GTE LENKURT
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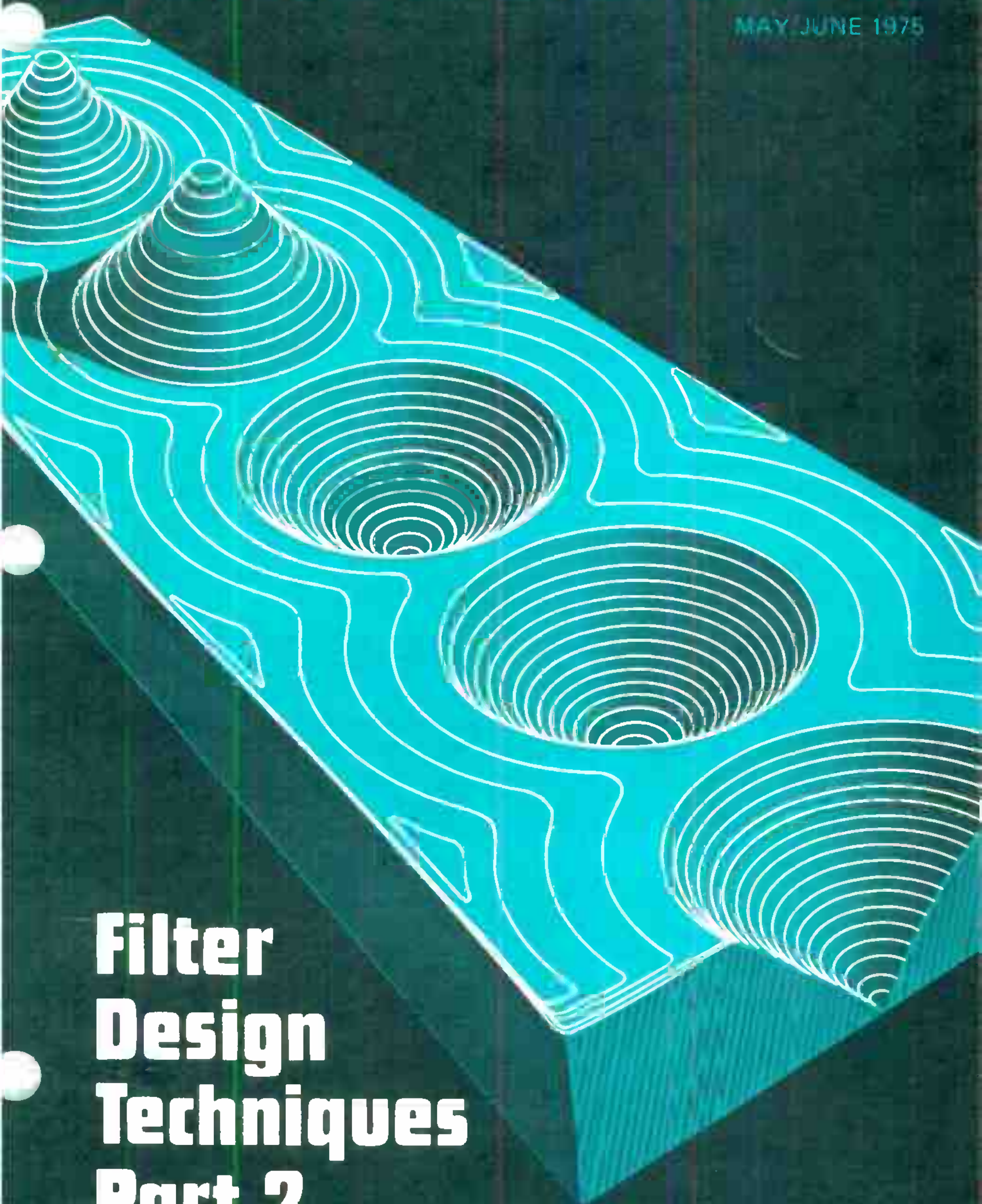


Filter Design Techniques Part 2

Also In This Issue:
RESONANT TRANSFER

Progress in the miniaturization of electronic circuitry has necessitated new approaches to the selective processing of electromagnetic waves. Among the most effective are those which make use of active elements in RC networks.

The April, 1975, issue of the *Demodulator* began a discussion of the techniques used in designing electromagnetic wave filters. It treated them in terms of their resonance characteristics and attenuation functions, restricting itself to filter networks containing only passive components. In this issue, the discussion is continued, concentrating on the design of active filters having lumped resistive-capacitive (RC) components; this is one of four basic types of active filter, the other types being: active filters with distributed RC components; N-path filters, in which network parameters are time-variable; and digital filters, in which digital processing structures are used as signal-wave filters.

RC Networks

In the design of passive filter networks, it is assumed that the physical construction – the realization – of the design will consist of inductors, capacitors, and resistors connected in an RLC network. Since the functional resistive element may only be present in a distributed form, such as the resistance in an inductor's windings, rather than as a lumped component, it is a common practice to consider only the reactive, or LC, filter elements. In many cases, however, the physical structure of inductors makes the realization of such passive RLC and LC networks extremely difficult; for example, the bulk of a low-frequency inductor makes it incompatible with many of the modern integrated and printed circuit techniques. As an alternative, filters can be designed without an inductive element; that is, as strictly RC networks (see Figure 1). Such networks have the advantage of small physical dimensions, low cost, and relative insensitivity to interference from external electrical forces.

Without an inductive element, however, the properties of the RC network cause its frequency selectivity to be very poor and limit its range of application. In addition, it requires a large number of components to produce a response equivalent to that of an LC network, thus negating its size advantage. These deficiencies can be overcome through the use of one or more active devices, which are essentially amplifiers, as elements of the filter network.

An ideal active RC filter network can be most generally represented as an interconnection of passive resistive and capacitive elements, with active devices serving to compensate for the absence of inductors; the function of such a network is to selectively process

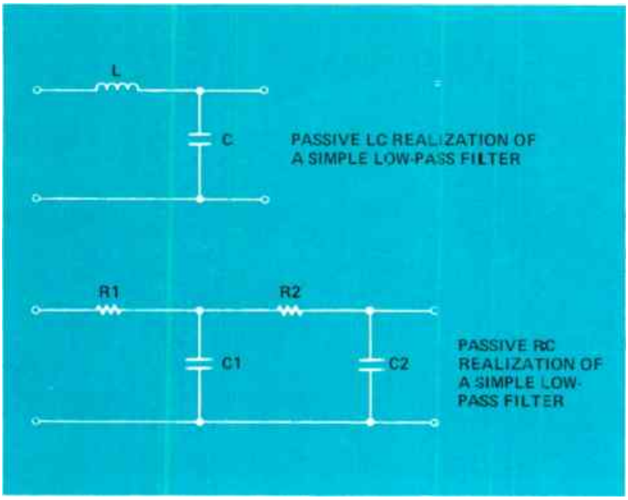


Figure 1. Passive filters can be realized as either inductive-capacitive or resistive-capacitive networks.

signals according to their frequency composition.

Techniques for designing active RC filters fall generally into two broad categories: simulation, which normally requires that an active circuit replace and perform the function of an inductor or other energy-storage element, and biquadratic filtering, in which the transfer function required in a given application is broken down into biquadratic factors – mathematical components of the transfer function – and filter sections realizing these factors connected in cascade.

Active Filter Simulation

The design of an active RC filter by simulation involves two basic steps: (1) designing a passive LC filter with either the image parameter or synthesis method (see the April, 1975, *Demodulator* for a discussion of these techniques), and (2) replacing each inductor in the passive LC filter with an active RC circuit capable of simulating the response of the replaced inductance. A variation of the second step results in the replacement of the inductors by resistors, resistors by capacitors, and capacitors by active circuits called frequency-dependent negative resistances (FDNR's).

The simulation approach allows use to be made of the well-established passive filter design techniques, which simplifies the process, and also takes advantage of the LC filter's inherent frequency stability and normally low sensitivity to variations in component value. The latter is quite often the deciding factor in determining what type of active circuit will be utilized in a given filter realization.

The problem of stability does not occur in passive filter networks because they contain no internal power sources. An active filter, however, may break into oscillation when any of its parameters, such as the values of passive components or the gain of active devices, change. The network transfer function of an active filter is also sensitive to changes in these parameters; even when stability is not affected, such variations may produce filter behavior totally different from the design requirements. The active circuits used in the simulation technique are thus evaluated not only in

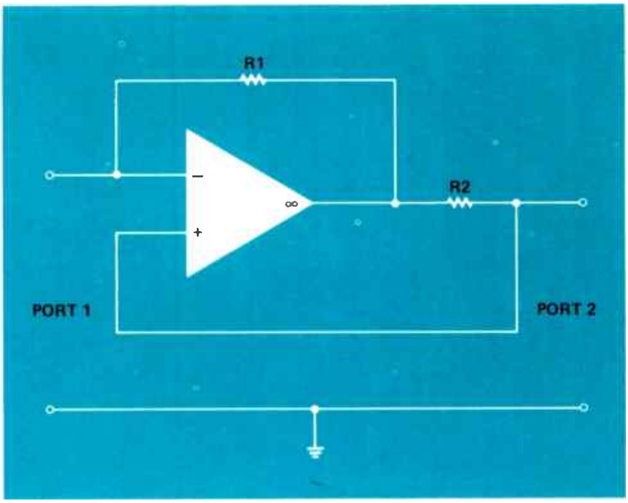


Figure 2. Realization of a current-inverting negative impedance converter (INIC) using an operational amplifier.

terms of their simulation properties, but also according to their sensitivity to parameter changes.

Once the LC filter is designed, it is necessary to decide upon the most suitable inductor-simulating circuit. Because an inductor is essentially an energy-storing device, any simulating circuit must also be able to store energy; the only other device with this capability is the capacitor, so an inductance simulator must of necessity contain some capacitive element.

Since the initial passive design procedure yields a complete filter configuration, active RC filter simulation has concentrated on the realization of an effective inductance simulation circuit. Most of the resulting designs have fallen into the positive- and negative-impedance converter or inverter categories, and have included such circuits as the negative impedance converter and the gyrator. In the last few years, all of these design approaches have been related and re-defined in terms of a "generalized impedance converter" (GIC), whose properties encompass the characteristics of the converter/inverter circuits.

Negative Impedance Converter

An impedance converter is a two-port network whose function is to change an impedance characteristic in some manner. Ideally, when the impedance converter is terminated at one port by an impedance, Z, the input impedance at the other port is directly proportional to Z for all frequencies; the proportionality in a network containing only an amplifier and resistors is determined by a conversion factor, or impedance transformation function, K. If an energy-storing device such as a capacitor is placed in the network, as is often the case, a complex frequency variable, s, appears in the relationship, making the conversion factor K(s). A capacitor serving as a terminating impedance does not introduce the frequency variable, so in this discussion, which deals with the simplest realizations, the (s) notation is not used.

Negative impedance is a characteristic of some circuits and components which causes a decrease in current for an increase in voltage, and vice versa; this property has been recognized for some time, and is exhibited by such devices as tunnel diodes and, under

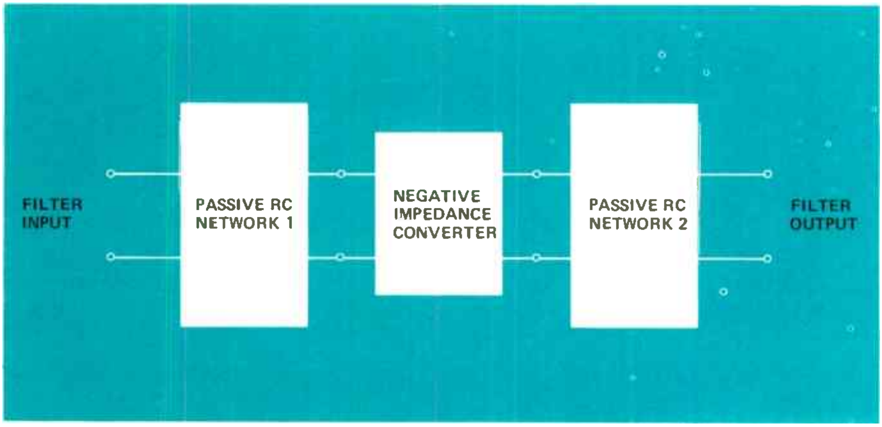


Figure 3. A negative impedance converter is typically placed between two passive RC networks to form a filter section.

certain conditions, vacuum tubes. A negative impedance converter (NIC) is a two-port network which converts an impedance terminating one port into its negative at the other port. For example, if port 2 were terminated by impedance Z_L, the input impedance, Z_{IN}, at port 1 would be related to it by

$$Z_{IN} = -KZ_L$$

where K is the network conversion factor.

A realization of an NIC is typically composed of one operational amplifier (op amp) and associated resistors (see Figure 2). The circuit is placed between two passive RC networks to form a filter section (see Figure 3), so that the negative impedance of network 2 created by the NIC can interact with the positive impedance of network 1. This interaction produces the desired filter response characteristic.

The op amp is generally preferred for active filter circuit realization because of its operating characteristics, its size reduction, the ease with which it can be incorporated into integrated circuits, and its ready availability as an inexpensive component. Theoretically, an op amp has infinite input impedance, zero output impedance, and infinite gain; these characteristics, along with the ability to produce a single output proportional to two inputs (see Figure 4), make the op amp ideal for active filter applications.

Gyrator

Impedance inversion is a property allowing the input impedance of a two-port network to be inversely proportional to its terminating impedance (see Figure 5). As with an impedance converter, the proportion is such that when one port is terminated by Z_L, the input impedance, Z_{IN}, at the other port is related by a conversion factor, K. The relationship in an impedance inverter is expressed as:

$$Z_{IN} = K\frac{1}{Z_L}$$

A gyrator is an impedance inverter in which K is a positive real constant, giving the gyrator the ability to transform a terminating capacitive reactance into an inductive reactance at its

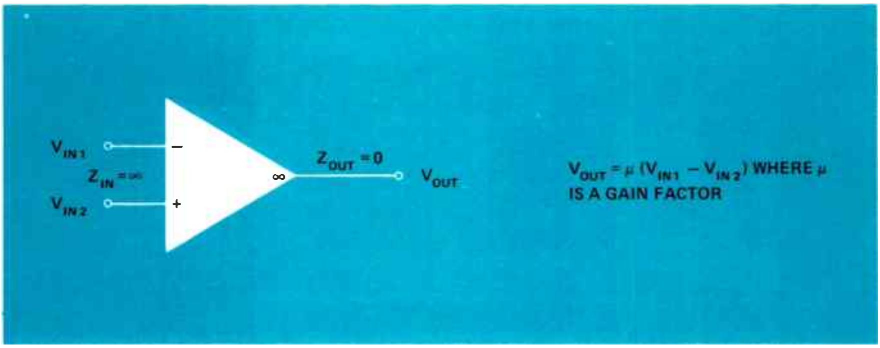


Figure 4. An operational amplifier is ideal for active filter realizations because of its impedance and gain characteristics.

input. Typically, a gyrator is realized as two op amps connected as parallel current-to-voltage converters (see Figure 6).

Generalized Impedance Converter

The properties of the impedance converter — both the positive and negative types — and of the impedance inverter — both the positive type, of which the gyrator is one realization, and the negative type — have been redefined and incorporated into the more universal GIC. This structure is defined as a two-port network whose input impedance is the product of its terminating impedance and some internal network function, which is actually the K factor. According to this definition, if the GIC were to be terminated at port 2 by Z_L, Z_{IN} at port 1 would be equal to KZ_L; if port 1 were terminated by Z_L and port 2 used as the input point, Z_{IN} would be defined by the term Z_L/K. The relationships are the same as those developed for the converter/inverter circuits, but they are here embodied in one network.

The realization of a GIC can be accomplished in several ways, one of which is shown in Figure 7. The various impedance elements can be physically introduced as capacitors or resistors, depending upon the simulation characteristic desired. In Figure 7, for example, making Z2 a capacitor and all of the other elements resistors

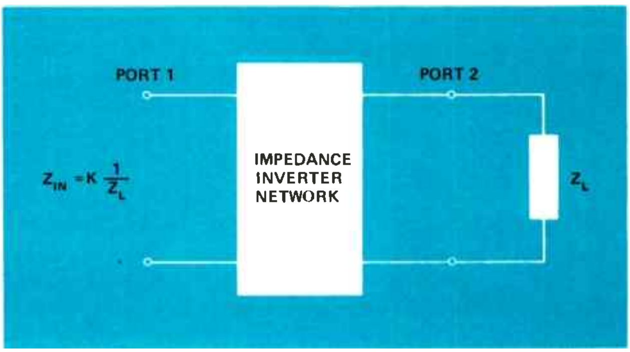


Figure 5. An input impedance inversely proportional to a terminating impedance is generated by an impedance inverter.

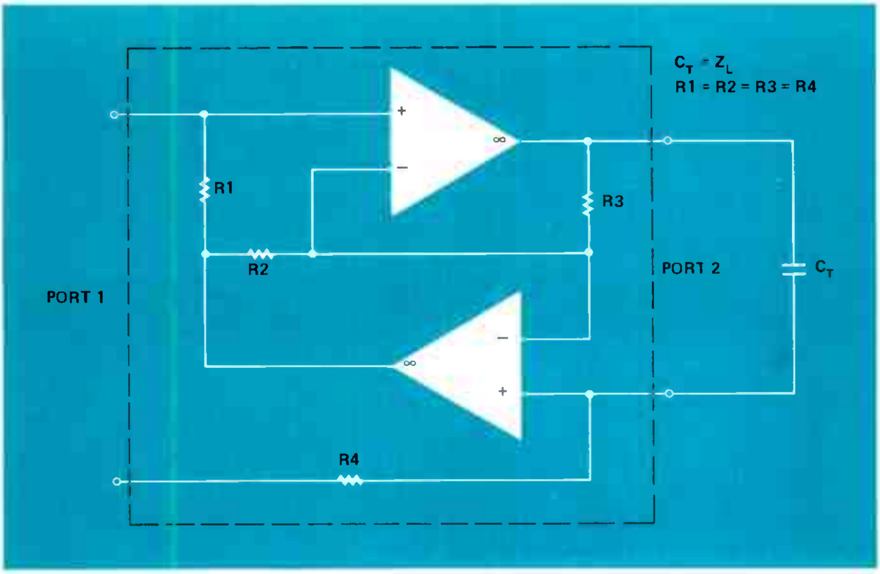


Figure 6. A gyrator can be realized as two current-to-voltage converters connected in parallel to form an impedance inverter network. The capacitive reactance terminating port 2 is made to appear inductive at port 1.

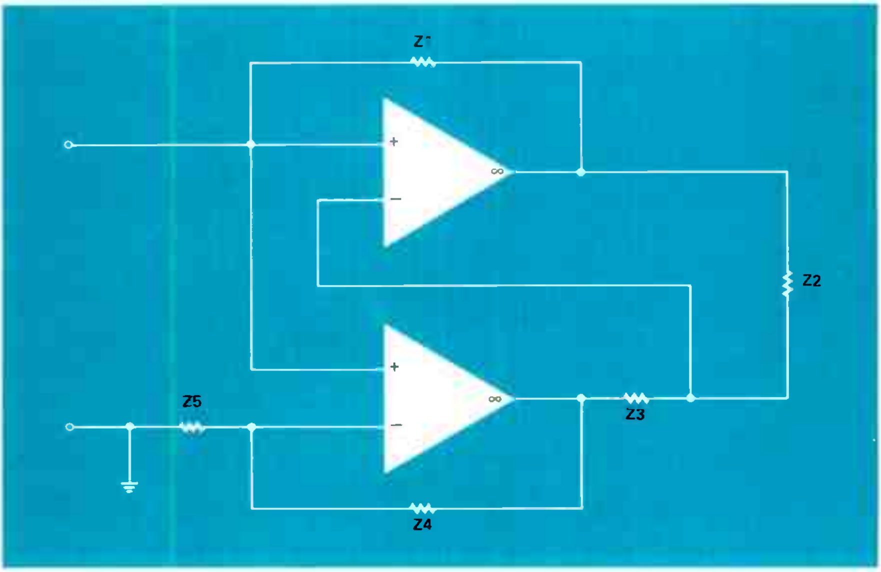


Figure 7. A generalized impedance converter (GIC) is a two-op amp network whose impedances (Z1–Z5) can be realized as resistors or capacitors, depending upon desired response.

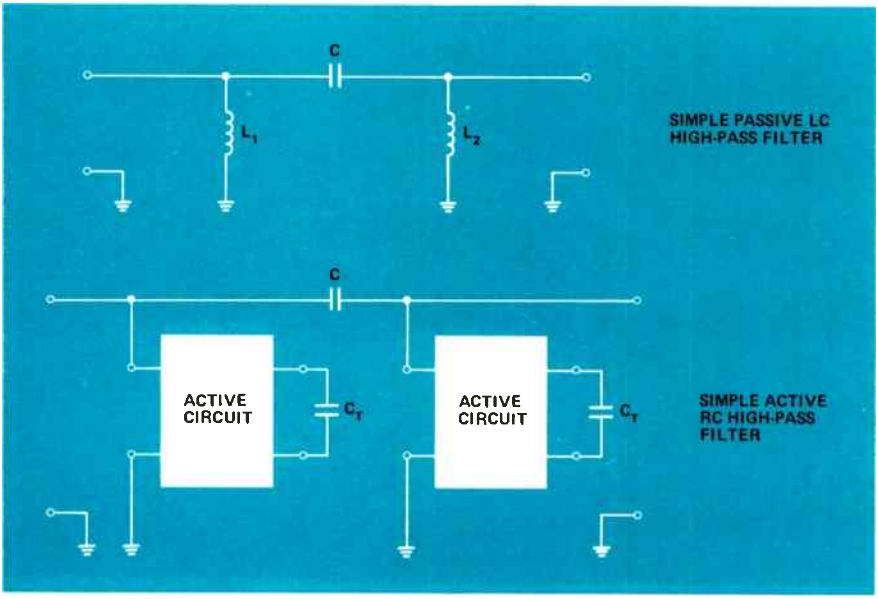


Figure 8. Active circuits commonly appear in filters as two-terminal components. The characteristics of the active circuits, whatever their realized structures, cause their terminating capacitors, C_T, to produce inductor-like behavior.

would result in a configuration equivalent to a gyrator.

Active circuits are most generally incorporated into filters as two-terminal devices (see Figure 8) to introduce inductance characteristics. In some cases, the network is used as a four-terminal device, in which the signal to be filtered is put in at one port and passed out at the other; this technique is shown in Figure 3. Use as a four-terminal device, however, complicates the filter design process because the properties of four, rather than two, terminals must be considered. For this reason, one of the ports is usually terminated and the two terminals at the other port used to connect the network with the other filter elements.

FDNR

A special application of the GIC results in an active circuit called a frequency-dependent negative resistance (FDNR). This is a two-port network which, when terminated at one port by a resistor or capacitor, produces unique impedance characteristics at the other port. The input impedance of an FDNR, or D, network is given by

$$Z_{IN} = -\frac{1}{s^2 D}$$

where s is a complex frequency variable and D is in units of farads squared. From this expression, it can be seen that the impedance characteristic of an FDNR is, indeed, negative and dependent upon the frequency of the signal applied to the simulated filter. Such a circuit can be realized with an appropriately structured GIC, as shown in Figure 9.

The properties of an active filter containing an FDNR are determined

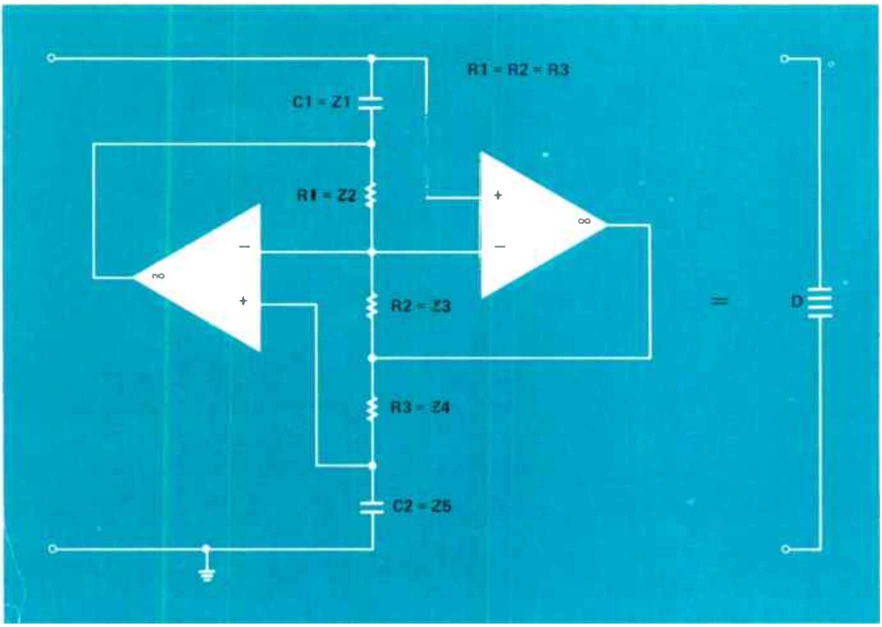


Figure 9. A frequency-dependent negative resistance can be realized as a GIC. The schematic symbol resembles that of a capacitor, but consists of four parallel lines.

through the use of an impedance transformation function, 1/s, in which s is once again the complex frequency variable. Multiplying every impedance in the initially designed LC filter by 1/s allows inductors to be replaced by resistors, resistors to become capacitors, and capacitors to become FDNR's (see Figure 10).

The attempt to create a suitable inductor-simulator for use in active RC filters has produced a variety of circuits, all of which generally depend upon the operational amplifier parameters. An alternate approach to the design of active RC filters is the "biquadratic decomposition" technique, in which the op amp also plays an important role.

Active RC Filter Synthesis

All filters, whether passive or active, can be described in terms of a transfer function, T(s), which is an indication of the efficiency – or inefficiency – of a filter in transferring a quantity at its input to its output; that is, T(s) is a filter transmission characteristic. In passive filters, this function is generally expressed as a ratio of input to output quantity (see April, 1975, *Demodulator*) to allow consideration of attenuation behavior. Active filters, however, contain devices which are essentially amplifiers, so it is more convenient to define their transfer functions as ratios of output to input quantities, or as gain factors. It is also normal practice to consider the quantities to be the input and output voltages, so that T(s) becomes a voltage transfer function.

The problem in active filter synthesis is to arrive at a filter design which, when realized with physical components, produces the transmission char-

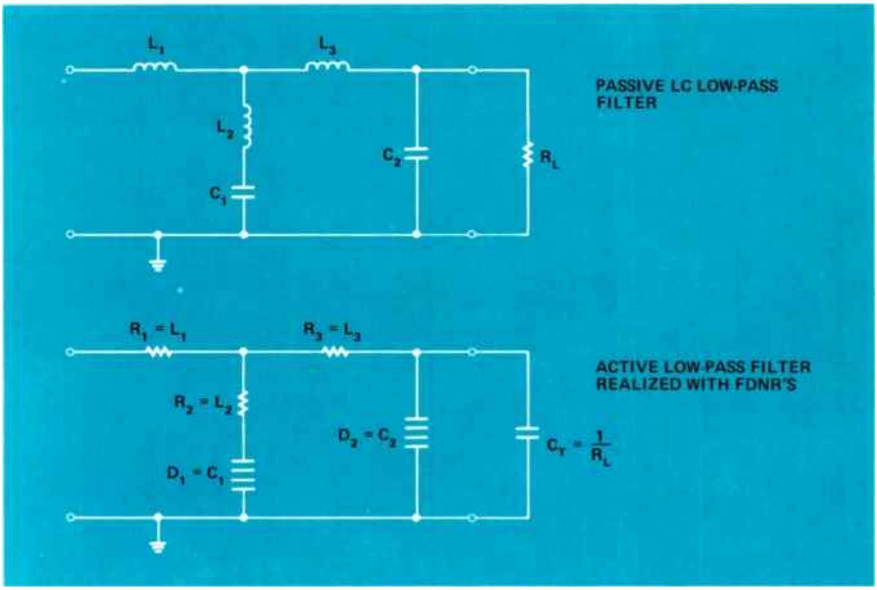


Figure 10. Introducing FDNR's into a filter allows all of the component responses to be simulated by other elements. Simulation techniques such as this result in structures which are very similar in schematic appearance to the passive filters they replace.

acteristic identified by the transfer function.

One approach to this problem requires that T(s) be used to determine an overall filter configuration, just as was done in passive filter synthesis. Another approach, which produces filters whose response characteristics are easier to design, is through realization with biquadratic filter sections.

Biquadratic Filter Realization

As in passive filter synthesis, the transfer function of a given active filter can be expressed as a ratio in the form:

$$T(s) = \frac{N(s)}{D(s)}$$

where N(s) and D(s) are polynomials representing the network output and input quantities, respectively.

Regardless of how complex the polynomial used to identify a quantity, it is possible to break the expression down into second degree, or quadratic, factors of the general mathematical form $P(x) = ax^2 + bx + c$. Performing this operation on both N(s) and D(s) produces a number of quadratic factors in the numerator and a number in the denominator. Each numerator factor is associated with a denominator factor, effectively dividing the filter T(s) into sections, each with its own transfer function t(s), of the form

$$t(s) = \frac{n_2 s^2 + n_1 s + n_0}{d_2 s^2 + d_1 s + d_0}$$

These biquadratic functions — so called because they are defined by two quadratic expressions — are mathemat-

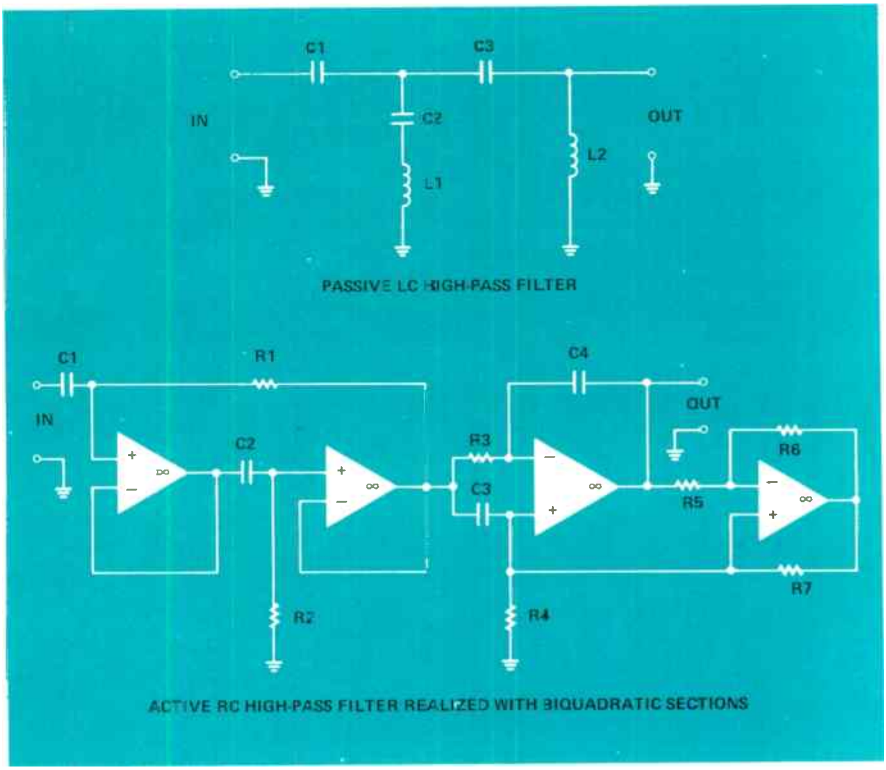


Figure 11. A filter structure can be realized with cascaded biquadratic sections, in which case it may bear very little resemblance to any of the simulation-designed filters.

ically manipulated to produce active filter sections realizing their individual t(s). Connecting the biquadratic sections in cascade – the output of one section becoming the input of the next – produces a complete filter with the desired transmission characteristics.

The physical realization of a biquadratic filter section varies according to a given application's requirements. When these needs have been considered, a decision is reached as to which of the many possible circuits would be most effective. A common biquadratic realization contains a minimum of one and a maximum of four operational amplifiers. The output of the section is taken at the output terminal of one of the op amps, taking advantage of the low impedance at that point to reduce the interaction between cascaded sections (see Figure 11).

When the type of circuit has been selected, it must be determined if the requirements can be met solely by component selection, or if the circuit has to be tuned.

Tuning

Since the amplifiers used in a circuit realization are not perfect, the actual transfer functions produced are likewise imperfect. The best method for dealing with this imperfection is by tuning the individual biquadratic sections; in this way, resistor and capacitor tolerances can also be compensated for.

Essentially, the tuning procedure consists of making resistance, amplitude and phase measurements on a filter structure, comparing these to the desired quantities, and changing resistance values to make the two values as close as possible. This process can be automated in integrated circuit fabrication by interconnecting a computer with measuring gear and a laser trimmer.

At GTE Lenkurt, the biquadratic filter realization has been used to create the filters used in the 262A Data Set, which processes data at 4800 b/s. At this bit rate, filtering must be extremely precise to avoid pulse distortion; the active biquadratic filter, specifically tuned to match the characteristics of each data set, provides this precision.

Active filters are essential elements in modern electronics applications because of their great savings in size and weight, and their compatibility with integrated circuit techniques. Despite some inherent sensitivity problems, active filters are also valuable because they can not only realize all of the network functions of passive filters, but can also realize transfer characteristics totally unattainable with strictly passive networks.

BIBLIOGRAPHY

1. Bruton, L.T. and D. Treleaven. "Active Filter Design Using Generalized Impedance Converters," *EDN*, Vol. 18, No. 3 (February 5, 1973), 68 – 75.

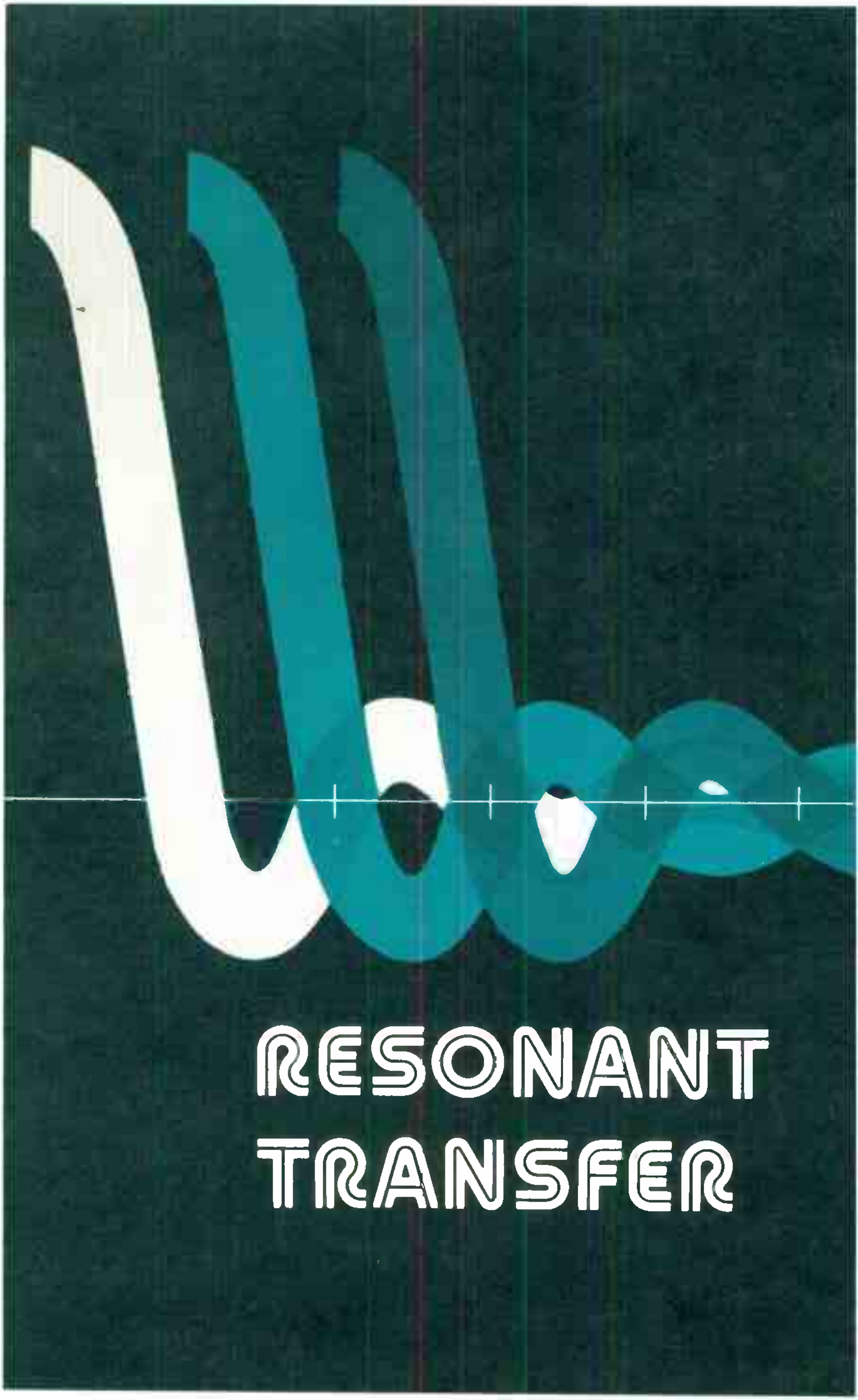
2. Hilberman, D., *et. al.* "A Review of Active Filter Terminology, Technology and Design," paper in preparation for IEEE Circuits and Systems Society. Draft: March 1975.

3. Huelsman, L.P., ed. *Active Filters: Lumped, Distributed, Integrated, Digital, and Parametric*. New York, N.Y.: McGraw-Hill Book Co., 1970.

4. Mitra, S.K. *Analysis and Synthesis of Linear Active Networks*. New York, N.Y.: John Wiley & Sons, Inc., 1969.

5. Schmidt, C. "Active Filters (Technical Memorandum No. 62/73/476)". San Carlos, California: GTE Lenkurt, Inc., 1973.

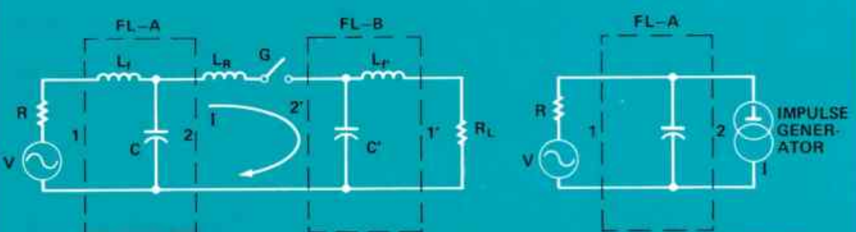
6. Temes, G.C. and S.K. Mitra. *Modern Filter Theory and Design*. New York, N.Y.: John Wiley & Sons, Inc., 1973.


RESONANT
TRANSFER

Resonant transfer is a novel technique for transferring energy between filters, with minimum energy loss. Use of this technique allows the design of zero-loss amplitude modulators, time-division switching systems, electronic 2/4-wire hybrids which do not require balance networks, simple data transmission filters, and unique FDM carrier systems.

The basic arrangement for the resonant transfer of energy between filters is shown in Figure 1. Generally, when the concept of resonant transfer is applied, it is presumed that energy from a source V is to be applied to an LC filter, FL-A, and is subsequently to be transferred without loss to a second filter, FL-B. The physical arrangement of the filters is such that shunt capacitances appear at the switched ports 2 and 2′, respectively. The two filters are interconnected by a very small inductor L_R, which is in series with an analog gate G. It is convenient for the purpose of explanation to visualize the existence of a series inductor (L_f and $L_{f'}$) within each filter. This inductor is not required for circuit operation but frequently is present as an overall characteristic of the filter and makes the discussion of resonant transfer easier to understand.

Operation of the circuit takes place during two distinct time intervals. One interval, T_s, is comparatively long, on the order of 125 microseconds; this is the time during which the analog gate G is open and charge accumulates on capacitor C due to the energy source V. The second time interval, τ, is



A. FILTER ARRANGEMENT FOR RESONANT TRANSFER OF ENERGY BETWEEN FILTERS FL–A AND FL–B.

B. EQUIVALENT CIRCUIT FOR FILTER FL–A.

Figure 1. Circuit arrangement for resonant transfer operation.

approximately one microsecond. This is the period during which the analog gate G is closed. The inductor L_R is selected to resonate with capacitors C and C′ so that exactly one-half sine wave of current flow occurs during the one-microsecond closure of gate G. The series inductors assumed to be in the filters appear as open-circuits to the abrupt change in current produced during τ. Thus, during τ, the circuit effectively consists only of the series resonant circuit made up of C, C′ , L_R, and G. Together, these elements can be considered the resonant transfer mechanism.

During the gate closure interval τ, the charge which existed on capacitor C is transferred to capacitor C′, thus reducing the charge on C to zero by the end of the closure interval. During the following T_s interval, while capacitor C is accumulating a new charge, capacitor C′ is being discharged through to the load resistor R. With proper filter design, the charge on C′ should be reduced to zero by the end of the T_s interval, when gate G closes.

Time-Domain Representation of Resonant Transfer

The timing sequence for closing analog gate G is shown in Figure 2A. As previously described, the gate is closed for τ microseconds out of each T_s microsecond interval. The signal

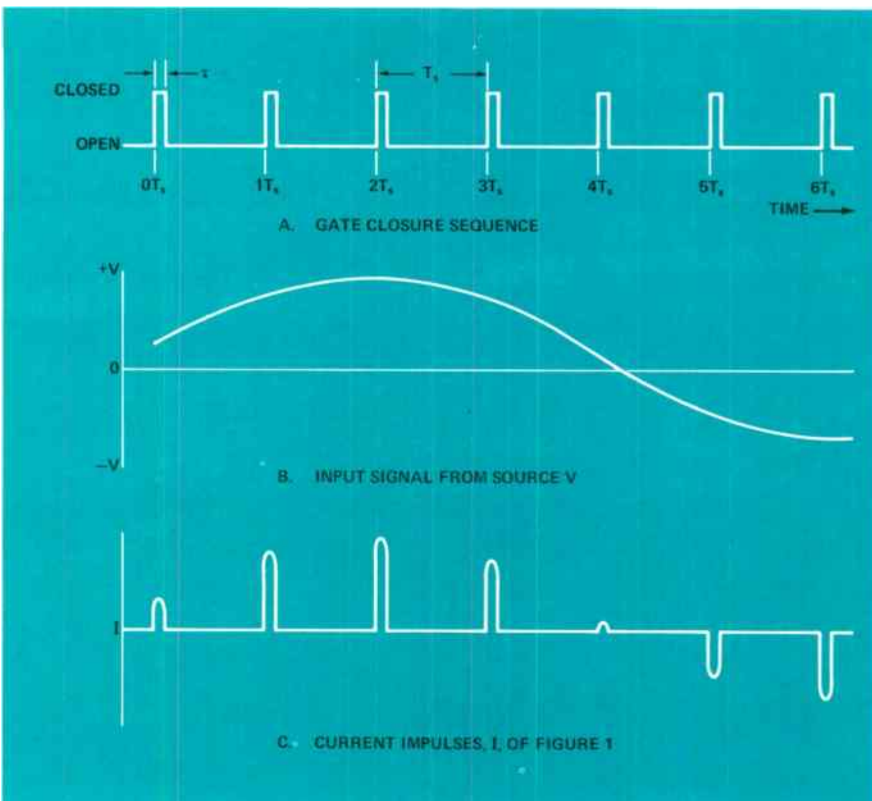


Figure 2. Timing for resonant transfer.

applied to the circuit by the source V is indicated in Figure 2B. At each switch closure interval the charge on capacitor C and, therefore, the magnitude of the current impulse (I) will be proportional to the signal amplitude. The resulting sequence of current impulses is shown in Figure 2C.

Because of the current impulse sequence appearing at the gated port (2′) of the filter, an equivalent representation of the resonant transfer process as applied to a single filter can be conceived. This is shown in Figure 1B. Here the source at port 2 is not a conventional generator; it is, rather, a special current source which produces the sequence of impulses shown in Figure 2C. This sequence of current impulses, in conjunction with the filters, has certain characteristics which are of interest.

Filter Impulse Response

As each current impulse is applied to the filter, it produces a voltage at the gated port which decays in an oscillatory manner. The general shape of the voltage waveform is that of a sinX/X pulse. While in actual operation the consecutive impulses at the gated port would be continuous, the voltage produced by three of these impulses would appear as shown in Figure 3. If the filter frequency response is designed properly, the response to each impulse will be such that the voltage wave will pass through zero at multiples of T_s subsequent to the impulse. If this condition holds, and the various multiples of T_s are examined, it will be noted that at any instant, $t = kTs$, only one impulse contributes to the magnitude of the voltage — the waveforms due to all other impulses are zero. In mathematical terms, the waveforms are said to be orthogonal at $t = kTs$. This type of resonant transfer filter operation might be termed "impulse response zero" operation.

One filter that meets the "impulse response zero" criterion which requires that the voltage response to a

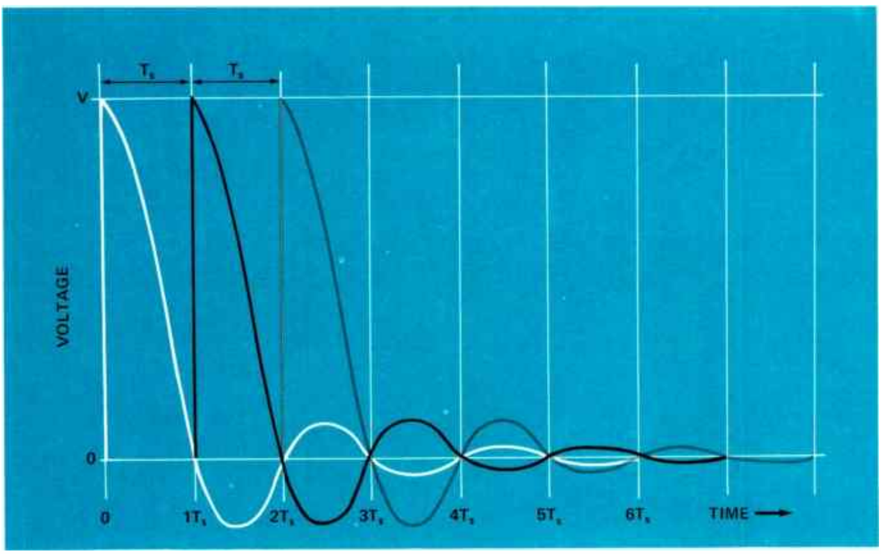


Figure 3. Voltage at gated port of filter due to three current impulses.

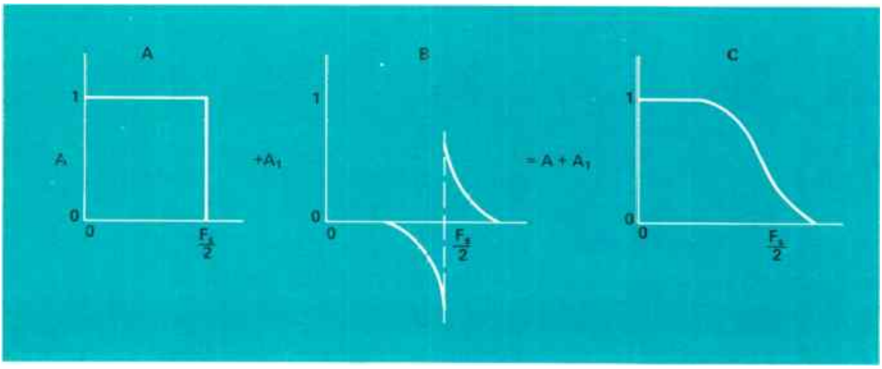


Figure 4. A characteristic having skew symmetry about $F_s/2$ as shown in (B) can be added to an ideal filter (A) characteristic to obtain an equal area curve having no abrupt transitions (C).

current impulse pass through zero at multiples of T_s is the "ideal" filter whose characteristic curve is shown in Figure 4A. This filter has a passband extending from 0 to $F_s/2$ Hz, where its stopband abruptly begins.

Unfortunately, because of the abrupt transition required from passband to stopband, such an ideal filter is physically unrealizable. However, a theorem by Nyquist, based on mathematical construets, states that if a transmittance having the characteristic of skew-symmetry (as shown in Figure 4B) is added to the transmittance of an ideal filter (Figure 4A), the resulting filter transmittance will have the same zero crossings, in the time domain, as the ideal filter. A suitable choice for the skew-symmetric shape leads to a final filter response having the gradual passband-stopband transition (in the frequency domain) shown in Figure 4C. One filter characteristic shape which is arrived at in this manner is the "raised-cosine" filter shown in Figure 5. The frequency response and the impulse response of the raised-cosine filter of Figure 6 are shown in Figures 7 and 8, respectively, where it can be seen that this filter is readily realizable.

It is reasonable to assume that if all the energy is removed from the filter

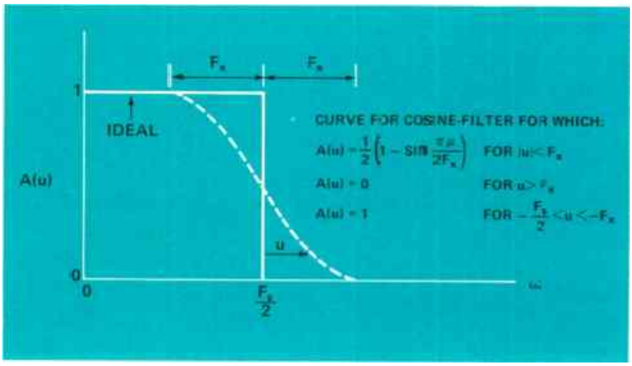


Figure 5. Comparison of characteristic curves of ideal low-pass filter and raised-cosine filter.

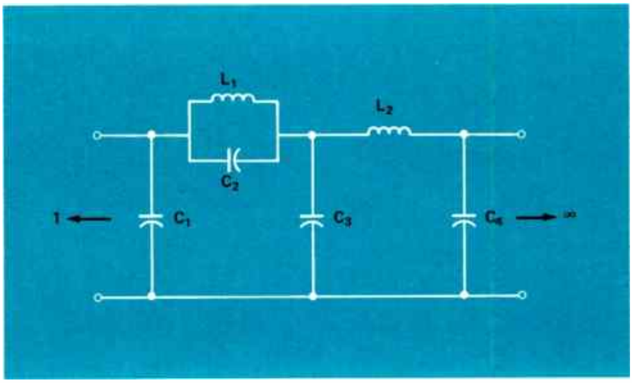


Figure 6. Schematic of a raised-cosine filter.

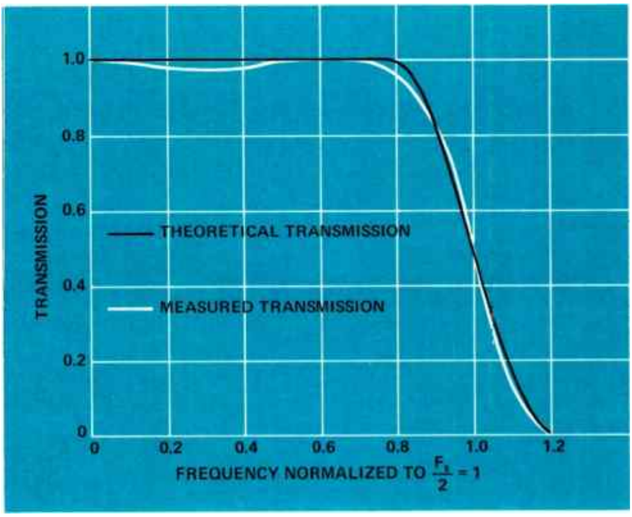


Figure 7. The frequency response of a raised-cosine filter with 25 dB attenuation at $F = 1.2F_s/2$.

at the time of an impulse, and that if none of this energy can be returned to the filter at subsequent switch closure intervals due to the zero-crossing criteria (subsequent switch closures will have zero voltage), then the resonant transfer technique is an inherently low-loss process. A loss on the order of 1.5 dB through a pair of filters connected as shown in Figure 1 is readily attainable.

Frequency Domain Response

In Figure 2C, the pulse sequence at the switched port of the filters is amplitude modulated by the input source V. This is presented pictorially in Figure 9, where A represents the frequency spectrum of the source and B shows the spectrum at the gated port. The input spectrum is repeated at the gated port as upper and lower sidebands spaced about multiples of the sampling frequency F_s. If the input frequency is designated as F_V, in Hz, the spectrum at the gated port is, then, $kF_s \pm F_V$ Hz, where k = 0,1,2,3, etc., represents the multiples of F_s. The point of interest is that filter FL-A of Figure 1 can be a

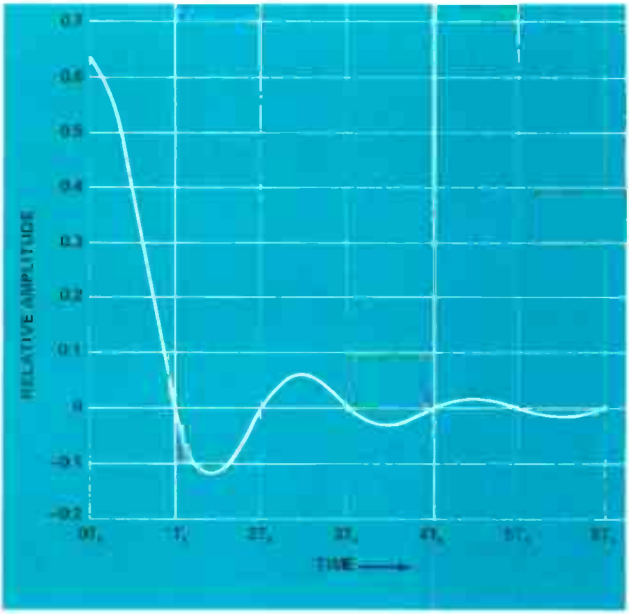


Figure 8. Impulse response of a raised-cosine filter.

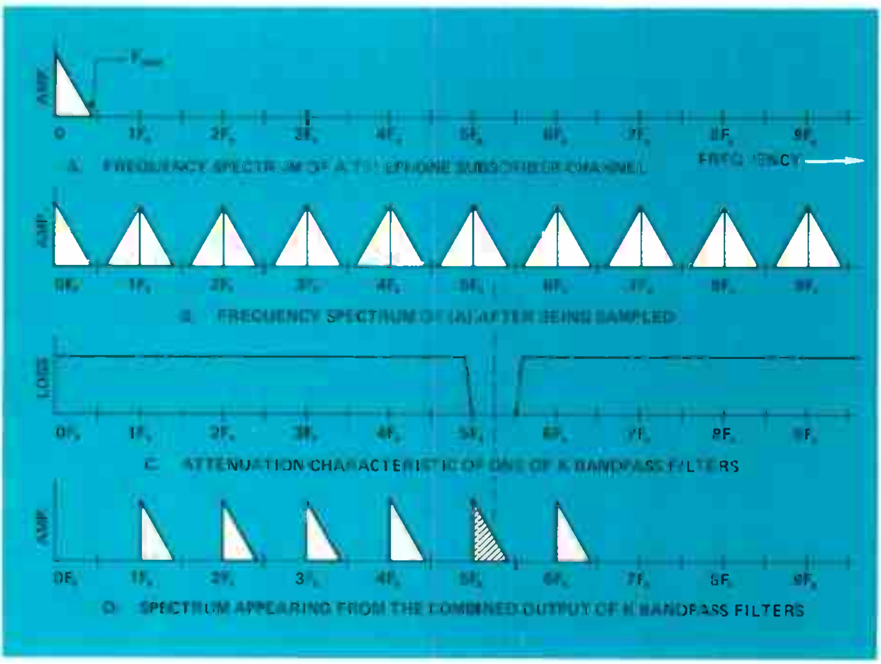


Figure 9. If lowpass spectra are sampled at a rate F_s, where $F_s > 2F_{max}$ and where F_{max} is the highest frequency of interest in the lowpass spectra, the lowpass spectra will be reproduced as sidebands about multiples of F_s.

lowpass filter while FL-B, as shown in Figure 9C can be a bandpass filter. The passband frequency of FL-B can be designed to pass a single sideband, as shown. This arrangement results in a zero-loss single sideband modulator.

If this process is repeated for several modulators, each having different passband frequency allocations, the several modulator outputs can be combined to form the mutlichannel, frequency-division multiplex line signal shown in Figure 9D. Two very significant differences appear in this modulation process compared with conventional single sideband modulators. Here, the modulator has a theoretical loss of 0 dB, and about 1.5 dB actual loss. A conventional modulator has a theoretical loss on the order of 3 dB and an actual loss on the order of 4.5 dB. Furthermore, a conventional modulator requires a carrier input frequency kF_s to generate each pair of sidebands ($kF_s \pm F_v$). Thus, six separate carrier frequencies would be required for conventional modulators, one for each modulator, to derive the spectrum of Figure 9D. With the resonant transfer technique, it is only necessary to apply the single pulse train shown in Figure 2A to all six modulators. This allows a substantial savings in system cost.

The Resonant Transfer 2/4-Wire Hybrid

The circuit of Figure 1 is, in addition to being a theoretically zero-loss configuration, bidirectional. That is, the source could just as well be placed at port 1′ and the load at port 1. In spite of the zero-loss characteristics of this circuit it may be used in applications where a high loss exists between an input or output port and a termination. It is then necessary to add gain while retaining the bidirectional transmission path. Since conventional amplifiers are unidirectional, this situation presents a problem. One possible solution is the use of "negative impedance" amplifiers, which are bidirectional but tend to oscillate unless the gain is quite low. The other possibility is to rearrange the circuit of Figure 1 to form a configuration which behaves just as a conventional 2/4-wire hybrid.

The resonant transfer 2/4-wire hybrid arrangement is shown in Figure 10. The filters may be all lowpass or a mixture of lowpass and bandpass as shown. In the operation of the circuit, a signal applied to the two-wire port charges a capacitor on the switched side of the lowpass filter. When gate G1 closes, the energy stored in this capacitor is discharged into the transmit bandpass filter using the resonant transfer process. Immediately after gate G1 opens, while the energy stored in the lowpass filter is still zero, gate G2 is closed. This causes transfer of energy from the receive bandpass filter to the lowpass filter. The interval which follows – from the opening of G2 until the closing of G1 – is very nearly equal to T_s. This choice of interval allows the energy applied to the lowpass filter from the receive bandpass filter to be transferred through the lowpass filter to the 2-wire termination. Thus, only the energy from the source at the 2-wire port is present at the switched side of the lowpass filter when gate G1 closes.

Generally, performance of the resonant transfer hybrid is very similar to conventional transformer hybrids. However, it offers several distinct advantages. The original objective in considering a hybrid was to be able to insert gain into the system. This is done as indicated in Figure 11, where a net gain in the West-East and in the East-West direction is desired. Insertion of the amplifiers as shown provides the required gain. Of course, it is

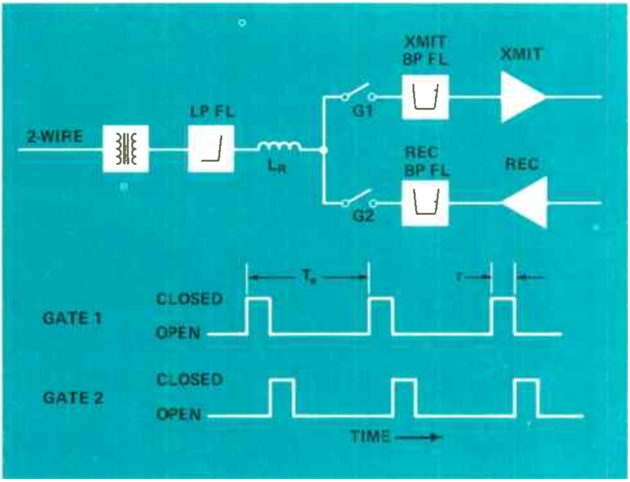


Figure 10. Resonan transfer 2/4 wir hybrid.

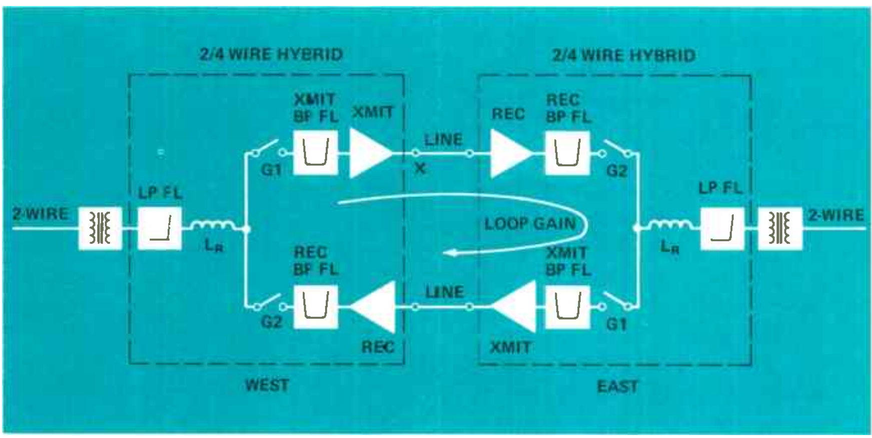


Figure 11. The use of 2/4 wire hybrids to allow insertion of amplifiers in transmission path.

necessary that the gain around the loop from any given point X back to X be less than 0 dB, or the circuit will oscillate; such oscillation is called "singing." It is the purpose of the hybrids to insert sufficient loss (called transhybrid loss) in the loop to allow this condition to be met and yet to insert very little loss in the direct transmission paths from West to East and East to West.

The resonant transfer hybrid ha about 1.5 dB loss in the direct trans mission paths; that is, from the hybri 2-wire port to its transmit port an from receive port to two-wire port The transhybrid loss — the loss fron the receive port to the transmit port is about 35-40 dB. By comparison, conventional hybrid has a loss in th direct transmission path somewhere o the order of 4.5 dB and a realizabl

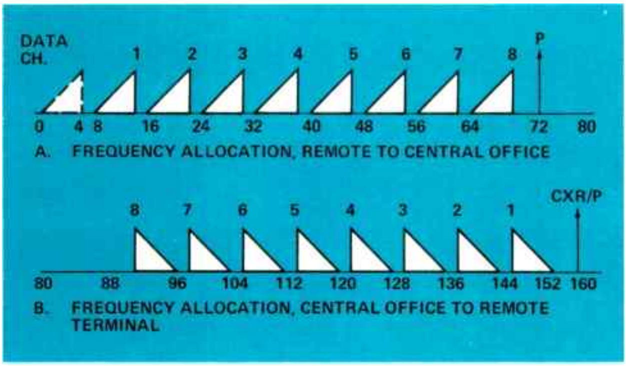


Figure 12. Line signals between central office and remote terminal.

transhybrid loss in the 35-40 dB range.

A conventional hybrid is a lattice or bridge type network which depends on balancing the bridge arms to achieve transhybrid loss. One arm of the bridge structure is a "balancing" network which must have an impedance characteristic that matches the impedance seen looking from the two-wire port into the two-wire line. This matching network is not required in the resonant transfer hybrid.

Advantages of the resonant transfer hybrid are: (a) it achieves the same transhybrid loss as a conventional hybrid while having 3 dB less loss in the transmission path, (b) it requires no balancing network, and (c) by using all lowpass filters it can be made to behave as a conventional vf-to-vf (voice frequency) hybrid, but by utilizing bandpass transmit and receive filters, it becomes both a hybrid and a modulator as well.

Resonant Transfer Frequency-Division Multiplex System

Using eight of the 2/4-wire hybrids with bandpass filters spaced at 8 kHz intervals in the frequency range from 8 to 64 kHz, the spectrum shown in Figure 12A can be formed. This basic 8-channel group allocation is used in the 48 channel multiplex system shown in Figure 13 for transmission in

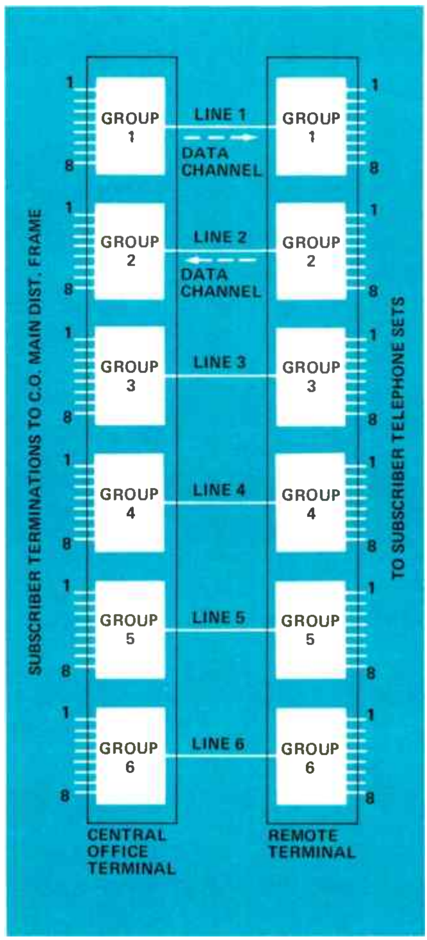


Figure 13. Arrangement of resonant transfer subscriber multiplex system.

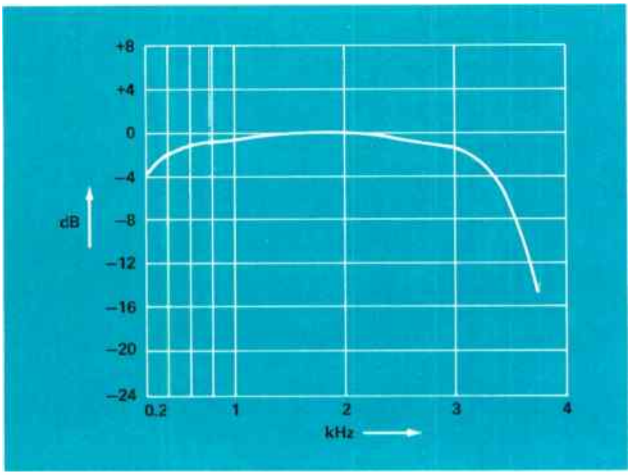


Figure 14. The channel response for a 64-68 kHz channel.

the remote terminal to central office terminal direction. The high frequency line signal shown in Figure 12B is used for transmission in the other direction; it is formed by modulating the spectrum of Figure 12A with 160 kHz and using the lower sideband. Both signals in Figure 12 are transmitted, in opposite directions, over a single physical cable pair between the two carrier terminals.

The system can be implemented in 8-channel increments to its ultimate 48-channel capacity, with one cable pair being required for each 8-channel group. Typical frequency response for the 64 – 68 kHz channel is shown in Figure 14.

The resonant transfer technique has possibilities in many aspects of communications, and, while its full potential has perhaps not as yet been realized, it holds the promise of affecting future trends in telecommunications.

BIBLIOGRAPHY

1. Bennett, W.R. and J. Davey. *Data Transmission.* New York: McGraw-Hill, 1965.
2. Getgen, L.E. "A 48-Channel Subscriber Multiplex System Using Resonant Transfer Modulation," *International Conference on Communications,* (June 1975).
3. Getgen, L.E. *The Use of Resonant Transfer for Combined Switching and Multiplexing.* Doctoral Thesis. Oregon State University, 1972.
4. "Hybrids," *The Lenkurt Demodulator,* January 1964.

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