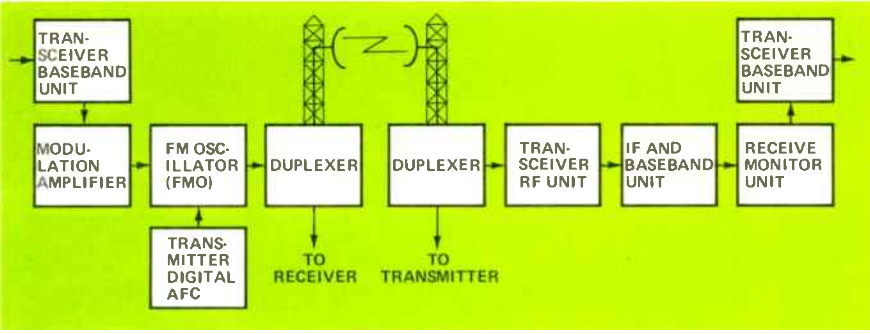

Figure 3. The basic units of the 70F1 microwave radio.

shipments and the locations of locomotives and freight cars is instantly available through the use of a microwave network tied to a central data processing center. Hot-box detection systems locate overheated journals (the rotating portion of a bearing) on moving trains. The hot-box detector transmits from remote sites, via microwave radio, information to a master station on the location of hot journals. This information is then transmitted by mobile telephone to maintenance crews along the line. An ideal use for a light-route system is in replacement of open-wire line along railroad tracks. Open-wire lines require a great deal of maintenance and expense that can be substantially reduced by use of a light-route system.

Another use of the light-route system is for telephone spur routes in a metropolitan area to reach small central offices. A tower in the center of the city provides a central transmission point from which several central offices can be serviced by light-route radios.

Three things of particular importance which a manufacturer must consider in the design of a light-route system are reliability, quality of performance, and economy. Adhering to these considerations, the GTE Lenkurt 70F1 radio, for example, contains such devices and techniques as a power fmo (frequency modulated oscillator), a digital afc (automatic frequency control), and a double-conversion receiver process.

FMO

The baseband information to be transmitted is applied to the transceiver baseband unit from multiplex equipment, or, in spur-route applications, from a backbone radio route (as when a few channels need to be branched at a repeater site). The transceiver baseband unit prepares the baseband signal, by way of level adjustments and filtering, for application to the modulation amplifier, where the conditioned signal is amplified in level to properly deviate the fmo (see Figure 3).

The fmo is a power (1 watt minimum) oscillator which provides a 2-GHz frequency modulated signal at its output. This means that the information contained in the original 36 voice channels is translated into a signal in the 2-GHz range which is of constant amplitude but of varying frequency. The fmo is frequency modulated by a varactor diode (a capacitance-varying device) which is coupled to the coaxial cavity resonator. The cavity determines the carrier frequency, and the variation in varactor

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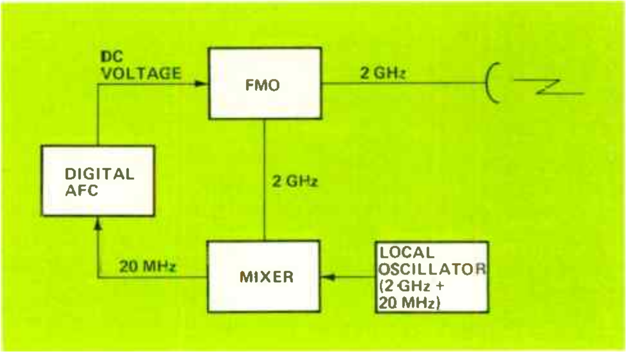

Figure 4. The digital afc controls the frequency of the fmo.

capacitance, due to the ac modulating signal, produces the frequency-modulated 2-GHz signal. This arrangement, in which the fmo produces a high power fm signal, eliminates the need for the conventional power amplifier, thus considerably reducing the cost of the system.

Digital AFC

The function of the digital afc is to control the frequency of the fmo. It does this by working within a closed transmitter loop as shown in Figure 4. The dc output from the afc drives the varactor in the fmo; the variation of dc voltage on the varactor varies the output frequency of the fmo so that it remains on the required 2-GHz frequency.

The 20 MHz from the mixer (see Figure 5) is amplified to a level sufficient to operate the digital IC's (integrated circuits). The signal is passed through a bandpass filter which allows only a band of frequencies centered around 20 MHz to pass. This prevents false locking of the afc on spurious tones such as 10 MHz or 40 MHz. The amplified 20 MHz sine wave is clipped so that distinct pulses are formed. These pulses are input to a counter, which is a string of IC's whose function is to count the number of input pulses. Each time 4096 pulses are counted, one pulse is generated by the counter. Effectively, this means that the number of input pulses is divided by 4096. The output from the counter, after processing the 20 MHz input

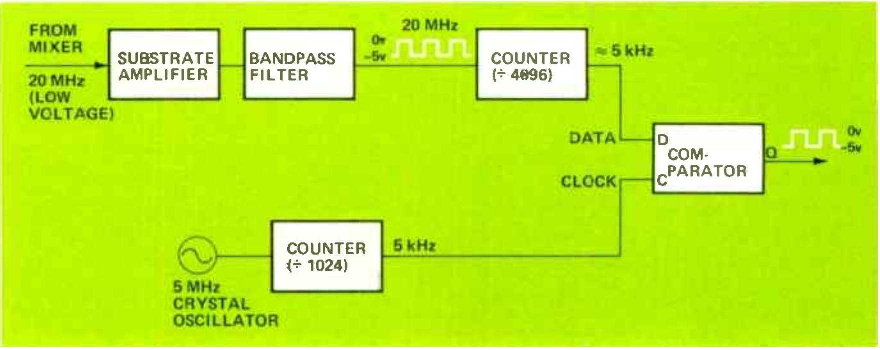

Figure 5. The basic components of the digital afc.

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signal, is approximately 5 kHz, and is applied to one lead (data or D lead) of a comparator.

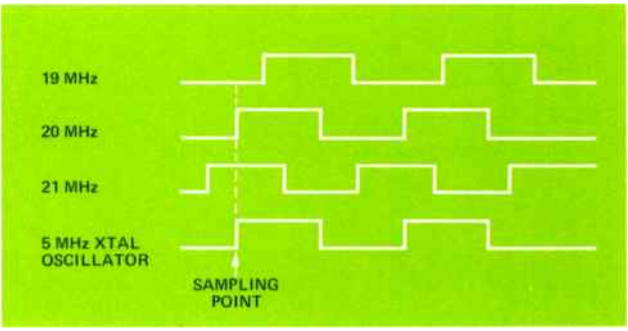
In another circuit, the output of a 5 MHz crystal oscillator is also fed into a counter which yields one pulse per each 1024 pulses of input. For a 5-MHz input from the crystal oscillator, an output pulse frequency of approximately 5 kHz is obtained. This output is fed to the second lead (clock or C lead) of the comparator. The output lead (or Q lead) of the comparator has as its output whatever binary state (high or low, one or zero, etc.) exists at the data lead when the clock pulse goes high. Each time the clock goes high, the output of the comparator will be whatever value is on the D lead. A zero on the data lead causes a zero output when the clock goes high. The output from the counter that is connected to the crystal oscillator is a constant frequency source. It is the 20-MHz output from the mixer where the variance in frequency will occur. If the mixer output were exactly 20 MHz, both outputs to the comparator would be the same. If, however, the output from the mixer were 21 MHz, the frequency from the counter would be slightly higher than 5 kHz (see Figure 6). The 21-MHz pulse stream will cause the input to the data lead of the comparator to rise sooner than would a 20-MHz pulse stream. The position in time (or phase) of the counted down output from the mixer at the time of sampling (which is determined by the output derived from the crystal oscillator source) determines whether the frequency is above or below 20 MHz. A frequency above 20 MHz will cause the comparator to give a high or 0v output, while a frequency below 20 MHz will give a low or −5v output. This output is amplified and integrated to yield a dc voltage. The variations in this voltage control the diode varactor in the fmo, which in turn keeps the output frequency stable at 2 GHz.

The Receiver

At the receiver, amplification is done directly at the input frequency by the rf pre-amplifier after being filtered by an rf bandpass filter. The pre-amplifier output is filtered by a second bandpass filter to suppress any amplifier noise generated on the image frequency (see Figure 7).

The filtered signal from the pre-amplifier is connected through a hybrid splitter to the balanced receiver mixer. The multiplied output of the LO/RO (local oscillator/reference oscillator) is injected into the mixer through the second port of the hybrid splitter. A three stage IF amplifier selects the 70-MHz product of the mixer and connects it to a second

Figure 6. The timing relationship of different signals as applied to the frequency comparator.



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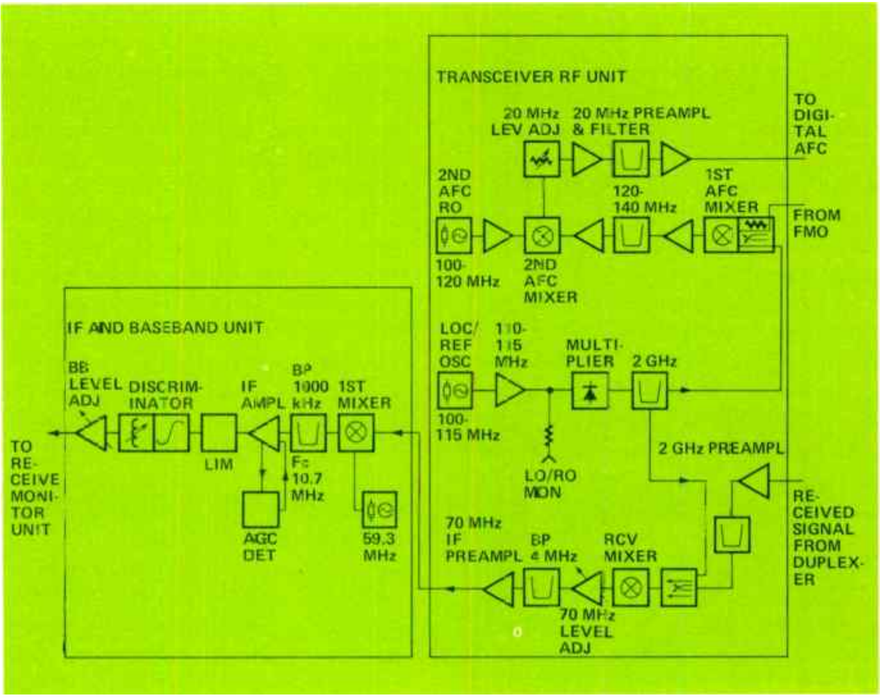

Figure 7. Amplification at the receiver is done directly at the input frequency.

receiver mixer (double-conversion process) in the receiver IF and baseband unit. The 70-MHz signal is mixed with the output of a 59.3 MHz crystal oscillator, and a bandpass filter selects the 10.7 MHz second IF signal. At 10.7 MHz, IF amplification is far easier and less expensive than it would be at 70 MHz, since transistor gain is quite adequate at this frequency, and the relative bandwidth of the filter for 36 channels is much wider (approximately 7 times wider) than it would be at 70 MHz. Also, the discriminator design is much simpler and easier to linearize at lower frequencies, and the complete circuit can be placed on thick-film substrates. The 10.7-MHz frequency modulated signal is demodulated in the discriminator and amplified to provide the necessary baseband levels.

Hot-Standby Protection

A protected system is one which has been provided with some type of security against total system failure. In many microwave systems this protection is in the form of "hot-standby" equipment which will take over the function of the main system should it fail. Protection for the GTE Lenkurt 70F1 transmitter is achieved by switching to a second radio through an rf coaxial relay, a technique that considerably simplifies the transmitter protection arrangement.

The transmit logic circuitry monitors the transmit pilot alarms, transmit afc alarms, and transmit power alarms from both transmitters. If an alarm occurs on the primary transmitter, the logic will cause the output to switch to the standby transmitter. If both transmitters are in an alarm condition, the

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transmitter with the lowest priority alarm will continue to operate.

The receive logic circuitry monitors the receive pilot alarms and receive noise alarms. If a pilot alarm occurs, the logic will cause the receiver in alarm to be muted and the other receiver to be switched on line. If pilot alarms occur in both paths, no switching takes place.

A receive noise alarm occurs when the noise exceeds the preset level (nominally 58 dBrnc0). A receive noise alarm will cause the receiver in alarm to be muted and the other receiver to be switched on line. If both receivers have noise alarms, both receive paths will be muted.

Testing

Calibration of a microwave system is greatly simplified by use of a loop-back type of test set, such as the GTE Lenkurt Type 48615 Transceiver Test Set. This set enables one person to calibrate the system from one location rather than requiring one person at each location.

The first function of the test set allows a receiver deviation check. The receiver baseband level is set up with a calibrated 70-MHz frequency-modulated signal, which is used as a reference; transmitter deviation is then adjusted with reference to the calibrated receiver. In the loop-back function, the near-end transmitter transmits a signal into the test set where the rf frequency is shifted (usually by 50 MHz), and loops the signal back to the near-end receiver, where the proper deviation adjustments can be made for the system. The use of the test set is a cost savings, since one person can perform noise loading, set levels, and accomplish routine radio checks from one location.

The technology that has been developed and applied to the design of high-density microwave systems can also be applied to light-route radio systems. These systems can now provide the industrial and common carrier user with an economical form of communication which is both reliable and of high quality.

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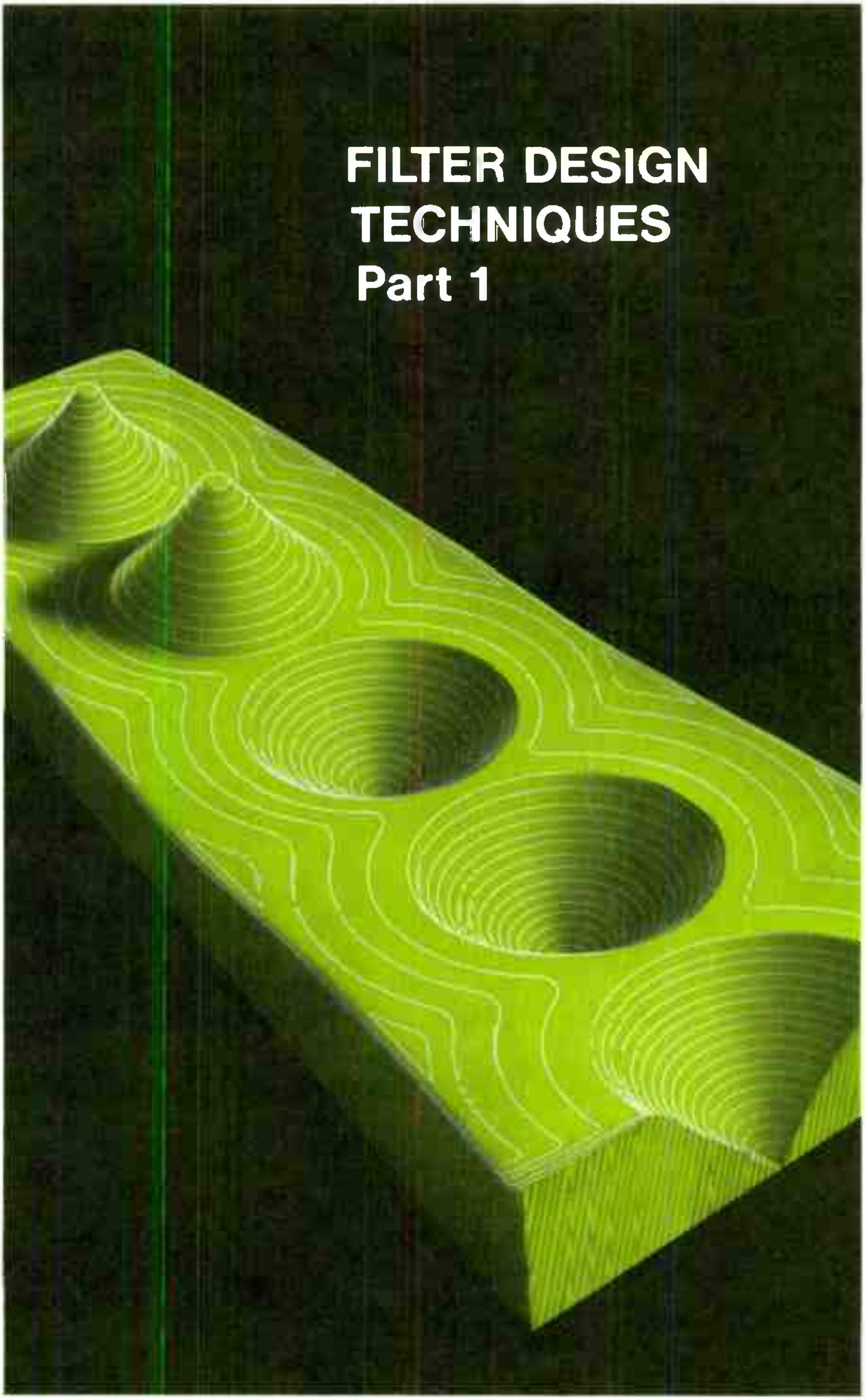
FILTER DESIGN TECHNIQUES Part 1



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Filters in modern technological applications serve to direct, channel, divide, integrate, and transform, but in all of these functions the basic principle remains the same: selective processing of electrical energy and signal information.

Filters are essentially devices with particular transmission characteristics which can be designed to accept or reject – to selectively process – portions of the electromagnetic frequency spectrum. In this, filters behave like interconnected tuned circuits, which are combinations of resistance, inductance, and capacitance whose reactances produce unique frequency responses.

When an alternating current source is connected to a tuned circuit, the combined effect of capacitive reactance (X_C), inductive reactance (X_L), and resistance (R) constitutes an impedance (Z) – an opposition to the flow of alternating current – which varies with the frequency of the applied source. At low frequencies, X_C is much greater than X_L or R, while at high frequencies, X_L is the largest element. At some intermediate frequency (f_o), X_C equals X_L, and the circuit is said to be resonant, or in a state of resonance.

The effect of the resonant state on a tuned circuit depends in great part upon the manner in which the components are connected. The impedance of a series tuned circuit appears as a large inductive or capacitive reactance at frequeneies above and below resonance, thus limiting the current flow in the circuit. At resonance, the effects of X_L and X_C cancel one another, leaving the impedance at a minimum level – equal to the value of R – and allowing maximum current flow (see Figure 1a).

At frequencies below resonance in a parallel tuned circuit, maximum current passes through the low reactance of the inductive component, as at high frequencies current passes readily through the low reactance of the capacitive element. At resonance, however, the reactances are equal, resulting in the lowest level of current through the circuit (see Figure 1b).

While a tuned circuit produces an amplitude peak or trough at a single resonant frequency, passive filter networks – which are also treated as combinations of resistances, inductances, and capacitances – provide constant transmission for a range of frequencies (the passband) and a high degree of attenuation to all other frequencies (the stopband). A low-pass filter, for example, is intended to reject all signals above a specific cutoff frequency; to accomplish this, it produces increased attenuation to the signal as the source frequency is increased (see Figure 2), much as impedance increases in a parallel-resonant tuned circuit as resonance is approached. A high-pass filter, on the other hand, is designed to reject all signals

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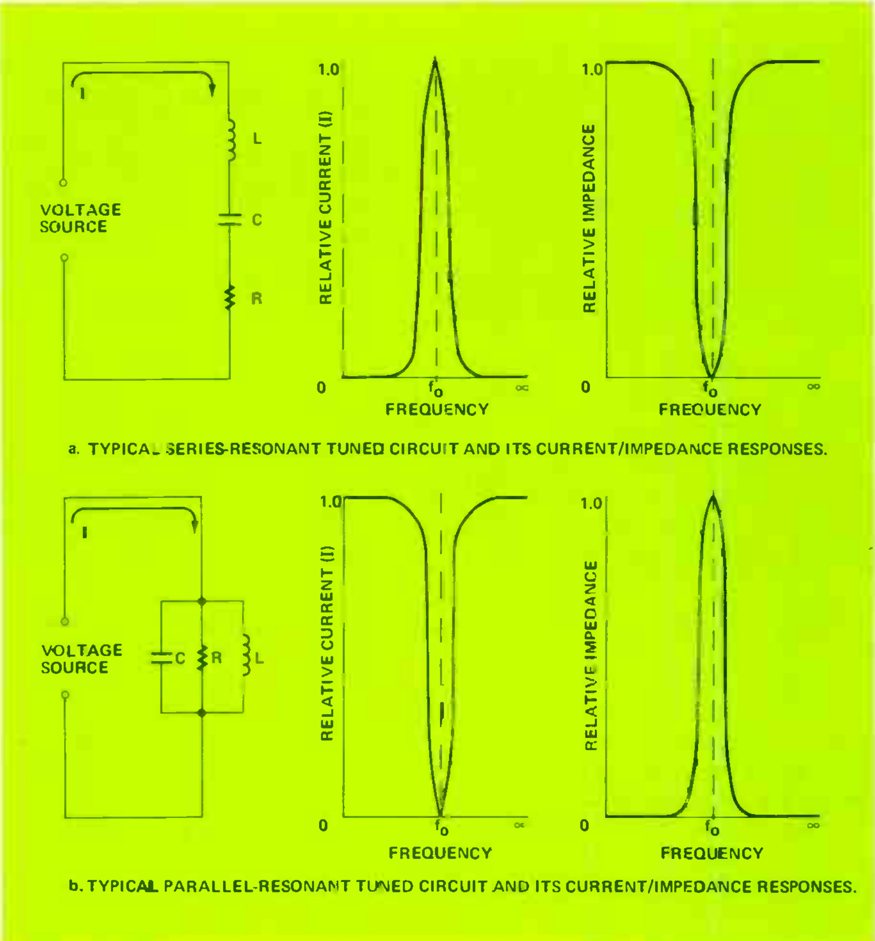

Figure 1. The interaction of component characteristics causes a peak or trough in the current/impedance response of a resonant circuit at one frequency (f_0). Theoretically, the maximum amplitude achieved by either element is infinity.

below cut-off, so its attenuation decreases with frequency, as the impedance of a series-resonant tuned circuit decreases when resonance is reached.

Thus, one way to view filters is as a collection of resonant circuits, the sum of whose resonance characteristics determines the overall performance of the network. This is essentially the approach taken in the image parameter theory of filter design.

Image Parameter Theory

The image parameter approach to filter design grew out of the study of transmission lines in the early days of electromagnetic communications. At that time, long lengths of wire were used to send telegraph and telephone signals, so it was natural that they should become the focus of considerable study. It was soon recognized that the action of a transmission line, re-

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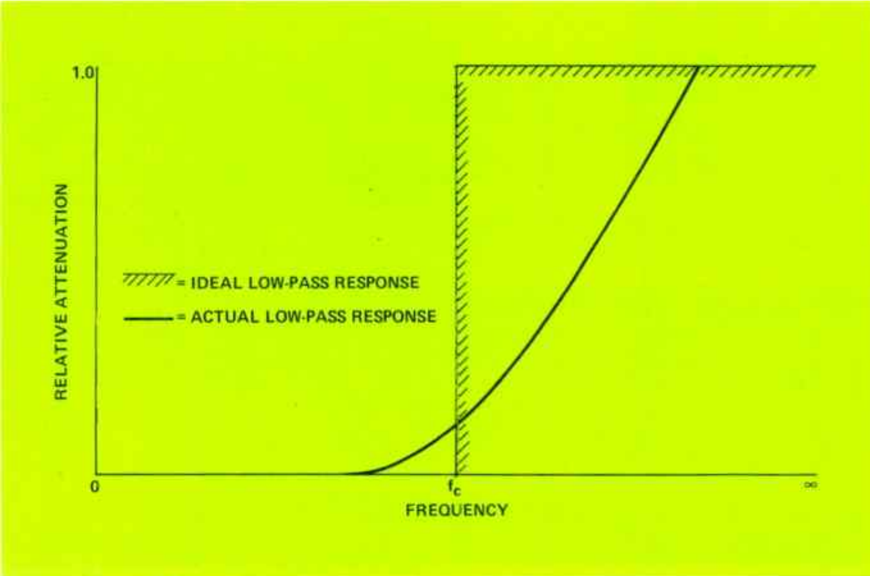

Figure 2. Low-pass filters reject signals above a specific cut-off frequency (f_c) by providing increased attenuation. The ideal is instant rejection at f_c; the actuality is a curve increasing with frequency.

gardless of its length, can be explained in terms of resistance, reactance, and impedance, although lumped physical components may not be present. Dividing such a line into successively smaller segments, a basic unit is eventually arrived at which can be described as a network with two input terminals and two output terminals, with each pair of terminals defining a port (see Figure 3). This two-port transmission network, which is one of the most common types of simple filter, possesses transmission and impedance properties which depend upon two quantities: image impedance and image transfer function.

Image Impedance

Image impedance is the filter network equivalent of the characteristic impedance of a transmission line.

The characteristic impedance, Z_o, of a line is that opposition to ac which, when used to terminate one end of the line, causes the input impedance to be of the same value. Image impedance, Z_i, is actually an extension of this concept, taking into account the non-symmetrical nature of many filter networks, where the im-

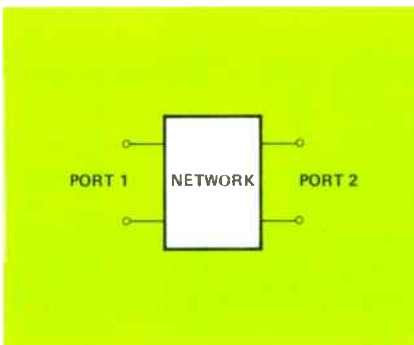

Figure 3. A four-terminal network has two ports, either of which may be used as an input point.

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pedance looking into one port may differ from that looking into the other. In such asymmetrical networks, there are two image impedances, one for each port. Where a Z_o termination of a transmission line causes a Z_o input impedance, terminating one port of a filter network causes the other port to exhibit its own Z_i as the input impedance; that is, one Z_i appears as the image of the other (see Figure 4). A symmetrical network closely resembles a transmission line, in that the image impedances of both ports are equal (terminating one port in its Z_i results in the same Z_i appearing as the input impedance).

The image impedance of a given port can be defined as the geometric mean of the network short and open circuit conditions. That is, in a two-port network, the image impedance of port 1 is expressed as

$$\sqrt{\frac{Z_{11}}{Y_{11}}}$$

and the image impedance of port 2 is

$$\sqrt{\frac{Z_{22}}{Y_{22}}}$$

where Z_{11} is the impedance presented to an input signal at port 1 with port 2 open, and Y_{11} is the admittance (inverse property of impedance) presented at port 1 with port 2 shorted. Z_{22} and Y_{22} express the same properties when the input is at port 2 and port 1 is open or shorted. Image impedance, then, approximates the impedance of a network in a state half way between short and open, and represents an average figure for the range of conditions between the two extremes.

As the basic filter unit in image parameter design, the half-section is the smallest two-port combination of elements exhibiting an image impedance characteristic. In image parameter

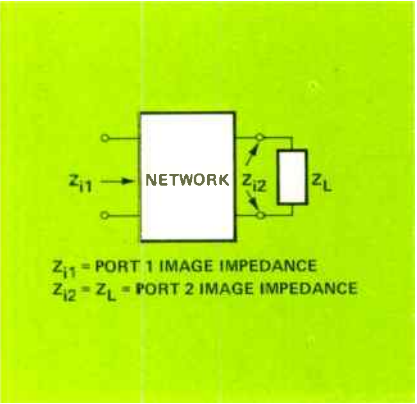

Figure 4. When one port of a two-port network is terminated in its image impedance, the image impedance of the other port becomes the network input impedance.

filter design, half-sections are combined in cascade to produce the required filter response. In connecting the output terminals of one half-section to the input terminals of the next (cascade connection), it is necessary that the image impedances be matched through the length of the filter. In this way, maximum power transfer can be attained at passband frequencies, where the image impedances appear.

For example, the half-section for a basic low-pass constant-k filter — in which the product of the impedances is independent of frequency — would consist of a series inductance and a shunt capacitance (see Figure 5a), producing a stopband with attenuation increasing to infinity with frequency, as shown in Figure 2. To make a full low-pass filter section, two half-sections are joined so that their image impedances are properly matched (see Figure 5b); successive sections can be added to provide whatever response characteristic is required.

The m-derived half-section is similar to a constant-k unit with a resonant

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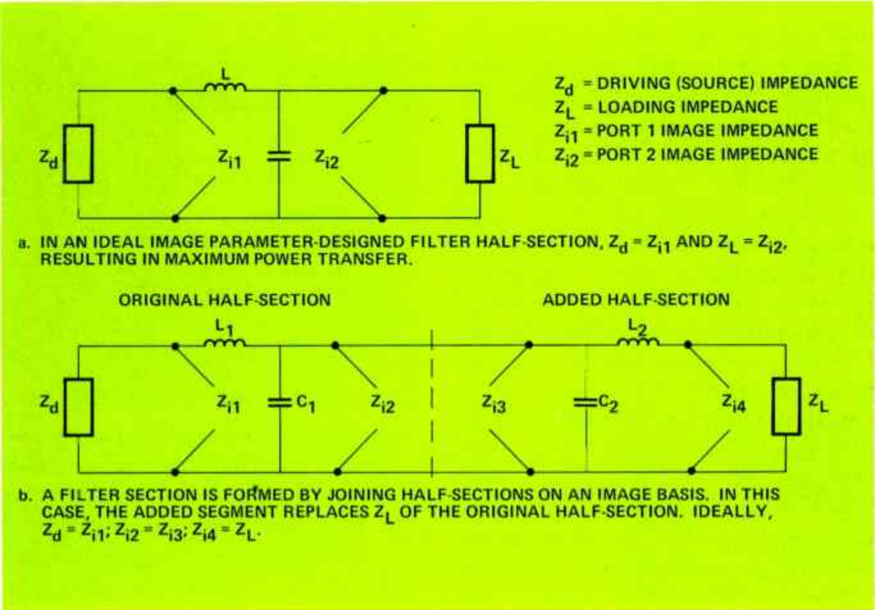

Figure 5. In image parameter filter design, it is assumed that the various half-sections are terminated in their image impedances.

segment introduced (see Figure 6a). The stopband of this filter has a peak at the resonant frequency of the tank circuit, with the rest of the band exhibiting constant-k type characteristics (see Figure 6b). The advantage of the m-derived filter is that several m segments can be added and the attenuation peaks – which theoretically go to infinity – can be placed anywhere in the stopband, providing extremely sharp cutoff points and increased attenuation at frequencies where the circuit design requires them. This allows the filter to produce only the stopband attenuation levels required, eliminating the components that would be necessary if infinite attenuation were sought at all stopband frequencies.

Image Transfer Function

Closely related to the image impedance of a filter is the image transfer function (τ), which is a measure of the response of a two-port network when it is terminated at both ports – driven and loaded – by its image impedances. The image transfer function consists of two terms: an "image attenuation constant" (α), and an "image phase constant" (β). These elements are related by the expression $\tau = \alpha + j\beta$ (where $j = \sqrt{-1}$), so that if both ports of the network are terminated by their image impedances, the attenuation and phase shift seen in the output constitute the image transfer function.

When a series of half-sections are joined to form a more complex filter, and the filter is terminated at both ports by impedance corresponding to the Z_i of the network, then the resultant impedance appears nominally resistive to passband frequencies. In this case, power is absorbed from the source and transferred to the load, with the amount of transference de-

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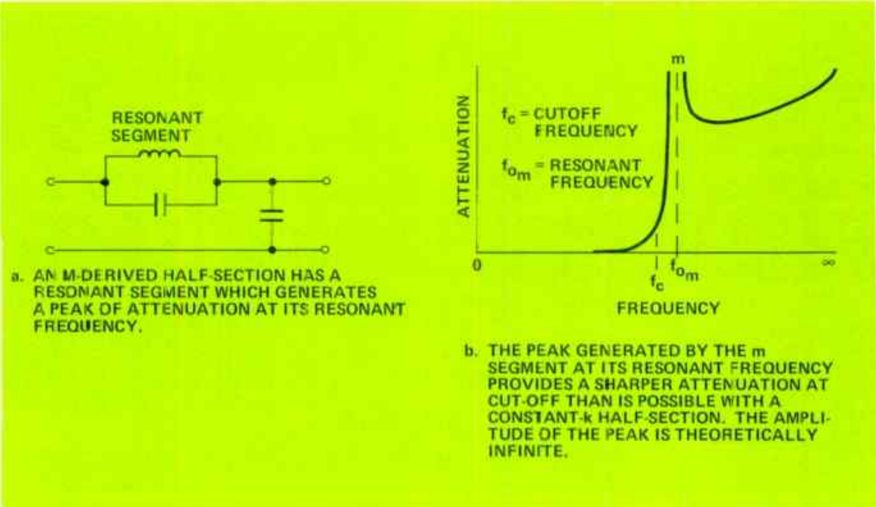

Figure 6. The m-derived filter is the most common image parameter design because of its attenuation peaks. By joining half-sections into a complex filter, several peaks can be established to make optimum use of the stopband with minimum physical components.

pendent upon the attenuation introduced by ∝. The closer the impedances in the network are matched, the lower the attenuation factor and the greater the power transfer.

To frequencies outside of the passband, Z_i appears as a reactance, which cannot dissipate power. As a result, no power can be transferred and the stopband frequencies are effectively rejected.

Image Parameter Design

The image parameter technique has been in use for so long that its application is fairly well systemized. The first step in designing an image parameter filter is the use of templates to generate a curve, on specially designed graph paper, representing the required image attenuation constant, ∝, in dB, and the points in the frequency spectrum at which the m-sections will be resonant to produce their characteristic peaks. The m-derived section is much more commonly employed in image parameter filters than constant-k types because of its rapidly rising and frequency-variable attenuation peaks.

The design technique is basically a trial-and-error method in which the templates are moved around until the sum of all the stopband peaks approximates the network attenuation requirements (see Figure 7). Then, because a perfect impedance match is impossible to achieve, slight adjustments – "fudge factors" – are introduced to make the design more realistic. Once the stopband peaks have been established, the value of m which will provide each peak is known. From this, it is possible to calculate the component values that would resonate at the proper frequencies, and thus arrive at an optimum design for the m-sections. Values for the other elements in the filter can be derived from the known network cutoff frequency.

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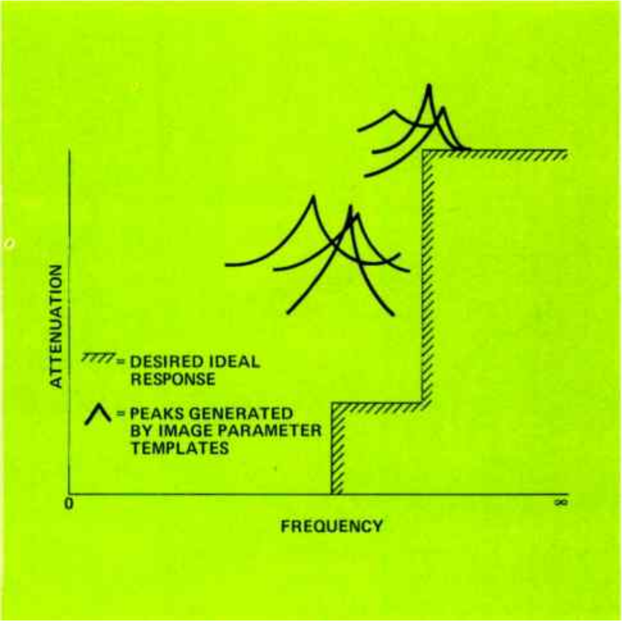
Figure 7. In designing an image parameter filter, attenuation peaks are generated by templates on special graph paper to approximate the desired ideal response, as shown by these approximations for a low-pass filter.



When the image parameter-designed filter is built, it provides a fairly good impedance match to the driving and loading impedances in the passband. When the match is exact, the passband response is at its best – that is, attenuation is minimum. This condition, however, cannot be achieved with any certainty in practice because real network terminations are generally resistive, rather than frequency-dependent impedances. Thus, although an impedance match may be obtained when the network image impedances equal the resistive terminations, the match at other frequencies in the passband is not exact, which can result in badly degraded response. This degradation causes an arbitrary ripple shape to appear in the passband, making it difficult to predict the response of a given filter before it is actually built (see Figure 8). For this reason, image parameter design is today generally limited to constant resistance networks such as all-pass filters. For other applications, modern passive filter design is accomplished with synthesis techniques.

Filter Synthesis

Modern filter design concerns itself with idealized mathematical models which describe filters in terms of complex current and voltage relationships. The elements in the models are defined in terms of a complex frequency variable, s; when the model equations have been solved, mathematical transformations are used to extract component values which would realize the model in physical terms.

The complex frequency variable is expressed as:

$$s = \sigma + j\omega$$

where σ is a real component and $j\omega$ is an imaginary component. It should be noted that the imaginary factor is not physically imaginary; the terms "real" and "imaginary" are mathematical designations for two distinct parts of a

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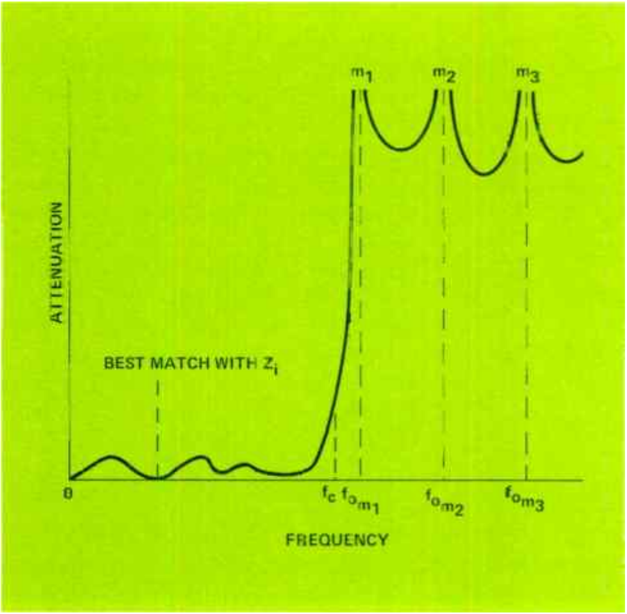

Figure 8. The passband response of an image parameter-designed filter may exhibit an arbitrary ripple shape, as shown in this response curve for an m-derived filter with three m-resonant components.

complex quantity or function, and do not indicate actual existence or nonexistence. A term is considered imaginary when it is related to a strictly mathematical construct, $\sqrt{-1}$, which is represented by the symbol j. Thus, in the expression $s = \sigma + j\omega$, the "real" and "imaginary" factors define two actual aspects of a particular complex frequency.

In relation to frequency, an ideal filter is one which produces no loss of transmitted energy within its passband, yet provides infinite attenuation of stopband frequencies. Such ideals are, of course, unrealizable, given the non-ideal nature of physical components. The problem, therefore, is to achieve a filter design which approximates as closely as possible the ideal for a given application.

Transfer Function

Synthesis filter design begins with an attempt to find a transfer function providing the best approximation of the required filter attenuation and phase properties. The transfer function expresses a mathematical relationship between a filter output quantity and an input quantity; most commonly these quantities are voltages, but they could as easily indicate a current-to-voltage or voltage-to-current ratio. Essentially, the transfer function is a measure of how efficient – or inefficient – a filter is at transferring a quantity at its input to its output port.

In discussing passive filters, it is most convenient to consider the transfer function as an input-to-output ratio; this allows attenuation, rather than gain, to be dealt with.

Represented by the symbol T(s), the transfer function of a passive filter can be expressed in the form:

$$T(s) = \frac{P(s)}{Q(s)}$$

where P(s) and Q(s) are polynomials – algebraic terms consisting of a con-

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stant multiplied by variables – representing the input and output quantities, respectively. Through various mathematical manipulations, the P and Q polynomials can be treated in such a way that their roots – those quantities which are multiplied together to form the algebraic terms – take on frequency characteristics. Designating the roots of P as p_1, p_2, etc., and the roots of Q as q_1, q_2, etc., the transfer function can be written:

$$T(s) = \frac{(s-p_1)\,(s-p_2)\ldots(s-p_p)}{(s-q_1)\,(s-q_2)\ldots(s-q_q)}$$

where s is the complex frequency variable and the polynomial roots are complex frequencies. Thus, when the value of complex frequency variable s equals the value of any of the roots of P, the numerator becomes zero, resulting in a zero transfer function. For example, if the complex frequency represented by p_1 is the same as that represented by s, then $(s-p_1) = 0$. The product of any number of factors multiplied by 0 is still 0, so $(s-p_1)(s-p_2)\ldots(s-p_p) = 0$. Zero divided by any other number is zero, so where $s = p_1$, the transfer function T(s) is zero, which also is true when s equals any of the other roots of P.

By the same reasoning, when s equals any of the roots of Q, the denominator becomes zero and T(s) has an infinite value (any number can be infinitely divided by zero).

Depending upon what quantities are being related, p_i and q_i (any of the roots of P and Q) represent poles or zeros; if gain is being considered, p_i identifies a "pole of transmission" and q_i identifies a "zero of transmission." When dealing with passive filters, however, it is more convenient to consider attenuation characteristics, since there is no gain in a passive filter; p_i in this case is used to identify a "zero of attenuation" and q_i a "pole of attenuation." This is, in fact, a logical step, since an attenuation pole – a point at which infinite attenuation is presented to a signal – would necessarily produce a zero of transmission. In the following discussion, the poles and zeros of attenuation are being considered. It is thus possible to define a transfer function in terms of its composite poles and zeros, reducing the need to cope with complex polynomial expressions.

Poles and zeros are plotted in what is called the s-plane, where the horizontal axis is σ and the vertical axis is $j\omega$ (see Figure 9); the critical frequencies at which poles appear are theoretically points of infinite attenuation and the zero appearances are critical frequencies exhibiting zero attenuation. The zero critical frequencies are the points at which the filter would oscillate. If the zeros were to appear in the right half of the s-plane, it would indicate that the response of the network grows without bound for any input; in other words, the network would be unstable. A zero appearing on the $j\omega$ axis would also imply that the network oscillates with no input signal. Since both of these events are impossible with passive elements, the zeros are limited to the left half of the s-plane plot.

Synthesis Design

The first step in the synthesis filter design technique is solving what is called the approximation problem.

Since the network being synthesized consists essentially of lumped reactances terminated at each end by a resistor (see Figure 10), the power applied to the network must either be transferred to the load or reflected back to the source, since no power can be dissipated in an ideal reactance. In actual practice, complete power trans-

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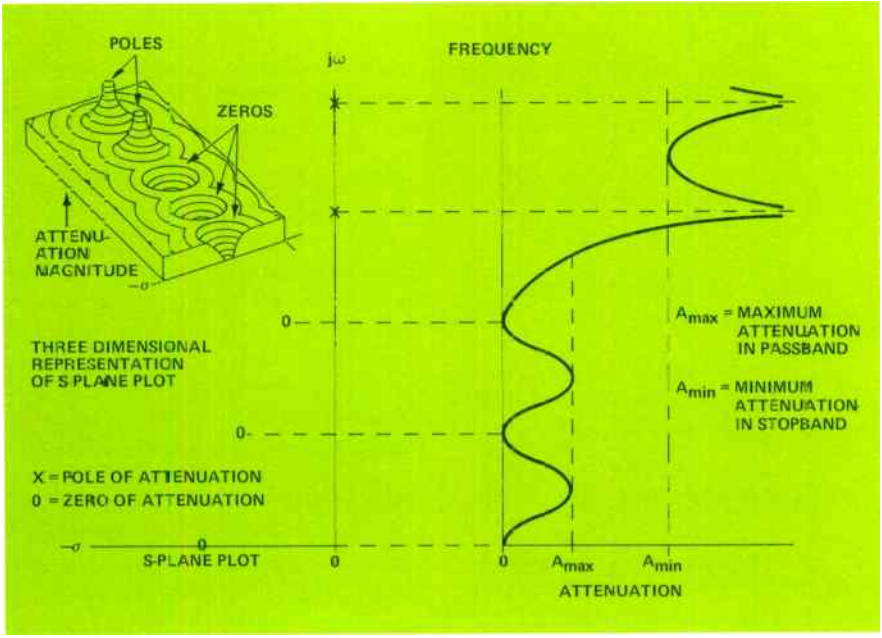

Figure 9. From pole-zero information, it is possible to ascertain the response of a projected filter design, as shown in this plot for a lowpass filter. In practice, filter design is most commonly done with mathematical formulae rather than plots.

fer does not occur, so there is always a reflected component to be considered in the design of a filter; this component is expressed in terms of a reflection coefficient, ρ_i. The reflection coefficient can be defined as the ratio of two polynomials representing maximum possible power (e) and the difference between maximum possible power and power actually delivered to the load (f). Expressed in relation to the complex frequency, these polynomials are related by:

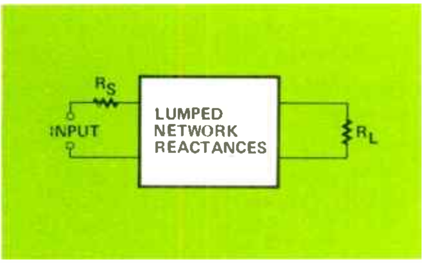

Figure 10. A filter network is essentially a group of reactances terminated by a load (R_L) and source (R_S) resistance. Unlike an image parameter-designed filter, the terminations in synthesis design are not constrained to match the network impedance characteristic.

$$\rho_i = \frac{f(s)}{e(s)}$$

Under actual design conditions, ρ_i is usually given as a percentage figure: a 100% reflection coefficient means that all of the power is reflected back to the source, and a 0% coefficient means that all of the power is transferred to the load.

The total power in a reactance network is always constant, and can be

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determined from the sum of the reflected and transferred powers. This is stated in the Feldtkeller relationship:

$$H(s)H(-s) = 1 + K(s)K(-s)$$

where H is a transmission transfer function having to do with the maximum transfer of power, and K is a characteristic function indicating loss in the network. These functions are defined in terms of the reflection coefficient: H is the ratio of the maximum power available for delivery to the load, e(s), to the power actually delivered, p(s):

$$H(s) = \frac{e(s)}{p(s)}$$

and K is the ratio of maximum deliverable power minus power actually delivered, f(s), to power actually delivered to the load:

$$K(s) = \frac{f(s)}{p(s)}.$$

Thus, the transmission and attenuation functions can be treated as ratios of polynomials capable of generating poles and zeros. Since the poles and zeros are contained within the H and K terms, it is not necessary to create an s-plane plot to deal with them; they can be more easily handled in this form by mathematical manipulation.

When a filter is specified, it is usually in terms of its loss or its phase/delay response. If the loss response is known, a value for K can be determined and from this, H can be derived; K can likewise be obtained from H when the phase/delay response is specified. Once values of H and K are known, both sets of polynomials can be factored into odd and even parts, and the parts combined to give required input impedance functions. For example, dividing the difference between the even parts of H and K by the sum of the odd parts gives a value for the input impedance of port I with a far-end open circuit (Z_{11}):

$$\frac{H_{even} - K_{even}}{H_{odd} + K_{odd}} = Z_{11}.$$

From the impedance functions, component values can be calculated. In practice, most of these design steps are now done by computer calculation, greatly increasing the complexity of the filters which can be designed, and providing the special mathematical handling required to conserve accuracy in the design of large filters.

Design Realization

Once a filter has been designed, it must be constructed – realized – using physical components. During the design process, these components are assumed to be resistors, capacitors and inductors; in reality, the resistive element is quite often present in distributed form only, such as the resistance in an inductor's windings, so that the realization of an RLC filter network is generally composed of only inductors and capacitors. Figure 11 shows typical realizations for four of the most common filter types: the low-pass, which rejects signals above a specific cut-off frequency, the high-pass, which rejects signals below cut-off, the band-pass, which rejects signals on either side of a selected passband, and the band-stop, which passes signals on either side of a selected stopband. Among the myriad uses of such LC filters are receiver input preselection, suppression of unwanted sidebands and harmonics, impedance matching, and multiplexing.

The realization of a filter design is not required to consist of inductors and capacitors. Depending upon overall circuit applications, portions of the network may be replaced by crystals,

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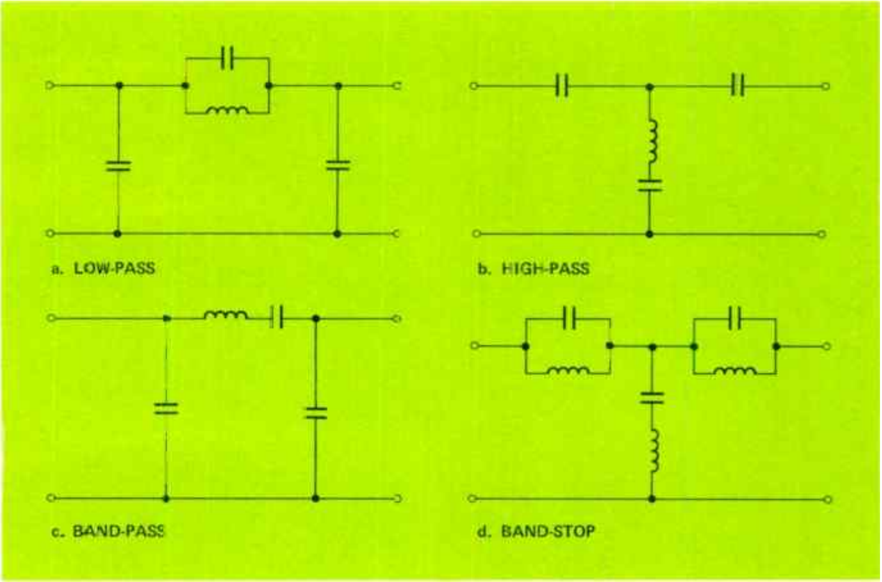

Figure 11. The basic filter types can be realized as strictly LC structures, as these typical networks illustrate.

mechanical resonators, and operational amplifiers (for active filters), or the entire network may consist of a wave-guide cavity or a space in a strip transmission line.

Crystal Filters

Certain crystalline materials – notably quartz – exhibit a piezoelectric property, in that they can be set in mechanical resonance by an electrical field; the frequency at which this resonance occurs is a function of the crystal's size and the manner in which it is cut. This transducing property, which allows translation of electrical signals into mechanical energy and vice versa, makes it possible to realize a filter with crystal resonators – plates of piezoelectric material with a metal electrode on each side – rather than electrical components, since the frequency at which the crystal resonates is effectively passed while adjacent frequencies are blocked.

In designing a crystal filter, the image parameter and/or synthesis techniques are first used to obtain component values for an equivalent LC filter; information on crystal size and shape, electrode dimensions, and electrode placement is then used to transform the LC network into a crystal resonator. Several such resonators can be coupled together to form a filter with the response characteristics required by a given application.

Crystal filters are widely used in the telecommunications industry to provide channel separation in multiplex systems because of their low cost and size reduction compared to LC networks. These advantages are obvious in those structures which combine several elements into one physical unit, such as GTE Lenkurt's polylithic filter.

Microwave Filters

The uses of filters in the microwave frequency range are the same as those

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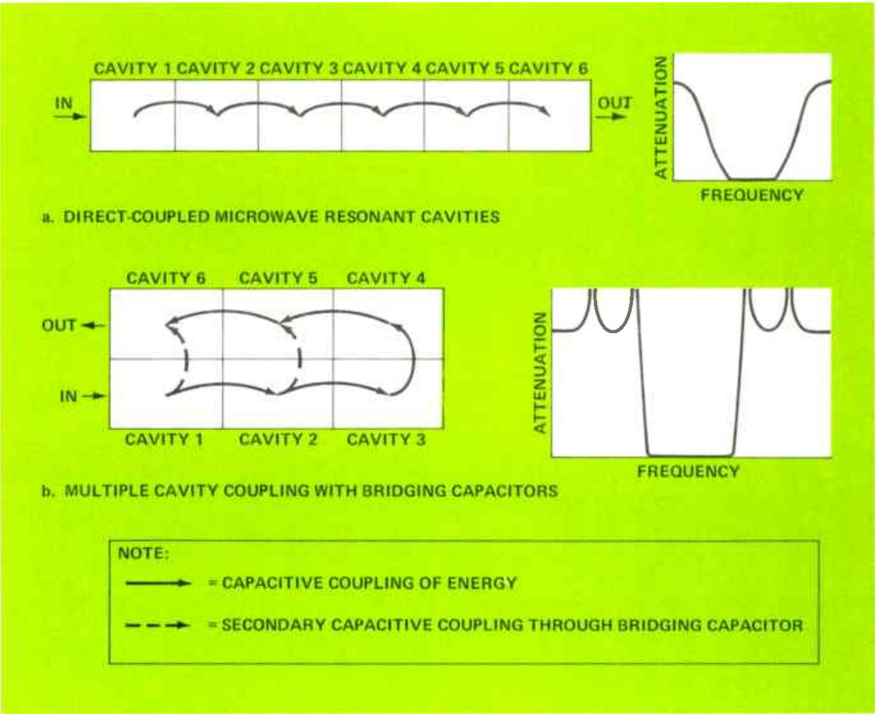

Figure 12. In an experimental approach to multi-cavity filters, bridging capacitors allow cancellation of portions of the microwave energy, producing attenuation peaks which improve the overall filter performance.

at lower frequencies: they can be used to select, reject, and channel electromagnetic energy of different frequencies, and provide for maximum transfer of that energy from one point to another.

In the design of microwave filters, the lumped inductive, capacitive, and resistive components of lower-frequency networks are replaced by the characteristics of such typical microwave circuit elements as waveguides, coaxial and strip transmission lines, and resonant cavities. The nature of these microwave elements is such, however, that under certain conditions their frequency behavior can be made to approximate that of lumped components. Because of this, the most common approach to microwave filter design is the derivation of structures from equivalent lumped-element filters generated by image parameter or synthesis techniques. Indeed, if a lumped-element filter is designed to have the same frequency response characteristics as a microwave filter, it is possible to replace almost every lumped element with an equivalent microwave element.

Although derived from them, microwave filter realizations do not physically resemble lower-frequency LCR networks at all. The elements seen in microwave circuits consist of such devices as stubs – short- or open-circuited sections of transmission line used as reactive components –

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and resonant cavities – regions of dielectric surrounded by conductive walls. For example, low- and band-pass filters used at lower microwave frequencies can be realized by open- and short-circuited sections of coaxial line. At higher frequencies, resonant cavities can be coupled together to produce required responses. In one such application, waveguide sections are constructed with cavities built into them; to allow for adjustment of each cavity's resonant frequency, screws are often extended through the waveguide wall to vary the capacitance of the cavity. Variations of this technique are currently under development which will produce a filter frequency response whose stopband contains the attenuation peaks – and the advantages these provide – found in m-derived sections at lower frequencies. One of these new techniques involves coupling resonant cavities not only in cascade, but also in parallel (see Figure 12), allowing cancellation of energy at different points in the filter.

This discussion has been concerned with the design of passive filters – filters with no source of energy within the network – according to image parameter and synthesis techniques. In Part 2, the design of active filters will be discussed.

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