JOURNAL OF

The British Institution of Radio Engineers

(FOUNDED IN 1925 - INCORPORATED IN 1932)

"To promote the advancement of radio, electronics and kindred subjects by the exchange of information in these branches of engineering."

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31st YEAR OF PUBLICATION

Members will share our pleasure in noting that during this year we celebrate the 31st anniversary of the Institution's *Journal.** Publication has not, of course, been regular throughout the whole of that period; indeed, the original *Journal*, published during the early years of the Institution's history, was issued at irregular intervals. Nevertheless, those issues contain much useful material, particularly on the early applications of radio science.

It was not until 1939 that the Institution was able to embark on regular publication of a *Journal*, but soon afterwards war-time conditions affected both printing of the *Journal* and the discussion of engineering topics. Despite those difficulties, quarterly publication was maintained until 1945, when the present format was adopted.

Regular monthly publication of the *Journal* has been continuous since January 1949 except for the omission of one issue in 1950, which was occasioned by a strike in the printing industry. Since 1949, the yearly volume has increased in size from 464 to 650 pages. The progress of the Institution has therefore been reflected in the growth of the *Journal*, not only in size but also in its acknowledged international reputation. Today the *Journal* is sent to 75 countries throughout the world, and its authority as a source of original material on all aspects of radio and electronic engineering may be gauged from the frequent references to its papers which appear in other journals.

The range of subjects discussed in the *Journal* also reflects the diverse industries in which radio and electronic engineering techniques are employed. Reference has been made in previous editorials to the ever-increasing use of electronic techniques in industry. An example is the application of automation, which is currently receiving much attention: it is interesting to recall that one of the first papers on the subject was published in the Institution's *Journal* in 1947,† while other aspects were discussed at the Institution's 1954 Convention on Industrial Electronics. The vast expansion of nuclear energy projects also calls for highly specialized electronic techniques, which were the subject of Brit.I.R.E. Conventions in 1951 and 1954. Important developments are also taking place in the fields of communications, u.h.f. propagation, and colour television, and papers have been regularly published on all those subjects. The achievement of reliability is, again, a vital problem both in communications and in industrial applications, and one which has been discussed at Institution meetings and in the *Journal* on a number of occasions.

One of the Institution's objects is to keep all professional radio and electronics engineers informed of the latest advances in these various fields, and all members, by their association with the Institution, acknowledge their obligation to contribute to the furthering of that aim. The Programme and Papers Committee constantly endeavours to promote the publication of original papers and surveys, and it is to those members engaged in research, design and development that the Committee looks for constant support in maintaining and still further improving the value of the *Journal*.

There is a sense of achievement in reviewing the past and anticipating the future. Although juvenile in comparison with the well-known weekly periodical "The Engineer," which celebrated its centenary on 4th January last, the *Brit.I.R.E. Journal* has a fellow feeling in offering that journal its congratulations.

* The first issue of the *Journal* was published in October, 1926.

† J. A. Sargrove, "New methods of radio production," *J.Brit.I.R.E.*, 7, pp. 2-33, Jan. 1947.

NOTICES

New Year Honours List

The Council of the Institution congratulates Group Capt. Gordon Thripp (Member), on his appointment as an Officer of the Most Excellent Order of the British Empire. Gp. Capt. Thripp, who is the Deputy Principal of the Indian Air Force Technical Training College at Jalahalli, was Chairman of the Bangalore Section of the Institution from its inauguration until 1954.

A full account of Group Captain Thripp's career was published in the *Journal* for March 1953.

Students' Essay Competition

As announced in the December *Journal*, the General Council is to award prizes of 10 guineas and 5 guineas for essays by Students on the subject of "Problems of Technical Training." The approach should be primarily from the viewpoint of the trainee, and should cover training in both theory and practice.

The Competition is open to all registered Students of the Institution, and to Graduates who are under 23 years of age on the closing date for entries, which will be 1st April. Essays should not exceed 3,000 words, and should preferably be typewritten, on one side of the paper only. Council reserves the right to publish outstanding contributions in the *Journal*.

New Committee of the Wellington Sub-Section

The new Committee of the Wellington Sub-section of the New Zealand Section, for the session 1955-56, has recently been elected. The Chairman is J. W. N. Simpson, M.Sc., B.E.(Hons.) (*Member*), and A. Ryland (*Associate*) is Honorary Secretary. Other members of the Committee are:—B. A. Bernon (*Associate Member*); W. A. Penton (*Associate Member*); W. C. Lee, B.Sc. (*Associate Member*); W. H. Heald, B.Sc. (*Graduate*) and R. E. Greene (*Associate*).

South East Essex Technical College

The extensions to the Science Department of the South East Essex Technical College, Dagenham, were officially opened on 19th November by Geoffrey Marchand Esq., C.B.E., M.A., J.P., Chairman of the London Regional Advisory Council for Higher Technological Education. The Institution was invited to send a representative, and a member of the Education and Examinations Committee, Mr. D. A. Crowther, B.Sc.(Eng.) (*Associate Member*), attended on the Institution's behalf.

In his speech, Mr. Marchand stressed the need for technical education. With University degree standards rising, it was becoming increasingly difficult for students to take degrees by part-time study. He felt that the future role of technical colleges would be to provide "Sandwich" courses leading to alternative qualifications. This required the co-operation of industry.

The Department gives courses for B.Sc. degrees, and City and Guilds of London Institute courses in Telecommunications and Servicing. In conjunction with the Engineering Department, Electronics and Radio subjects are taught for the Higher National Certificate in Electrical Engineering.

Special Courses in Higher Technology

The Regional Advisory Council for Higher Technological Education in London and the Home Counties has recently published details of the courses commencing in the Spring and Summer terms. The Bulletin contains details of twenty-eight part-time courses of postgraduate or post-H.N.C. standard of direct interest to radio and electronics engineers. The subjects are, in general, fairly specialized, and are covered by courses of from six to twelve lectures. There are also full-time courses in nuclear techniques and measurements, work study, industrial relations, production planning and control.

Copies of the bulletin may be obtained from The Secretary, Regional Advisory Council, Tavistock House South, Tavistock Square, London, W.C.1, price 1s. 6d.

Conference on Automation

The North London Productivity Committee of the British Productivity Council is to hold a Conference on "Automation—The Present and The Future," on Thursday, 8th March, at Enfield Technical College, Queensway, Ponders End, Enfield. The speakers at the Conference will be announced later.

Members of the Institution may obtain further particulars from the Hon. Secretary, 126, The Ridgeway, Enfield.

DESIGN OF STUDIOS FOR SMALL BROADCASTING STATIONS*

by

Ronald F. Goodsman (Associate Member)†

SUMMARY

Practical considerations in the siting, design and construction of studio buildings are given and methods of making studios and associated control rooms acoustically suitable are discussed. Reference is made to the studio building of the Trinidad Broadcasting Company, which incorporates one large, two medium, and two small studios. The equipment and circuit facilities provided in these studios and in the central control room are described.

1. Introduction

The object of this paper is to bring under one heading some of the basic information which the practising engineer needs when embarking on the design and equipment of broadcasting studios. In very large organizations, the design engineer has the services of several specialists at his disposal. In small organizations—especially those overseas—it often happens that the engineer operating a broadcasting station also has to be responsible for much of the detailed work during the planning and building of studios. The literature dealing with the many aspects of the subject which he has to cover is scattered, and under the circumstances it is very easy for some important point of design to be forgotten. It is hoped that the paper will be of value to such engineers. Some notes on the equipment designed for the Trinidad Broadcasting Company are appended.

2. Preliminary Considerations.

In a large broadcasting organization, it usually is possible to set aside studios for each purpose, and to provide separate studios for talks, drama, music, etc. Each kind of studio can therefore be designed with a single end in view.

In smaller organizations such separation of function is not possible owing to the limited number of studios which can be made available, and the very large number of uses to which each must be put. It is most important therefore, to make the studio arrangement as flexible as possible. This will entail compromise, and the first step when planning studios should be to decide how much of this is necessary.

It is better to be able to fulfil all the functions required of a station with some degree of compromise, than to fulfil some very well, and the others very badly. For example, it would be bad planning to provide one large studio which is excellent for musical ensembles, if the station was also required to handle a number of talks, and disk programmes. Whilst it is true that, with suitable treatment, a large studio can be used for almost any purpose, it is also true that for the same cost, two smaller studios could be built.

The latter arrangement would enable the station to deal with the bulk of its programmes without any great compromise, and in addition, would provide an additional studio for rehearsal purposes. The advantage of this extra facility cannot be over-emphasized.

2.1. *Studio Location*

The location of a studio building has a large effect on its cost. Studios in heavy traffic or industrial areas need a very expensive form of construction in order to insulate them from the ambient noise. Studios located in quieter surroundings do not necessarily need such heavy insulation.

* Manuscript received 21st September, 1955. (Paper No. 338.)

† Trinidad Broadcasting Co. Ltd., and Rediffusion (Trinidad) Ltd., Port of Spain, Trinidad.

U.D.C. No. 534.861.1:621.396.712.3.

Land on the outskirts of a town is usually cheaper than in the town itself, and larger sites are often available. Apart from the cost aspect, size is often an important consideration for broadcasting concerns who wish to have room for future development.

Studios in towns often have to be multi-storied in order to obtain enough floor area on a restricted ground area. This kind of construction has to be very elaborate in order to be successful owing to the difficulty of preventing sound transmission through the floor-ceiling partitions. It is therefore relatively expensive.

However, sometimes important advantages can be gained from locating a studio building in a town area. The question of location should be carefully considered at the outset.

2.2. *Building Layout*

Layout is the next matter requiring consideration. Several factors will affect this. Studios of varying floor area require ceilings of different heights. From a building point of view, however, it is desirable to change the roof level as seldom as possible. This being the case, it is sometimes necessary to change the floor level in single-storied buildings in order to obtain the required ceiling heights. At the same time, frequent changes of floor level can be most irritating to occupants, and it is desirable to reduce them to the minimum. This can be done by grouping all studios of approximately the same height together.

Studios primarily used for orchestras and bands should be placed as far away as possible from studios normally operating at a lower sound level. This reduces the amount of sound insulation which has to be incorporated into the building structure in order to isolate acoustically the two groups of studio. Studios used for musical purposes are often used for audiences. The separation of these studios from the smaller ones which do not deal with audiences also helps in the control of visitors to the studio building.

Corridor space needs careful consideration so that space shall not be wasted. It can be used to separate studios operating at widely different sound levels as described above, as well as providing easy access to studios. It should also be arranged so that audiences can enter and leave those studios intended for them by passing the minimum number of other studios. In noisy locations, corridor space can sometimes be used to good effect if placed on the periphery of the studio space where it provides sound insulation against the ambient noise.

Balance and Control cubicles need to be arranged so that operators may have the least restricted view into the studio when seated in their operating position. Unless careful attention is given to this detail, the operator is liable to find himself having to look in one direction to see his controls, and another to see into the studio. In large studios, where several people may be engaged at one time, it is good practice to place the cubicle floor level well above that of the studio so that the operator can see over the heads of people standing up.

Figure 1(a) shows an outline of the Trinidad Broadcasting Company's studio building in which the above points have been borne in mind as far as practicable. The shaded area has its floor level raised 2 ft. above ground level, and the ceiling height in this area is 10 ft. All other studios are built at ground level so that all except the largest have a ceiling height of 12 ft. without any change in the roof level. The large studio requires a height of 20 ft. and this has

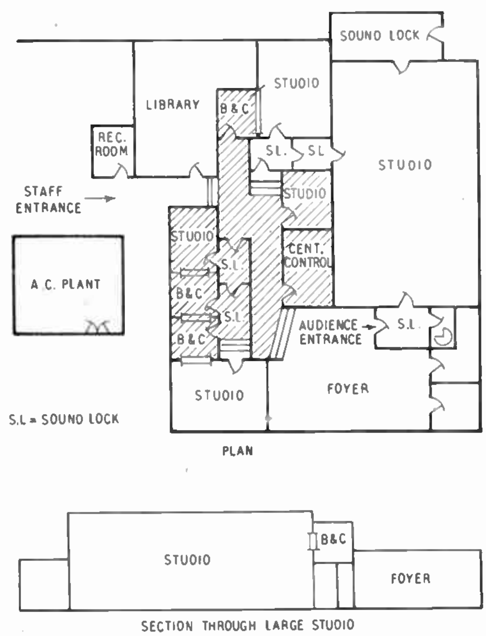


Fig. 1(a)—Arrangement of Trinidad studios. Shaded area is 2 ft. above ground level.

a separate roof. As will be seen by the vertical section through this studio, the balance and control cubicle has its own floor level which gives the operator a perfect view into the studio.

With only one exception, sound locks are provided for each studio door (see Section 4.1). Audiences can enter and leave the large studio without going near any other studio. A separate entrance is provided for staff. The sound lock for the large studio rear door is enlarged so that it can also serve the purpose of a properties room where devices used for studio shows can be stored. A door from this prop room leads on to the exterior of the building so that when necessary access to the room may be obtained without passing through any studio.

Studios 2 and 4 with their associated Balance and Control cubicles are in visual contact with each other, although acoustically quite separate. This arrangement has advantages when it is necessary for either cubicle to control both studios simultaneously.

Figure 16 shows the floor plan and one elevation of a studio suite proposed for Barbados Rediffusion Service Ltd. In this the natural contour of the site was used to obtain varying studio heights with the minimum amount of soil disturbance.

The important difference between this layout and that for Trinidad is that there is no continuous concrete slab forming the roof. Each studio has its own slab resting on the inner section of its double wall. This construction greatly reduces sound transmission through the roof structure.

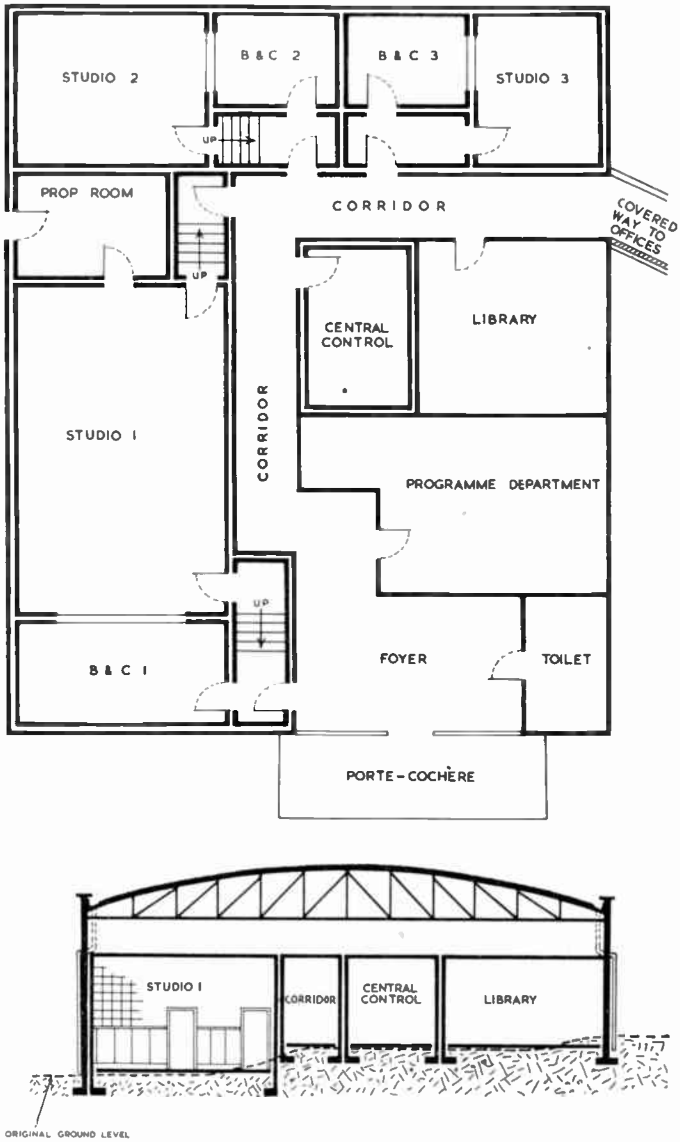


Fig. 1(b)—Plan and elevation of Barbados studios.

Weather protection for the building is provided by a corrugated sheet roof supported on steel trusses, and separated from the concrete slabs by a four foot space. This latter will accommodate the air conditioning ducts. Precautions must be taken to isolate acoustically the ducts from the roof slabs, and to prevent the ducts from short circuiting the gaps between slabs (Sect. 3.3.).

The question of access to and movement within the building has been given special attention. Note, for instance, the route between the Record Library and the small studios which use most records, the relationship of the Prop Room to the "business" end of the large studio where most live shows take place, and the separation of audience and staff entrances.

3. Construction of Building

The basic construction of a studio building as distinct from its acoustic treatment must provide three main features, namely, studios of suitable size and proportions, sound insulation from ambient noise, and sound insulation between studios and control rooms, etc.

3.1. *Size and Proportions of Studios*

The floor area of a studio will depend on the function it has to fulfil. A small talks studio may have an area of only 100 square feet, and little advantage is to be gained by making it more than 150 square feet. A general purpose studio suitable for play production and small musical ensembles of two or three pieces should not be less than 300 square feet. Six people can work comfortably in this area. Studios of about 1,500 square feet are suitable for small orchestras, and can accommodate a small audience.

Studios much larger than this can be considered as theatres or cinemas, and need special consideration. They are unlikely to be encountered in studio buildings envisaged in this paper.

The ratio between length, breadth and height of a studio has a considerable effect on its performance. Standing waves will be possible between both pairs of opposite walls, and between floor and ceiling. The frequency of the standing waves will be determined by the distances between the reflecting surfaces. Care must be taken to ensure that the frequencies of all possible waves which can be set up between opposite pairs of reflecting surfaces do not coincide, and are not harmonically related to each other.

This can be done if the ratios of length to breadth and breadth to height are both made equal to the cube root of 2. This gives a ratio of height : width : length as 1 : 1·26 : 1·6. This places the standing wave frequencies at intervals of one third of an octave.

Although this ratio is the most desirable, it can usually only be applied in the case of small studios. With large studios it requires an impractically high ceiling height, and Volkman as quoted by Olsen gives the following compromise ratios:—

Studio Volume (cubic feet)	*H* : *W* : *L*
3,000–10,000	1 : 1·25 : 1·6
10,000–30,000	1 : 1·6 : 2·5

3.2. *Sound Insulation*

Sound can enter a studio by many different routes simultaneously—walls, ceilings, floors, doors, windows, and air-conditioning ducts. All of these routes have to be treated before adequate sound insulation can be obtained. They should be considered in relation to each other.

Nothing is to be gained by making the insulation of one path much greater than that of any other. It would be useless, for instance, to make the loss through a wall equal to 70 db. if it stands on a floor which transmits sound between the areas separated by the wall with a loss of only 30 db. Nothing would be gained in this case by making the insulation of the wall more than about 35 db. The total insulation of the system would then be only slightly less than that of the floor alone, which is the limiting factor.

Few structures provide equal insulation throughout the audio frequency range. In particular, they vary considerably in their behaviour when subjected to impact and vibration noise. For instance, a massive solid partition might provide good insulation against sound waves striking it, but be capable of transmitting impact noise very efficiently.

In general it is easier to insulate against noises whose energies are concentrated in the middle and upper frequency ranges than against low frequency sounds. In most cases, if the low frequency insulation is adequate, the insulation against higher frequencies will also be adequate.

Many heavy building materials—especially concrete—can transmit low frequencies with little attenuation. It is usually uneconomic to secure insulation in this region by increasing the thickness of partitions, etc. As a general rule sound insulation against low frequencies requires discontinuous construction.

It is not usually difficult to provide as much insulation in the upper frequency range as may be required. In this range, air leaks may well be the limiting factor. High frequency sound can travel quite easily through small cracks in plaster and around doors and windows, etc., and a few of these will often be sufficient to provide quite a serious leak.

It is not always necessary to require the building structure to provide all necessary sound reduction. Before the amount of reduction which has to be incorporated in the building is finally decided, the noise problem should be examined to see whether it is possible to reduce its level before it reaches the building. Impact noise and machinery vibration can usually be dealt with in this way, and it is usually cheaper to reduce this noise at source than to incorporate additional sound insulation in the building.

3.2.1. Sound transmission through partitions

The transmissivity of a partition is a function of the logarithm of its mass. Doubling the thickness of any given partition will increase its sound insulation by not more than 6 db. This is the case if the partition is a homogeneous one, and if it is solid. The statement it not always true of very light structures. Doubling the thickness of these might only increase the insulation by 3 db. Nor is it true if cracks or other form of defect which transmit sound efficiently are present.

The transmission loss in decibels through various partitions which are commonly met in building practice are given in Table 1.

Table 1

Transmission Loss of Common Building Partitions

Material	200 c/s	800 c/s	2,000 c/s
$\frac{3}{8}$" plywood	18 db.	24 db.	35 db.
$\frac{1}{4}$" plate glass	28	32	45
2" hardwood	28	36	44
3" hollow-clay blocks, plastered both sides	40	46	48
$4\frac{1}{2}$" brick, plastered both sides ...	45	51	52

It should be noted that these insulations will only be realized if the construction is perfect, and if there is no other transmission path flanking the partition. In most practical cases, the performance of partitions will be inferior to the figures indicated above, for the reason that flanking transmission paths cannot be avoided.

The sound insulation provided by a partition of given mass per unit area can be increased if the partition is split in two and erected with an air space between each half. If the insulation of one half is n db. the insulation of both halves erected as a solid partition without an air space would be approximately $n+6$ db. It would appear that the insulation of the split partition including an air space would be $2n$ db, which would represent a considerable improvement over the previous case.

Unfortunately, however, this desirable increase can never be fully realized in practice as the coupling between both halves of the partition due to the floor and the air space etc. cannot be reduced to zero.

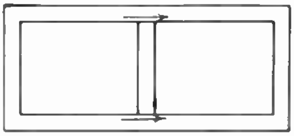
The air-space between double walls has appreciable stiffness and provides coupling. The effect of this increases inversely as the mass per unit area of the wall. With brick or hollow-clay block walls, the air space should not be less than 2 ft., and usually cannot be made more than about 4 ft. With this separation, the insulation of the composite wall may be as much as 12 db. more than the insulation of either part of it.

As the coupling between the two halves of a cavity wall increases, the insulation decreases. Every attempt should therefore be made to keep this coupling to the minimum value set by the air space. Building regulations often require the halves of a cavity wall to be tied together in some way. Any kind of rigid tie should be avoided as this will have a markedly adverse effect on the wall's performance. Flexible ties made of expanded metal or having one of the various patented constructions give the necessary mechanical strength without interfering too much with the sound insulation. Undesirable coupling can also be provided by badly designed door and window frames. This matter is dealt with in Sections 4.1 and 4.2.

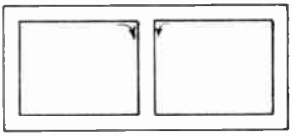
The cavity between walls is sometimes filled with some acoustically "dead" substance, such as sand, in the belief that the insulation of the wall can thereby be improved. Such treatment is based on the belief that the filler absorbs sound in the cavity and therefore prevents it being transmitted, which is a fallacy. The amount of sound absorbed is quite small. The coupling between the walls, however, is greatly increased by the filler, and therefore the total insulation of the partition is reduced. Some improvement over the single cavity wall can sometimes be secured, however, by hanging an absorbent blanket in the cavity, in such a manner that it does not couple the walls.

In this connection, it should be noted that materials which absorb sound efficiently do not necessarily act as good sound insulators. A substance such as rock wool, for instance, which is most useful as an absorber would be useless as a partition—even if it was adequately supported. Conversely, a plastered brick wall which is a good insulator, would be considered to be a poor absorber.

Acoustic treatment applied to walls only affects the transmissivity of the walls indirectly, in so far as it reduces the sound intensity (loudness) inside the room itself. Such reduction is not very large.



(*a*)



(*b*)

Fig. 2(a).—Diagram illustrating short-circuiting of cavity walls. The short-circuit can be provided by a floor, ceiling, or joining wall.

Fig. 2(b).—Diagram illustrating how short-circuit can be removed by "floated" construction.

Sound transmission along walls must be considered as well as transmission through them. In Fig. 2a two rooms are shown having outer and separating walls of the cavity type. At first glance it might be thought that the sound insulation of the party wall being high, isolation between the rooms would be good. This would not be the case, however, as sound could travel with little loss along the inner skins of the walls, thus virtually short-circuiting the party wall. Sound insulation would thus be poor. Fig. 2b shows how this state of affairs can be improved. Here, the inner walls are quite separate and there is no direct conduction path between the rooms—provided the same considerations are applied to the floor and ceiling.

The actual attenuation between a sound source in a room and a point in an adjoining room is not merely the transmission loss through the separating partition, even if other transmission paths can be ignored. It is also dependent on the ratio of area of the partition to the total sound absorption of the receiving room.

Reduction Factor
$=$ Transmission loss through partition $+ 10 \log_{10} A/S$
where

$A =$ Total absorption in receiving room
$S =$ Area of partition in ft.^{2}

It will be seen from this equation that for a wall of given size and construction, the sound reduction due to it will increase with the size of the room receiving the sound, since the total absorption will be large. Alternatively, the reduction factor will increase as the reverberation time of the receiving room is decreased.

An important effect of this occurs in the wall joining a studio to its Balance and Control Cubicle. The latter is usually a good deal smaller than the studio, but the wall separating them is common. The sound reduction between the studio and cubicle will therefore be less than that which exists between the cubicle and studio.

As will be seen later, the formula quoted can also be applied to doors and windows.

3.2.2. Ceilings and floors

Ceilings and floors are the parts of a building most difficult to deal with acoustically. Unless strict attention is paid to them, they will nullify the sound insulation provided by the walls. In any case, the insulation between two rooms cannot be better than that provided by the floor or ceiling.

Both usually have to be made of concrete for mechanical strength, long wear, freedom from insect attack, etc. These very properties make them very efficient conductors of sound. This applies especially to floors which receive impacts due to footsteps and other factors.

A cavity wall standing on a concrete floor and surmounted by a concrete ceiling can be represented by Fig. 2a. Both the ceiling and floor effectively short circuit the sound insulation provided by the wall.

If very high values of sound insulation are needed, it is essential to use "floated" construction. This entails building a room within a room and isolating one from the other by means of a compliance to form a low pass filter. This form of construction is the province of the highly specialist designer and builder, and it can easily be ruined by careless construction, or by the omission of some important point in the original design.

Several methods of making a floated construction have been devised, and some have been patented. Basically, however, they all start the process by forming a false floor—often of concrete—inside the room to be treated. This is isolated from the real floor by means of rubber blocks or some kind of resilient mat. Very careful design is necessary to obtain the correct low-pass filter effect under actual operating conditions. The weight on the compliance will affect the cut-off frequency. This weight will include the slab, its covering, and anything which stands on the floor. Therefore, if the best effect is desired, the frequency—and therefore the weight—must be kept as constant as possible under all conditions. This usually means making the concrete slab massive in comparison with the weight it is intended to support.

The next step is to form a ceiling on hangers suspended from the real ceiling. The hangers have a built-in compliance which is kept under pressure when the ceiling is completed. One end is cast into the real ceiling, the other arranged to support the reinforcing elements around which the false ceiling is then cast. The ceiling should be heavily constructed for best results.

Provided that false floors and ceilings constructed as above are not rigidly connected to the walls but have some kind of airtight seal, they will add considerably to the acoustic insulation. It should be noted, however, that they will only be effective against air-borne sound radiated from the surface of the real floor and ceiling. They will not prevent vibrational energy imparted to the structure from reappearing elsewhere—e.g. in the walls.

If insulation values of the order of 80 db. are needed between rooms, it is necessary to complete the floated construction by building false walls between the false floor and false ceiling. A complete "room within a room" will then have been constructed, and provided great care has been taken to prevent any rigid contact between the real and false structures, excellent insulations will be achieved.

It should be pointed out, however, that the artificial compliance between the false and real floor (or ceiling) is not the only compliance in the acoustical circuit. The air between the two structures is also effective, and thin "sandwiches" of it between the two can have enough stiffness to nullify the artificial compliance. The space between the false and real rooms should therefore be kept as large as possible.

When cost considerations prevent complete floated construction, steps must be taken to minimize sound transmission through the structure. This can be done to some extent by introducing breaks in floors, and mounting walls and ceilings on some kind of resilient substance which will at the same time give the required mechanical strength.

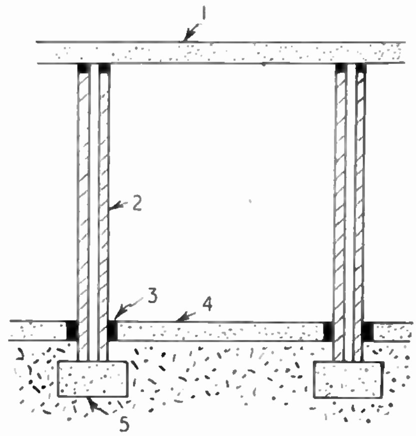


Fig. 3.—Diagram illustrating method of introducing discontinuities in walls and floors.

1 Concrete ceiling
2 Cavity wall
3 Flexible joint
4 Concrete floor
5 Foundation

Figure 3 shows a room in which breaks have been made in the floor, and between the walls and ceiling. The breaks consist of two inches of felt. The most important aspect of this arrangement is the discontinuity which is made in the floor. This minimizes the transmission which can take place between room and room along the floor, which is the most important route to safeguard.

The ceiling still acts as a direct coupling between the rooms, but when this is only subject to airborne noise, this continuity is less important than it would be in the floor. It is usually difficult to break the ceiling structure without introducing weather leakage problems. It will be noted that coupling between the walls and ceiling has been reduced by the flexible material between them.

When this kind of compromise construction is necessary, the considerations mentioned under the heading of Studio Layout (2.2) become very important.

It is most important that flexible couplings be kept flexible *at all times*. They must never be short-circuited by any form of rigid connection. Builders frequently fail to appreciate their true function and think that once they have been incorporated in the main structure, they can then be covered with plaster "for the sake of appearances." Naturally, such treatment completely nullifies the desired effect. Compliances can be covered without short-circuiting if the covering is fixed at one side of the compliance only. This point is further discussed under the heading of "Doors."

Although it might appear from Fig. 3 that a discontinuous floor structure will result in complete isolation of one slab from another, this is not the case in practice. Both the slab and wall rest on the same soil, and this is capable of sound transmission. The coupling provided by the soil is therefore the limiting factor in the amount of isolation which can be provided.

The method of breaking up floor slabs is, however, well worth-while. The only practical difficulty involved is insuring that nothing which could couple the slabs is allowed to find its way into the gap. For this reason, the gap should be made at least two inches wide as it can then easily be cleared of the debris which inevitably finds its way in during building operations. After building has finished, the gap can be filled with some bituminous or other resilient compound.

The floor covering will hide this from view without interfering with its purpose.

4. Internal Structures

The foregoing has only considered the main building structure, and the various values of insulation given assume that floors, ceilings and walls are each continuous, i.e. that none of them have openings of any kind. In practice, of course, this is not the case as openings have to be made for doors, windows, and air-conditioning ducts. It is difficult to make the insulation provided by these as high as that of the main building structure, and the effective insulation of the whole building is likely to be determined by that of the openings. These must therefore be given careful attention.

4.1. *Doors*

The low frequency insulation provided by a door will be very largely determined by its mass, and structural discontinuities. High frequency insulation is chiefly affected by the airtightness of its seal.

The heavier a door, the stronger have to be its hinges and frame and the greater the difficulty in hanging it properly. In most practical doors therefore, discontinuity of construction is used for low frequency absorption more than the mass of a solid door.

Discontinuous construction is secured by making the door a sandwich of materials of different densities. Attenuation takes place at each change of density. Materials used for the sandwich are often wood and lead. These, however, make heavy doors which are also expensive. Wood and acoustic insulation can be used to good effect, and provide doors which are a good deal lighter and cheaper. They are naturally less efficient than doors made of heavier materials.

The frame on which a door is hung needs special attention. It should be designed so that constant opening and shutting of the door will not cause it to move, otherwise the seal will be disturbed. It should also be arranged so that noises made when the door is opened or closed are not transmitted into the wall. If the door is hung on a cavity wall, the door-frame must hide the cavity without coupling the two walls.

The door seal must be arranged so that it makes an airtight joint between door and frame with the minimum pressure on either. It should be remembered that any pressure exerted on the seal is ultimately transmitted to the hinges. The hinge size will therefore be determined by the door closing pressure as well as by its weight.

The seal should also be arranged in such a way that it can be inspected easily and replaced when worn. In order to prevent wear as much as possible, the seal must be prevented from rubbing against the frame.

The portion of seal at the bottom of the door is the most difficult to arrange satisfactorily owing to the wear which takes place on the sill.

A design which takes account of the above points is shown in Fig. 4.

The door is made of sheets of hardwood and acoustic tiling firmly bolted together. It is faced on both sides with ¼ in. plywood, and the exposed edges of the laminations are covered with a light frame.

The door frame is made in two halves, one hung on each section of the cavity wall. Each frame is firmly bolted to the wall, a layer of acoustic insulation being placed in the join. This makes an airtight joint and also introduces a measure of resilience which reduces transmission from the frame into the wall.

The cavity is covered with a fillet of ¼ in. wood. It should be noted that this is fixed to one frame only, and is a sliding fit in a slot made in the other frame. This allows relative movement to take place between the walls on either side of the cavity, and makes the frame appear solid without introducing coupling.

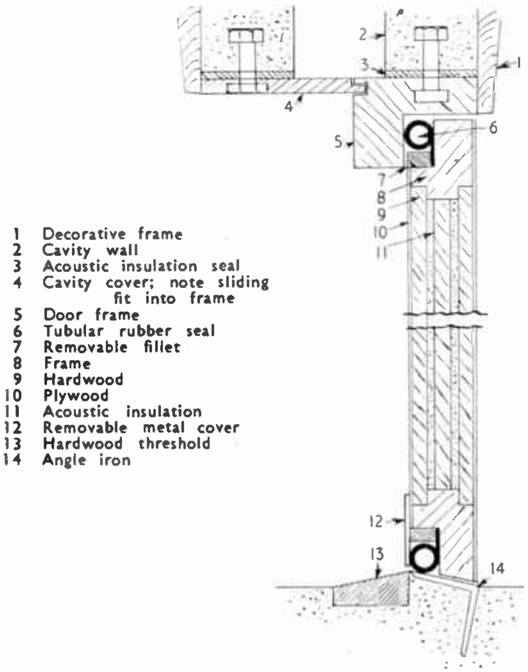


Fig. 4.—Diagram of sound-proof door (not to scale).

The seal takes the form of a sponge rubber cylinder with a tough rubber sheath and web. This compresses quite easily, and restores itself to its original shape on being released. A perfect seal can be produced by deforming it by not more than ⅛ in. It will be seen that the door can be opened and closed with the minimum sliding action on the rubber surface.

The door sill consists of a piece of angle-iron cast into the floor and sloped very slightly. This provides a surface against which the rubber on the lower side of the door can compress, and which at the same time can take heavy traffic without losing its shape.

The door seal can be inspected and removed without having to dismantle the door. Since it requires only light pressure to compress it, deformation of the door is unlikely to happen and compression can be applied at one point only.

A telephone-kiosk type of handle is used. This has a tongue working on to a sloping striking plate in the frame. Movement of the handle through a few inches compresses the rubber by about ⅛ in.

Doors constructed in this manner have an insulation of about 35 db. which is less than the walls in which they work. This difficulty can be overcome if two doors are used, and spaced far enough away from each other so that their insulations become additive. This effect is increased if the doors are placed at right angles to each other.

It is usual to place the two doors in a separate enclosure to form a sound lock or lobby. If sound absorption is applied to the walls and ceiling of this lobby, very good overall insulation can be obtained. If the doors are separated sufficiently, it can be arranged that one door is always closed even when people are entering and leaving a studio.

4.2. *Windows*

The factors already mentioned under the headings of cavity walls and doors also apply to windows.

The insulation obtainable from a sheet of glass of normal thickness is very small compared with that of the walls in which it is set. Double construction is therefore necessary, and the spacing between panes must be large enough to prevent the air space from acting as a coupling. Fig. 5 shows how the insulation of a double window made of 21-ounce glass (~2·5 mm. thickness) varies with the spacing of the panes. As is to be expected, the spacing is chiefly of importance at low frequencies.

The panes must be mounted in such a way that all joints are airtight. At the same time, they should be resiliently mounted. It is also an advantage if the windows can be removed for cleaning without damaging the frame.

Figure 6 shows the details of a double window. Each pane is set in sponge rubber, and can be removed from its seating simply by removing a fillet. Each window is sloped to minimize unwanted effects of light and sound reflections. The space inside the window is

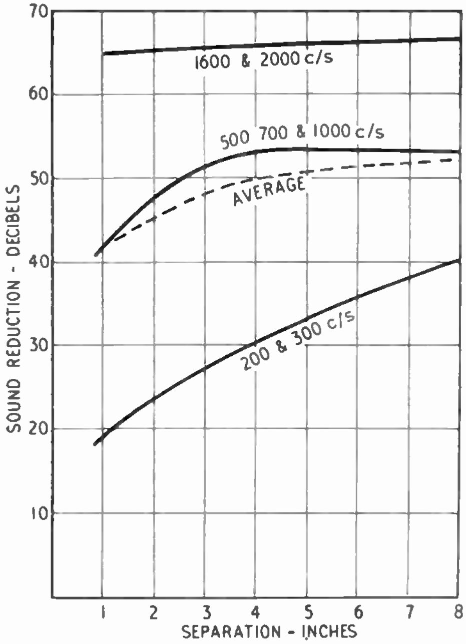


Fig. 5.—Sound reduction through double 2·5 mm. sheet glass windows of various spacings.

lined with acoustic tiles which have the double effect of covering the cavity without coupling the walls, and also of providing sound absorption inside the window itself.

The frames in which the windows are set, follow the general lines of those already discussed for doors. Two quite separate frames are made, and bolted to the wall against a pad of acoustic insulation.

4.3. *Ventilation Ducts*

Ventilation ducts are normally formed out of sheet metal, and with the areas they ventilate, form a closed air circuit. Air is passed from a plant house where it is warmed or cooled according to circumstances, through a system of ducts into the studios, and then via a duplicate set of ducts back to the plant house.

If the ducting is arranged in the most convenient manner from the ventilation point of view, it will provide a number of acoustic defects, as follows:—The ducts themselves will effectively couple walls which should be left free of each other. Sound will be conducted efficiently both along the duct walls and along the air column inside them. Noise will be able to travel with little attenuation from the plant house into the studios and also between the studios themselves. The movement of air can cause noise in the duct itself, at the point where it leaves the duct and enters the studio, and, if the velocity is too high, in the studio itself.

The design of the ventilation duct system has therefore to be considered under the following headings.

4.3.1. Wind noise

The production of noise by the movement of air can only be kept at an acceptable level if the velocity is made low. For broadcasting studios the highest practical velocity is about 600 ft. per minute, but it is usually held at a lower value than this. Since for any given volume of enclosure and occupancy the number of air changes required is fixed, it follows that the air velocity inside the ducts can only be made small if the duct cross-sectional area is made large.

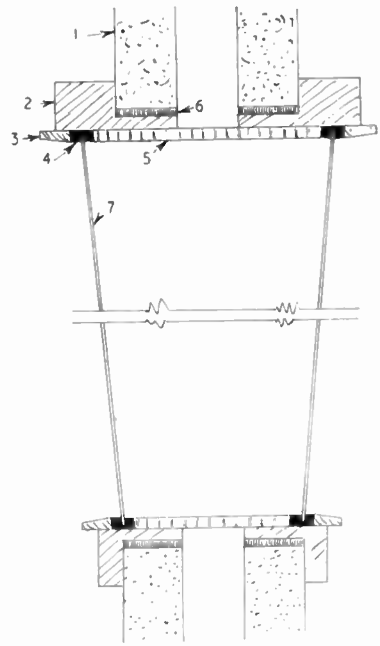


Fig. 6.—Diagram of sound-proof window (not to scale).

1 Cavity wall
2 Hardwood frame bolted to wall
3 Removable hardwood frame holding glass in position
4 Sponge rubber
5 Acoustic tile lining window cavity
6 Acoustic insulation to provide air-tight seal between frame and wall
7 Sheet glass

A low air velocity will, however, only keep noise low if turbulence is avoided within the duct. This can be done by avoiding sudden changes in cross section, and keeping bends as smooth as possible. A change of 20 deg. in duct direction can be introduced without introducing noise. When sharp bends are necessary, air-guides are usually introduced inside the bend to smooth out the air flow. A bend of 20 deg. will provide about 8 db. of attenuation for frequencies of 4 kc/s and over, and a decreasing amount for lower frequencies. It will not attenuate frequencies of 400 c/s and below.

Turbulence can also be caused by grilles and dampers. It is difficult to prevent this, but the effects can be mitigated by careful consideration of their shapes, and of their location relative to sound absorbing sections.

4.3.2. Sound transmission along duct wall

It is good practice to provide some form of thermal insulation around ducts carrying air which is more than a few degrees higher or lower than ambient temperature, usually by lagging with glass wool or asbestos. If this lagging is properly applied, it will help to damp out sound transmission along the duct wall. Its effect, however, will vary with frequency and will be quite small at the lower end of the spectrum. This means that plant rumble, etc., will be transmitted into the studios unless discontinuities are provided in the duct wall.

Flexible canvas connections are used for this purpose, and are used at any point where coupling by a continuous duct has to be avoided. The most important flexible coupling of any system is usually that which connects the fan to the ductwork.

Sound can also enter a duct through the wall as well as travel along it. Lagging assists in this direction but does not in itself provide the whole solution. So far as is possible, ducts should not be run in noisy locations unless insulated against sound entry through the wall. The proper placing of absorbing sections is also important in this respect.

4.3.3. Sound transmission inside ducts

Acoustic coupling through the air inside ducts can only be avoided if sound absorbers, or silencers, are introduced. Essentially, these consist of lengths of sound absorbing material arranged parallel with the duct walls so that air passes over them, but does not impinge on them. Baffles placed at right-angles to the direction of air flow must be avoided at all costs. Not only do they increase the air resistance of the ducts, but they cause excessive turbulence and so introduce more noise than they absorb.

The sound reduction per unit length of a duct depends on its cross-sectional area, its aspect ratio, and the coefficient of absorption of its surface. These factors are related in the following empirical formula.

$$R = 12{\cdot}6\,(P/A)\,a^{1.4}\ \text{db.}$$

where

R = reduction per foot of duct in decibels
P = perimeter of duct in inches.
A = area of duct in square inches.
a = coefficient of absorption of surface.

It will be seen that for any given area of duct, R will be large when P and a are large. A will be determined by the quantity and velocity of air passing along the duct.

The ratio P/A will increase for any given area of rectangular duct as one side is made longer than the other. A square duct has a small value of P/A. A circular duct has a smaller value of this factor than any other shape.

It is therefore advantageous to make ducts of as compressed cross section as possible.

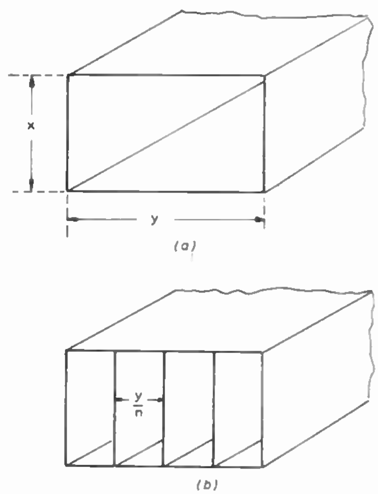


Fig. 7.—Diagram showing duct.
(a) Without splitter. (b) With splitters.

There is, however, a limit to the lengths to which one can go in this direction. If the ratio P/A for the whole duct is made very large, air resistance will be increased. The effective perimeter can be increased over some sections of the duct by introducing longitudinal splitters as shown in Fig. 7 and this is the method used when designing silencers.

Figure 7a shows a duct of width y inches and depth x inches. Its aspect ratio (P/A) is therefore equal to $\frac{2x+2y}{xy}$.

Figure 7b shows the same duct divided by splitters into n sections. The aspect ratio of the duct as a whole is the total exposed surface/area, or $\frac{2nx+2y}{xy}$. This assumes that the splitters have no thickness.

The aspect ratio for each section is $\frac{2x+2y/n}{xy/n}$ which reduces to $\frac{2nx+2y}{xy}$. The aspect ratio of the duct as a whole is therefore equal to that of any one section.

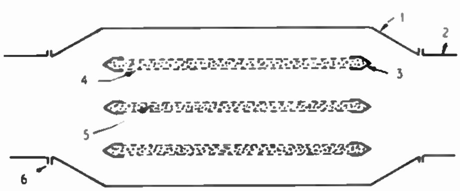


Fig. 8.—Diagram showing section through duct silencer section. Note tapers on edges of splitters.

1 Metal housing
2 Untreated duct section
3 Metal framework of splitter
4 Perforated metal
5 Felt
6 Bolting flanges

In order to make a silencer efficient at low frequencies, the absorptive material needs to have considerable thickness. The duct has therefore to be increased in size to accommodate the silencer, so that the total amount of free space between splitters is equal to that of the untreated duct. Fig. 8 shows how this is done. The edges of the splitters are tapered to reduce wind resistance and prevent turbulence.

When silencers are required to be very efficient—i.e., when the attenuation per unit length is to be a maximum—splitters can be made of felt 2 in. or more thick, supported between perforated metal. Silencers which are less efficient but which are adequate in many locations can be made with one of the various proprietary brands of acoustic insulation. This material can be formed into sections and made self-supporting. A further simple method which has been used successfully when specialist materials are not available consists of making a splitter framework of thin three-plywood, and lining it with acoustic tiling. Both the latter types of section are fitted inside a suitably enlarged metal duct section. The former type is usually made as a complete duct section, and provided with flanges which bolt to untreated duct sections.

The positioning of silencers needs careful consideration if the best effects are to be obtained.

Even after all anti-vibration precautions have been taken in the Plant Room, the noise there will still be considerable. A silencer will therefore be needed in the first portion of duct connecting the house to the studios before any branches are made. Since this duct carries all the air, its cross-section will be larger than that of any other duct, and the silencer will thus be less efficient. If, however, the plant house is placed at some distance from the studios, sufficient length should be available to make a silencer to give about 30 db. attenuation, at 250 c/s. It will be easier to incorporate an equal or greater amount of attenuation in the branch ducts on account of their reduced size.

The remaining silencers must provide acoustic insulation between studios and their control rooms, besides contributing to the isolation of the plant room. Even with small ducts, silencer lengths of several feet will be needed to provide adequate performance. Branch ducts feeding studios must be designed to provide this length. Fig. 9 shows three arrangements of duct work connecting two studios. Arrows indicate possible sound leaks. In Fig. 9a, a single duct feeds both studios without any special precautions being taken. Although this is quite a satisfactory arrangement so far as ventilation is concerned, it is very bad acoustically. A better arrangement is that shown in Fig. 9b in which the duct is placed outside the studios and a single silencer separates the two. The arrows indicating sound leak paths demonstrate, however, that this arrangement has shortcomings.

The best arrangement is that shown in Fig. 9c. The silencers are placed close to the point at which the ducts enter the studios, and the increased length gives greater attenuation. A

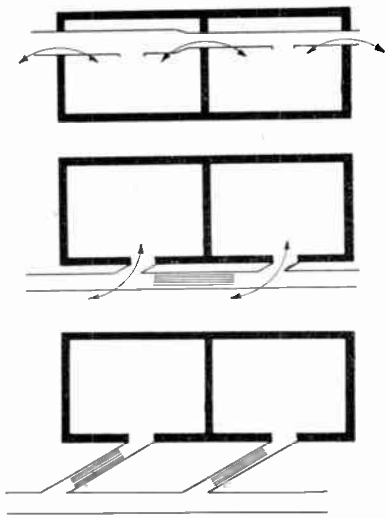


Fig. 9.—Diagram showing possible arrangements of air conditioning ducting. Arrows indicate sound leak paths. Dampers, etc., are not shown.

further advantage is that the silencers affect only the branches feeding individual studios, and so do not impede the flow of air in the main duct.

It should be remembered that the exhaust ducts require exactly the same treatment as the supply ducts.

The noise existing in the duct at the point where it enters the studio does not represent the actual air-conditioning noise level inside the studio. The latter is less than the duct noise by an amount equal to 10 $\log_{10}$ (total absorption inside the studio/duct cross sectional area). This amounts to about 10 db. for average sized studios.

The performance of an air-conditioning system is usually considered satisfactory if the noise introduced into a studio by the system does not increase the ambient level by more than 5 db. The introduced noise includes that transmitted from the plant room and from other studios, and also noise caused by the movement of air.

5. Acoustic Treatment

After the studios have been built and sound transmission problems eliminated acoustic treatment then has to be applied in order to give the studios the correct reverberation time.

Reverberation Time (R.T.) is defined as the interval necessary for a sound to decay to 60 db. below its steady state value, after the source producing it has been switched off. It is a function only of the volume of the room and the total amount of absorption contained in it.

Sabine's formula is

$$\text{R.T.} = \frac{0{\cdot}05\,V}{A}$$

This was modified by Eyring to

$$\text{R.T.} = \frac{0{\cdot}05\,V}{-S\log_e(1-a)}$$

where

V—Volume of room in ft.^{3}
A—Total absorption in sabins.
S—Surface area.
a—Average coefficient of absorption.

All measurements of absorption are related to free space. Since this has no boundaries to reflect sound, the latter cannot build up. It can therefore be regarded as having 100 per cent. absorption.

The unit is taken as one square foot of window space, and the absorption provided by any given material is compared with this. If the absorption provided by one square foot of the material is equivalent to 0·6 square feet of open window, it is said to have an absorption coefficient of 0·6. Alternatively, the absorption provided by the material is said to be 0·6 Open Window Units (O.W.U's) or sabins. The total absorption provided by any given area of the material is its coefficient multiplied by the area in square feet.

The optimum value of Reverberation Time for any given enclosure varies with frequency, the volume, and to some extent on the purpose for which it is to be used. A room to be used for speech, for instance, will normally require a shorter value of reverberation than would the same room used mainly for music. By allowing the sound of one syllable to die away before the next one is spoken, a short reverberation time aids speech clarity. The same conditions would cause music to sound flat and unnatural, because the ear expects an appreciable reverberation period which allows sounds to merge into each other to some degree.

The optimum value of Reverberation Time for a studio of any given volume is a matter about which there has been some discussion. Opinions as to its exact value in the middle frequency range differ somewhat. Opinion as to whether it should rise or fall at the upper and lower end of the frequency spectrum differ still more.

The fact of the matter seems to be that ideally, there is a Reverberation Time curve for almost every use to which a studio might be put. In practice, it is very difficult to arrange this. The curve shown in Fig. 10 does, however, give a guide to performance. If the studio reverberation time is based on this information during the original design, it will be approximately correct when constructed. Adjustment is always necessary on site in order to make a studio acceptable.

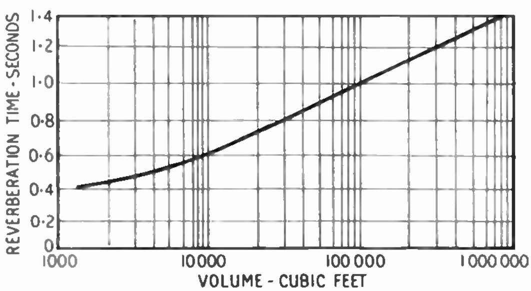


Fig. 10.—Graph showing studio reverberation time v. studio volume.

For fairly large studios which might be used with or without an audience, it is satisfactory to make the calculated reverberation time correct with half audience occupation—unless the expense of adjustable absorption can be entertained.

For small talks studios it is safer to err on the side of "deadness," as this provides more comfortable conditions for speakers. Some apparent liveness can always be introduced by suitable microphone placing.

A practical difficulty arises in making the high frequency time correct, as most materials which are efficient absorbers at low frequencies are even more efficient at high frequencies. If a choice is necessary—as it usually is—the low frequency part of the spectrum must be dealt with at the expense of the high.

A studio which has too low a reverberation time at high frequencies will sound rather lifeless, but it will be usable. A studio which is too live at low frequencies will sound "boxy" and will be quite unusable.

The first step to be taken when designing acoustic treatment is, therefore, to decide on how the low frequencies are to be absorbed. The type of treatment used will influence the general design quite considerably.

There are three main methods of absorbing low frequencies, namely, membrane absorbers, resonant panels, and thick blankets of fibrous material.

Membrane absorbers are very efficient units which can be tuned to any desired frequency. They behave like acoustic resonant circuits, the frequency and damping of which can be varied over quite wide limits. If several of them are tuned to various frequencies, they can be made to form a system which will give any required absorption over a desired frequency range.

Great care is necessary in their use. As they operate over a fairly narrow band, the correct "placing" of the resonant frequencies is a matter of great importance. It requires fairly accurate measurement, the means of making which are not always available to the engineer in the field.

Resonant panels are most useful devices, as they not only provide low frequency absorption, but also a surface which is both attractive and easily kept clean. They can be constructed of $\frac{1}{2}$ in or $\frac{3}{4}$ in. plywood, the panels measuring 5 ft. or 6 ft. high, and 1ft. 6in. to 3 ft. across. Variation in width distributes the frequencies of maximum absorption.

Each panel should be mounted on vertical battens fixed securely to the wall. Panels are fixed at the vertical edges only. The upper and lower edges are left free so that the panel can vibrate as a whole. A capping can be provided top and bottom in order to improve the appearance, but this must not be joined to the panel in any way. A space varying from 2 in.–4 in. should be left behind the panels, and filled with some absorbent substance such as glass or rock-wool. If this precaution is not taken, the panels can radiate.

Thick blankets of fibrous material nearly always find their place in studio acoustic treatment. They absorb over the whole frequency range, but fortunately, their behaviour at middle and high frequencies can be modified by the way in which they are covered. Their performance at low frequencies is not greatly affected by this factor provided the surface is not made solid and rigid.

Considerable thickness is needed in order to obtain adequate low frequency absorption. Reference to the table of Absorption Coefficients will indicate that the absorption coefficient of felt, for instance, at 125 c/s,

Table 2

Absorption Coefficients

Material	Thickness	Frequency in c/s 60	120	250	500	1000	2000	4000
Rock Wool faced with perforated metal	3″		·35	·6	·75	·85	·8	·7
	5″		·5	·6	·75	·85	·8	·7
,, ,, muslin facing	4″		·46	·61	·82	·82	·64	·6
,, ,, unfaced	3″	·17	·45	·7	·75	·75	·75	·85
,, ,, covered ⅛″								
,, ,, unperforated hardboard ...	3″	·34	·56	·5	·34	·25	·21	·21
,, ,, covered ⅛″ hardboard perforated 3/16″ dia. holes at ⅜″ centres ...	3″	·17	·5	·7	·68	·68	·6	·5
Brick (painted)		·011	·012	·014	·017	·02	·023	·025
Carpet (thick pile)	—	—	·09	·08	·21	·26	·27	·37
Heavy Curtains	—	—	·05	·12	·35	·45	·38	·36
Felt, painted cotton fabric cover ...	1″	·1	·16	·3	·42	·5	·38	·2
	2″	·32	·33	·38	·47	·52	·42	·25
	3″	·39	·4	·42	·5	·52	·42	·25
Insulating Tile	½″							
,, ,, ...	1¼″	—	·28	·56	·98	·78	·59	·49
Audience, per person		—	2·3	3·2	4·8	6·2	7·6	7·0
Upholstered chairs		—	—	3·1	3·0	3·2	3·4	

varies considerably with thickness, whereas at 1,000 c/s, little change in absorption occurs as the thickness increases above a minimum value.

This fact explains why large surface areas of thin felt, carpets and similar materials reduce the middle and upper range reverberation times severely without sufficiently affecting low frequencies. The same volume of felt applied thickly over a smaller area would have a much more balanced effect on the absorption.

The curve of frequency against absorption for thick fibrous substances can be varied over quite a considerable range according to the manner in which the surface is treated. If left without treatment of any kind, the curve will rise to a maximum about 1,000 c/s and remain at this high value throughout the upper frequency range. Surface treatment which tends to make it impermeable—e.g., painting, or covering with glued muslin, etc.—leaves the low and middle range almost unaffected, whilst lessening the high frequency absorption to quite a marked degree. Perforations in the surface allow curves of intermediate shape to be obtained, the effect on the high frequency response varying inversely as the degree of perforation. Thin sheet metal which may or may not be perforated can also be used as a covering.

No one material is capable of providing the correct amount of absorption at all frequencies. A mixture of materials is therefore necessary. It should be remembered that acoustic treatment cannot be confined to one area of the studio. It has to be distributed over all walls and ceiling, as in most cases, it is undesirable to have large unbroken reflective surfaces. This means that maximum efficiency of a material is not the only criterion to be considered. It is often necessary to use materials which are relatively inefficient at some frequencies in order to distribute the effect over a large enough area.

Probably the most universally used acoustic absorber is perforated celotex tile which is available in several proprietary patterns. It has the advantage of being very easily applied. It can also be coloured and redecorated without losing its efficiency so long as precautions are taken to ensure that its holes are not filled up.

Celotex tiles are effective over quite a wide frequency range, and can normally be relied upon to provide the bulk of absorption in the middle and upper ranges. Their absorption varies with thickness.

Acoustic plasters depend for their action very largely on the method of application which requires special skill. Their use should not be considered unless this skill is readily available. Considerable care also has to be taken in applying decoration.

The total absorption in a studio is not limited to that which is applied to walls. Any object inside the studio adds its quota, which might be quite a considerable portion of the whole. Tables, chairs and personnel are those chiefly encountered. The absorption due to them is shown in Table 2.

After the question of absorption has been considered, attention must be paid to the provision of diffusion. Although the reverberation time of small studios is quite short, it is not negligible. Since all sound generated is not absorbed, some of it must be reflected. The object of providing diffusion is to ensure that these reflections are dispersed and not allowed to set up objectionable standing waves. These will be produced if reflecting surfaces are allowed to be parallel to each other.

The frequency of standing waves which can be set up between two reflecting surfaces is a function of the distance between them expressed as the wavelength of the sound in air. It will be seen therefore, that in small and medium studios, standing waves are most likely to occur at frequencies for which it is most difficult to provide absorption.

Focusing effects are even more objectionable than pronounced standing waves, and must be avoided at all costs. They are caused by concave reflecting surfaces. If a concave surface is essential for some aesthetic purpose, it should be made as non-reflective as possible, and its focus placed well away from any microphone or listening location.

Diffusion can be provided by two main methods. One is to make flat surfaces non-parallel with each other. The other is to provide convex surfaces which disperse sound reflections and prevent them reinforcing each other.

Standing waves can also be minimized if absorbing material is disposed in such a way that necessary parallel surfaces are not highly reflective.

A combination of these methods is usually employed. One of the most effective ways of providing diffusion is to fit convex panels to the walls. These can be made of thin plywood supported on curved ribs, the latter being spaced at random to prevent the air columns of all panels from resonating at the same frequency. Such panels give some bass absorption, and can be filled with thick absorbing material if necessary.

Windows are usually the worst reflective surfaces encountered in a studio. They can however, be sloped in such a way that sound reflected from them is directed to the floor instead of back to the source. This slope is also of assistance in preventing troublesome light reflection.

6. Studio Equipment

The design and layout of studio equipment should be considered in detail during the early planning stage of the whole building, so that building plus equipment can be made into a single functional unit. In an endeavour to create elegant circuit arrangements, engineers sometimes forget that the only function of studios is to produce programmes. It is not surprising that such an attitude can result in equipment which defeats the object of the building as a whole. It can be avoided if the interests of programmes, engineers, and architects are related at the start of the project.

Studio equipment should allow the Programme staff* to carry out their operations with the maximum convenience and minimum fatigue. Provision of the correct facilities, arranged in the most effective layout, will secure speed of operation and relative freedom from operational error.

The question of speed is of paramount importance in commercial studios. The introduction of "commercials," many of them of very short duration, greatly increases the number of operations which have to be carried out during a given period as compared with non-commercial studios.

Ideally, the Operator should be able to reach any control or turntable without having to move

* In order to avoid confusion between functions, the person using microphones in the studio is always referred to as the "Announcer," although he (or they) may in fact be artistes not on the announcing staff. Similarly, the person operating equipment in the Balance and Control Cubicle is referred to as the "Operator," although he may, in various circumstances, be either an Engineer or a Producer.

from his seat, and the equipment layout should be such that he can operate on one part of it without disturbing another. He must also have a clear view of everything which is happening in the studio.

The following paragraphs list the facilities required in studios, and detail how they have been provided by the Trinidad Broadcasting Company Limited.

7. Facilities Required

Although the requirements of studios used for different purposes vary quite widely, most facilities fall under one of the following headings:

(*a*) Control of microphones and turntables by means of faders. It is often an advantage if they can also be controlled by a key, so that when the level has been set on the fader, the microphone or turntable can be switched in and out of circuit instantaneously. It is often an advantage to have one or more line inputs into a studio. This is particularly the case in continuity studios, in which the studio is required to link up with programme sources outside itself. Each line must have its own associated fader. The number of channels required varies, but seven appears to meet most requirements. This number of channels would be divided between turntables, microphones and lines, according to the use to which the studio is to be put. Each channel must be controllable by its own fader, the setting of which must not affect that of any other fader by more than about 1 db, in order to make mixing easier. In addition, there must be a master fader which controls the output of the entire equipment.

(*b*) The Operator in the Balance and Control Cubicle must be able to hear the output of his studio over a loudspeaker.

(*c*) He also needs to be able to hear the output of certain channels before they are placed on service, so that he can properly cue in turntables, etc. These circuits are variously known as "Cue" or "Prefade", and it is very convenient to control them by means of keys.

(*d*) It is very useful for the Operator to be able to permit the Announcer to hear the output of a turntable or of a line over the studio loudspeaker before the channel is placed on service.

(*e*) The Operator must be able to speak to the Announcer over the studio loudspeaker without moving from his seat. There are two conditions under which this facility might be used. The first of these is the "rehearsal" condition in which the Operator (who in this case will be a Producer) requires to give the Announcer instructions. He must, therefore, be able to interrupt the Announcer when the latter is using the studio microphone. The second condition will obtain when the studio is on service. In this case it is essential that the Operator should be *un*able to interrupt the Announcer while the latter is on the air. There is a big advantage, however, to be gained from leaving the speak-to-studio facility operative when the Announcer is not actually on the air, although the studio is—for example, when the studio is using a turntable or line input. Errors will be avoided if this changeover between the "rehearsal" and "on service" conditions can be made automatic when the studio is placed on the air.

Fig. 11.—General arrangement of studio control panel.

(*f*) When the studio is on the air, but the studio microphone is not in use, the Announcer should be able to speak to the Operator over the normal studio microphone, without going on the air.

(*g*) It is often necessary for the Announcer to place himself on service. There are many occasions when this facility increases the speed of operation. The level must, of course, be controllable by the Operator's fader, who in any case has over-riding control.

(*h*) The Announcer and/or Operator needs additional facilities, such as keys for operating time signals, remote control of tape recorders, telephone to Central Control, etc.

8. General Equipment Arrangement

The general arrangement of equipment is shown in Fig. 11. Three turntables are placed on a desk, all of them within easy reach of the Operator. Each turntable can be started by its own speed control lever, or by a foot switch located under the desk.

All faders and keys needed by the Operator are located on a panel let into the desk. This panel is tilted for easy operation, and is of such a height that he can work on turnables without being in danger of accidentally moving a key or fader.

Fig. 12.—Studio control panel with key and relay panel swung forward.

All faders are placed in a row, the master being that on the extreme right, and a key is placed above each one. Those above microphone faders are able to place the microphone on service when depressed, providing the fader is open. Those above the turntable faders are pre-fade keys which connect the turntable output to the pre-fade loudspeaker without placing it on service. The key above the master fader operates the "Speak-to-Studio" facility when depressed. When pressed upwards, it places the output of the pre-fade circuit onto the studio loudspeaker.

Relays are mounted behind the panel, and are just as accessible. Fig. 12 shows the control panel lifted to expose keys, faders and relays. The control panel relay rack and mixer form a single unit which is mounted in a metal screening box let into the console.

Amplifiers for microphones and turntables are of the plug-in type housed in a cabinet which is quite separate from the console. Connections between the amplifiers and control panel are made by multi-way plugs and sockets. Amplifiers can thus be reached without interfering with the desk. Also, the entire control panel can be removed from the console, if necessary.

Figure 13 shows the equipment arrangement in the main continuity studio. Control keys, warning light and lectern are integrated in one fixture in the centre of the table.

9. Circuit Details

A schematic of a typical studio equipment arrangement is shown in Fig. 15. The various facilities operate as follows:

9.1. *Mixing Circuit*

It was decided to adopt the simplest possible mixing circuit, as shown in Fig. 14. This form of circuit uses potentiometers which are simple and therefore relatively inexpensive. It is a very simple matter to ensure that the setting of one fader does not affect the level on any other, by using isolating resistors. These are made equal to twice the resistance of any one potentiometer.

Fig. 13.—Control panel in main continuity studio.

A seven-channel circuit of this type has a loss of approximately 20 db, but this does not present any serious problem, as the level into each channel can be made quite high. The value of potentiometer chosen for this circuit was 10,000Ω, as this is a standard design.

9.2. *Amplifiers*

Only two types of amplifiers are required in this system. The first is a voltage amplifier with a gain of about 80 db, to provide channel inputs from turntables and microphones. A cathode follower is very suitable as an output stage, and obviates the need for a transformer.

The second amplifier has to be capable of driving loudspeakers and lines, but requires a gain of only about 30 db.

Neither of these amplifiers needs to have gain controls. The use of large amounts of feedback and close tolerance resistors in the early stages ensures that the gain of all amplifiers of one type is the same. Amplifiers are thus interchangeable.

9.3. *Microphone Keys*

Microphone keys are wired so that in their resting positions they place a short circuit across the appropriate fader. As will be seen from Fig. 15, one microphone circuit has a pair of relay contacts in series with the key. This relay can be operated by the Announcer, who is thus enabled to place himself on service when necessary.

A further contact on the microphone key operates the studio loudspeaker cut off (STD. LSCO) relay, which ensures that the loudspeaker cannot normally be on at the same time as the studio microphone. A warning light, indicating when the microphone is on the air, is operated by an additional relay (WR) placed in parallel with STD. LSCO.

9.4. *Turntable Cue Keys*

Turntable cue keys are sequence-wired, and their output feeds an amplifier which drives a Cubicle loudspeaker. The operation of a cue key places the cue amplifier across the appropriate programme source. As the amplifier has a high input impedance, its shunting effect is negligible.

9.5. *Speak-to-Booth*

The microphone amplifier, having a cathode follower output, has an output impedance of only a few hundred ohms. Since the load which it feeds (the potentiometer) is 10,000 Ω, it is possible to place a 1,000-Ω resistor in series with the potentiometer without affecting the level across the fader by more than 1 db. The

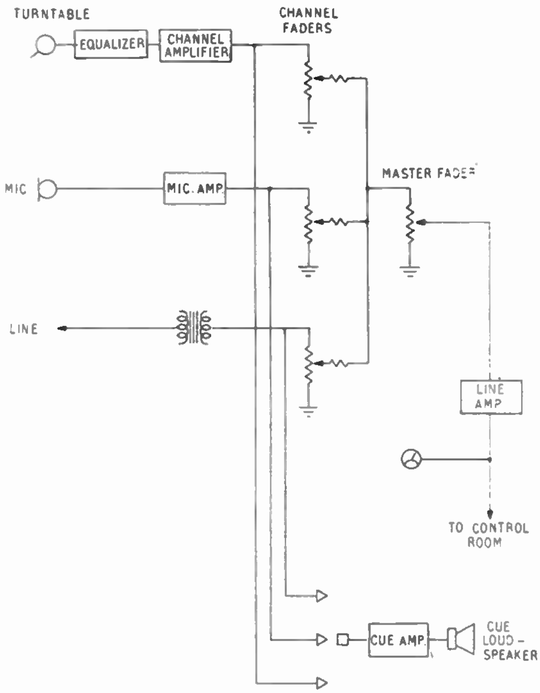


Fig. 14.—Basic mixing circuit.

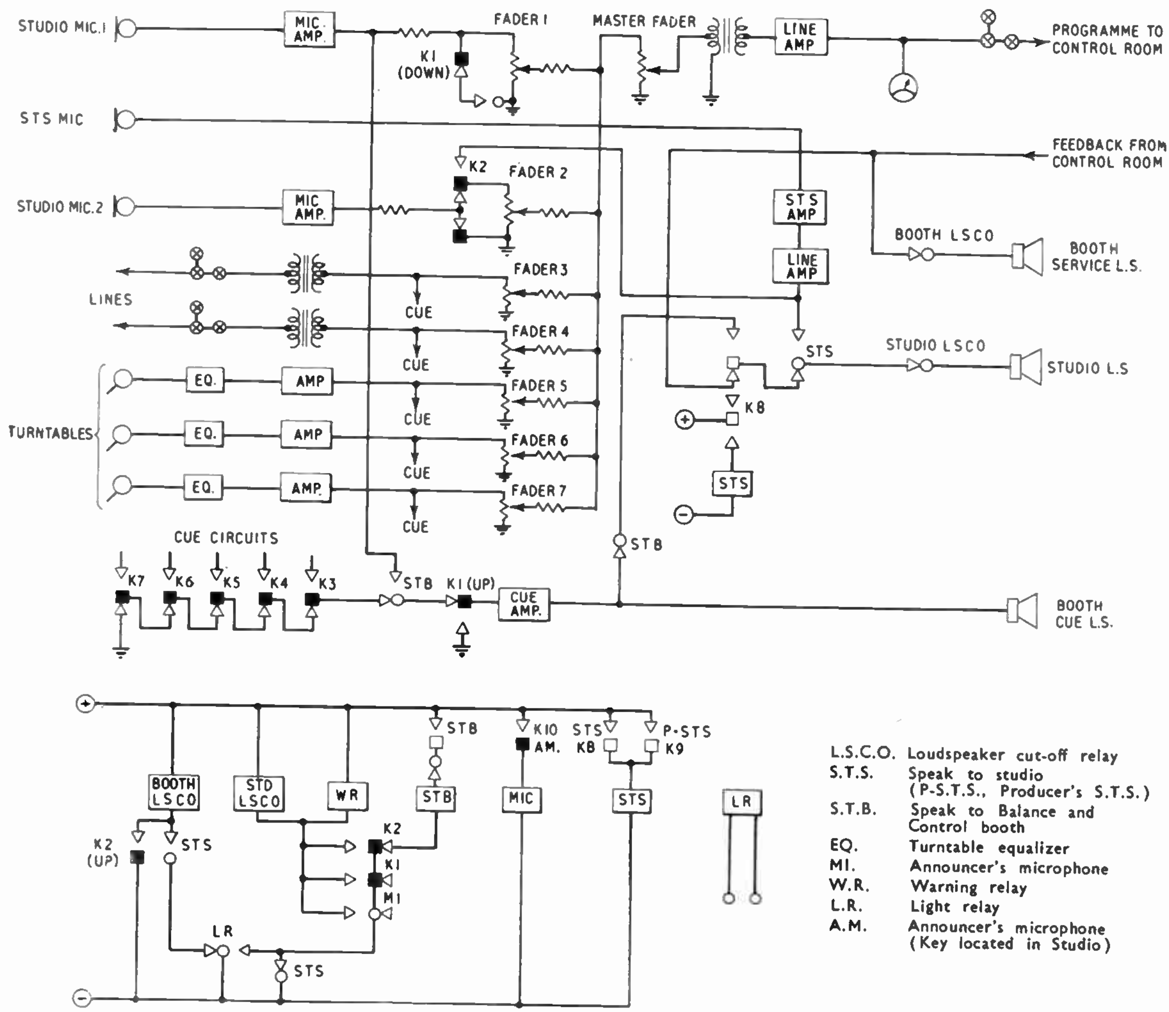


Fig. 15.—Complete studio schematic.

1,000-Ω resistor is, however, still quite large in comparison with the amplifier impedance, so that when the microphone key short circuits the potentiometer, most of the amplifier voltage appears across 1,000 Ω. This fact is made use of for the "Speak-to-Booth" facility.

When the STB key is depressed, the STB relay operates and transfers the cue amplifier input to the studio microphone amplifier output. The Announcer is thus able to speak to the Operator over the booth cue loudspeaker, using his normal service microphone.

This can, however, only happen when the microphone is not on service. Whenever the microphone key is operated, the STB relay coil circuit becomes open circuited, and the facility, therefore, cannot operate.

9.6. *Speak-to-Studio*

Operation of the "Speak-to-Studio" (STS) key operates the STS relay. The contacts of this transfer the studio loudspeaker to the STS amplifier, and the Operator is thus able to speak into the studio, using a microphone suspended over the console. This arrangement makes it a simple matter to provide a producer with his own talk back key, which may be located in any convenient position, often beside the desk. This key is shown in Fig. 15 as P-STS.

An additional pair of contacts on the STS relay are placed in series with the STB relay and operate whenever the STS key is depressed. This ensures that the Operator has priority over the Announcer during rehearsal conditions. It also ensures that the circuit cannot howl-round.

A further pair of STS contacts are placed in series with the common negative of all the microphone keys. During rehearsal conditions, operation of the STS relay will, therefore, open circuit the studio LSCO relay. This allows the Operator to interrupt the Announcer in order to give instructions, etc.

When the studio is placed on service, Central Control operates the normal red "on service" light, which gives warning to people about to enter the studio. This warning light is made to operate the relay "LR." "LR" has one changeover contact. When operated, this contact short circuits lower pair of STS contacts. Operation of the STS key and relay cannot, therefore, interrupt the Announcer whilst he is on the air. Whilst the studio is on service, but the Announcer is not on the air via the microphone—as for instance when a disk is being played—the studio LSCO contacts are in their resting position and the STS facility can operate. This is a most useful arrangement as it is often necessary for the B & C Cubicle to pass information to the Announcer.

9.7. *Booth Loudspeaker Cut-off*

When the Operator interrupts the Announcer during rehearsal, howl-round could take place via the booth loudspeaker if a cut-off was not provided. A pair of STS contacts provides this cut-off via the booth LSCO relay and the resting contacts of LR. When LR operates, however, the booth LSCO cannot normally be switched off, and the Operator is, therefore, assured of hearing a continuous programme once the studio is on the air.

9.8. *Operator/Announcer Working*

Additional flexibility can be secured if the B & C Cubicle is enabled, when necessary, to be used as an Operator/Announcer studio. Simple "disc jockey" shows can easily be handled by a single person. This facility is arranged by making the output of the STS amplifier available on an up pair of microphone key contacts. Thus, when the microphone key is pressed upwards, the STS microphone is connected to a service channel and the Operator can announce his own programmes.

An additional pair of contacts on this microphone key operates the booth LSCO.

The output of the STS amplifier and the input to the microphone key contacts are brought out to jacks on the front of the console. A patch cord, therefore, has to be inserted between these jacks in order to set up the facility. This precaution is taken to avoid accidental connection of the STS microphone to service when not required.

The microphone key is made to disconnect the cue amplifier input from the cue keys. This is necessary to prevent the output of the cue loudspeaker from being fed into the booth microphone when this is in use on service.

9.9. *Feedback from Central Control*

It will be seen from Fig. 15 that both the studio and booth service loudspeakers are not energized directly by the studio equipment. They are fed from Central Control. As will be seen from details of Central Control, this makes it very simple to provide any studio with an input from any network to which it may be connected. This makes for easy continuity without the studio Operator having to take any special steps in order to secure it.

Central Control also provides feedback for the loudspeakers during rehearsal conditions, and thus is automatically kept informed of studio usage.

10. Central Control

Figure 16 shows a schematic diagram of the equipment arrangement in Central Control. Fig. 17 shows a detailed arrangement of a selector key. Fig. 18 is a photograph of the Control Desk itself.

The Desk is arranged so as to be able to provide programmes for three separate networks. The output from any studio can be used for any network, or for all three simultaneously. The Desk also accepts inputs from a receiving station, outside broadcasts, etc. A total of twelve inputs are available and can be set up simultaneously.

Monitoring arrangements permit the Operator to monitor both programme sources and the outputs of transmitters and substations. Direct comparisons are therefore possible. Monitoring can be carried out on a loudspeaker or earphones and a meter of the same type as used in the studios.

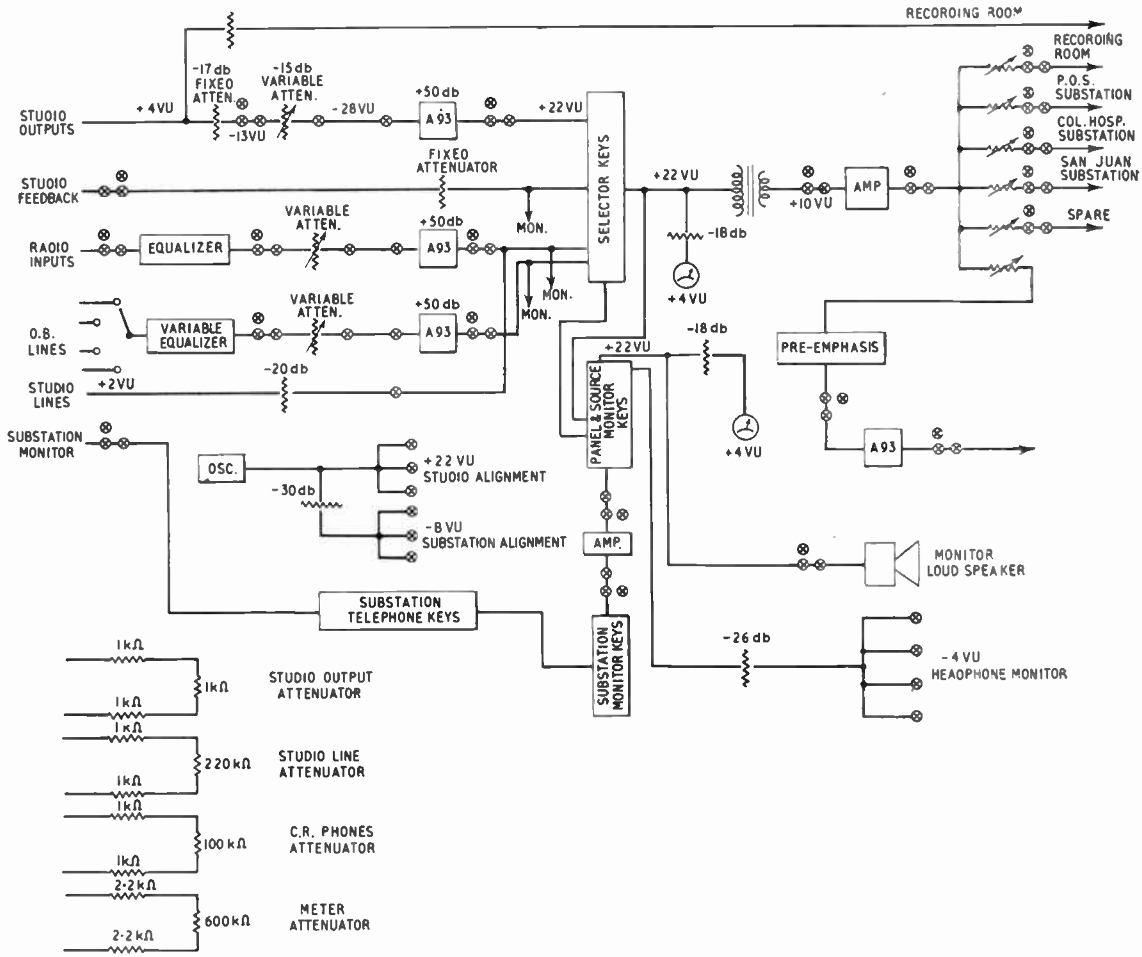


Fig. 16.—Control room basic schematic.

The telephone system is built into the Desk and enables the Central Control Operator to keep in touch with programme sources, as well as with sub-stations, etc.

10.1. *Programme Circuit*

All programme circuits are brought into the Desk via a patching board and a variable attenuator. In the case of studio outputs, which are at a higher level than those arriving via long telephone lines, a fixed attenuator precedes the variable. The output of the variable attenuator is used to drive a line amplifier (A.93), which gives an output level of 10 volts and has adequate power available to drive loudspeakers in Central Control and studios. It also has a low internal impedance.

The mixing device is a transformer. When more than one source is in use, the A.93 outputs are placed in series by the selector keys across the primary of this transformer. The impedances are arranged so that the level from any one source is not affected when other sources are added to the network.

The transformer secondary feeds a Limiter Amplifier, which has enough power available to drive all the output lines, with an adequate reserve. The Limiter is arranged so that when

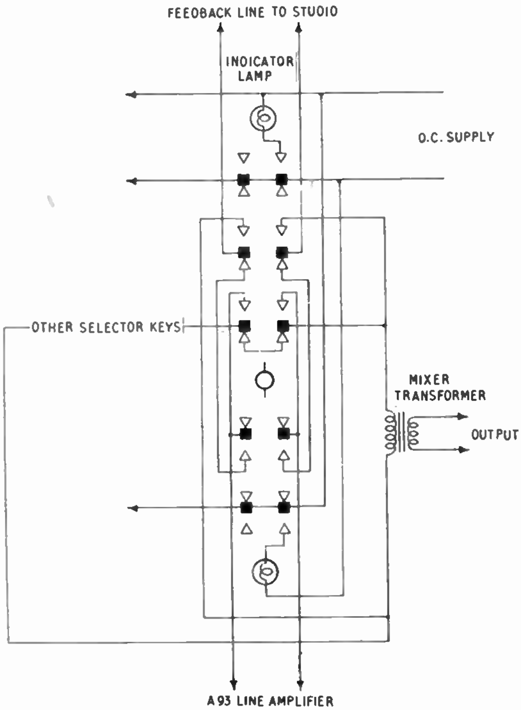


Fig. 17.—Wiring of selector key.

When lower contacts move down, output of line amplifier is returned direct to studio for rehearsal. When upper contacts lift, line amplifier is placed in series with mixer transformer. Output of entire panel is fed to studio.

the level is properly controlled, no limiting action takes place.

A source is connected to the mixing transformer by depressing a selector key. Operation of this key switches on a red "channel engaged light" above it, and also connects the transformer primary to the studio feedback line.

Operation of the selector key on A and B panels upwards, lights a green "channel available light" underneath the key. A key is put in this position when the Central Control Operator has checked a source and is sure that it is ready for service. On C panel, operation of a selector key upwards merely returns the A.93 output direct to the studio feedback line, thus putting the studio in a rehearsal condition.

Lines from the receiving station require equalization, and a fixed equalizer is wired into the circuit. Any radio line can be connected to this equalizer by means of the patching board. Outside broadcast lines differ in length and require a variable equalizer.

10.2. *Lines to Studios*

Lines from the receiving station circuit are taken via the patching board to the studios, where they can be put on service by the studio Operator. This arrangement makes for better continuity in the studio, and relieves the Central Control Operator of a function which should be properly carried out in the studio. When necessary, however, the input from the receiving station can be placed on service via a selector key.

10.3. *Monitoring*

Since the output of the A.93 is at high level, no further amplification is necessary for monitoring purposes. The monitor loudspeaker can thus be connected directly across the amplifier output via the monitor key.

Monitor inputs from substations are, however, at a considerably lower level, and an amplifier is necessary for these in order to bring them up to the same level as the A.93's.

11. Electric Installation Precautions

11.1. *Earth System*

The success or otherwise of a system such as this, which uses unbalanced circuits, depends very largely on the care which is devoted to the design of the earthing system. It is absolutely essential to avoid earth loops. This can be done if each channel is earthed at one point only, such as the amplifier input. All earth wires should be well insulated until they reach this point, and precautions taken to see that

Fig. 18.—Central control desk.

no earthed conductor (including cable screens) is earthed at more than one place.

11.2. *Mains Wiring*

All mains wiring is in conduit and routes are arranged so that they are always at right angles to microphone wiring, or if this is impossible, at least three feet away from it. The earthing of the conduit is most important. Some contractors connect this to the supply neutral. When this is done, the conduit is placed in parallel with the neutral, and in most cases carries quite a high proportion of the neutral current. This arrangement completely nullifies the screening effect of the conduit and makes the circuit behave like an unscreened single conductor, which can induce quite severe hum into microphones, etc. The cure is to ensure that the neutral is not connected to the conduit system at more than one point, and that this point is outside the studio building.

12. Acknowledgments

The Author wishes to thank the Directors of the Trinidad Broadcasting Company Ltd., Rediffusion (Trinidad) Ltd., and Central Rediffusion Services Ltd., for permission to publish this paper.

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PREMIUM WINNERS 1954

J. C. BOSE PREMIUM

Dr. S. Deb was born in 1921 at Sylhet, India, and obtained a First Class M.Sc. degree of Calcutta University in 1944. He then joined the Wireless Laboratory of the University as Research Assistant, and carried out researches on electrical discharges and radio valves. In 1947 he took up an appointment as Secretary of the Radio Research Committee, under the Council of Scientific and Industrial Research of the Government of India, and continued in this post until the end of 1950.

In January 1951 he returned to Calcutta University as a lecturer in the Institute of Radio Physics and Electronics. Dr. Deb received his D.Phil. degree in 1954 for research on radio valves and on the upper atmosphere. He is the author of a number of original papers published in Indian, British and American scientific journals, and he has been awarded the J. C. Bose Premium for his paper published in the *Brit.I.R.E. Journal* on "Decay of Emission from and Oxide-coated Cathode due to Adsorption of Matter Liberated from the Anode."

LOUIS STERLING PREMIUM

The Louis Sterling Premium for the most outstanding paper on television technique has been awarded to **Mr. Leslie H. Bedford,** O.B.E., M.A., B.Sc.(Eng.), (Past President) for his paper on "Problems of Television Cameras and Camera Tubes."

Born in London in 1900, Mr. Bedford is a graduate of the City and Guilds College and of the University of Cambridge, where he was in residence at King's College. He received his first appointment at Standard Telephones and Cables Ltd., in 1924, and in 1931 joined A. C. Cossor Ltd., where he was concerned with television research.

Just before the war, Mr. Bedford undertook receiver development work for the first radiolocation stations, and in 1943 was appointed an O.B.E. in recognition of his services. Since 1947 he has been Chief Television Engineer with Marconi's Wireless Telegraph Co. Ltd., and he is also responsible for certain Government development work with the English Electric Co. In 1952 Mr. Bedford was honoured by the City and Guilds of London Institute for his distinguished services in engineering by the award of a Fellowship.

Mr. Bedford was made a Vice-President of the Institution in 1945, and was elected President in 1948. He was chairman of the Television Engineering Session of the 1951 Convention and has served on a number of Institution Committees. A portrait of Mr. Bedford was included as a frontispiece to his Presidential Address in the January 1949 *Journal.*

MARCONI PREMIUM

Born in Glasgow in 1915, **Mr. William T. Brown** was educated at Glasgow High School. At the beginning of the war he joined the Air Ministry as a technical assistant, but left in 1941 to take a commission in the R.A.F. During the war years he was engaged in the operational development of radio blind landing apparatus, and enemy signals intelligence services.

Mr. Brown joined British Telecommunications Research Ltd. in 1946 and has been in charge of the V.H.F. Radio Multichannel Group since 1949. He has been awarded the Marconi Premium for his paper on "Some Factors in the Engineering Design of V.H.F. Multichannel Telephone Equipment," which was considered to be the most outstanding engineering paper published in the *Journal* in 1954.

Mr. Brown was elected an Associate Member of the Institution in 1955.

BRABAZON PREMIUM

The premium for the most outstanding paper dealing with radio or electronic aids to aircraft safety published in the *Journal* during 1954 was awarded to Messrs. J. W. Jenkins, J. H. Evans, G. A. G. Wallace and D. Chambers, for their paper on "A High-definition General-purpose Radar." The authors were all with A. C. Cossor, where Mr. Jenkins is head of the Research Division. Mr. Evans, who is an Associate Member of the Institution, is now with Cossor (Canada) Ltd.; an account of his career was published in the *Journal* in April 1953.

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# of current interest . . .

Post War Scientific Developments in Germany

The Parliamentary and Scientific Committee was recently addressed by Professor W. Heisenberg, For.M.R.S., Director of the Max Planck Institut für Physik, Göttingen, and Adviser on Atomic Energy to the Federal Government of Western Germany, on the subject of post-war scientific developments in Germany, particularly with regard to atomic energy.

Professor Heisenberg said that the two main research organisations re-established in Germany after the war were the Max Planck Society and the German Research Council, which sponsored research in entirely different ways. The Max Planck Society comprised about thirty-five independent research institutes, separate from universities and colleges of technology, dealing with research on physics, chemistry, biochemistry and medicine, and with applied research on metals and coal, etc. The German Research Council was founded in the early 1920's, and its aim was to help universities and colleges either by giving equipment or by the award of fellowships and grants to research workers and students.

At the end of the war, few scientific establishments possessed any equipment for nuclear research. Since 1951 a certain amount of equipment had been provided for research institutes, but a difficulty had been the heavy cost of really modern apparatus. It was hoped to co-operate with the European Council for Nuclear Research (C.E.R.N.), in the building of two particle accelerators at Geneva. A number of smaller accelerators had been built at university institutes, but these were not modern, and co-operation with C.E.R.N. on the construction of the large machines would be of great value. In the meantime, the German Government had considered the possibility of transferring the development of nuclear science mainly to private industry, and a group of large factories had combined to build a reactor station to act as a pilot plant for future development.

The great difficulty in planning a programme for applied nuclear science in Germany had been the complete lack of trained manpower. In the field of nuclear power Germany had very few trained scientists and even fewer engineers, although the shortage of technologists was not generally serious in other industries. The universities and colleges could, in time, remedy this shortage but at present they, too, lacked trained staff, and students had to be sent abroad to gain up-to-date knowledge of developments in nuclear science.

The British Council's 21st Anniversary

The British Council, which recently celebrated its 21st anniversary, was founded in 1934 as "The British Council for relations with other countries," an organization of independent status, but responsible to the Government for the direction of its policy. The purpose of the Council is to develop closer cultural relations between the United Kingdom and the rest of the world, and part of the Council's work is to provide information about Britain's scientific and technical resources and inventions.

In his contribution to the Council's 21st Annual Report, its Director General, Sir Paul Sinker, states that the Council's main task is the making and fostering of contacts between individuals, especially those of the same profession or calling who are dealing with similar problems in their respective countries. To this end, the Council has since 1945 arranged visits for nearly 36,000 overseas visitors to Britain, mainly professional people and technicians, including many who have come under United Nations and Colombo Plan schemes. During the same period, 90 visits to various countries have been arranged for British lecturers and experts, so that they may make known Britain's progress in social services, administration, science, medicine and the arts.

Under the Commonwealth and foreign university exchange schemes, 160 awards have been made to British, Commonwealth, and other university professors, teachers, and research workers. The Council also awards over 200 scholarships each year to Colonial and foreign students, nearly thirty per cent. of the grants being for the study of scientific subjects.

Many of the Council's overseas establishments provide educational facilities, mainly for the study of English, and conduct examinations on behalf of other bodies. In this latter respect, the British Council renders invaluable assistance to the Institution by providing facilities for the Graduateship Examination to be held in those countries where currency and other restrictions might otherwise make it impossible for students to qualify for membership of the Institution.

THE EFFECT OF ATMOSPHERICS ON TUNED CIRCUITS*

by

A. G. Edwards, M.Sc., Ph.D.†

SUMMARY

The effect of an aperiodic disturbance on a tuned system is considered theoretically, and a method is discussed for determining the Fourier component of the amplitude spectrum of the disturbance at the resonant frequency of the system. The available information on the lightning discharge is then summarized, and a prediction made of the amplitude spectrum of the atmospheric radiated from the main return stroke and its variation with distance from the source. Some experimental work is described in which statistical amplitude spectra of atmospherics from far and near storms are measured by recording simultaneously on tuned and untuned channels. The results obtained are found to be in keeping with the predictions from earlier work if allowance is made for scatter in source distance.

LIST OF SYMBOLS

$\varphi(t)$	disturbance considered
$\varphi'(t)$	disturbance produced in tuned system by $\varphi(t)$
ω	angular frequency
ω_0	particular (resonant) value of ω
$A(\omega)$	complex amplitude spectrum of $\varphi(t)$
$\lvert A(\omega)\rvert^2$	energy spectrum of $\varphi(t)$
L, C, R	inductance, capacitance and resistance of a circuit
q	charge
P_1	initial amplitude of ring. Voltage resonance LCR circuit. Phase θ_1
P_2	initial amplitude of ring. Current resonance LCR circuit. Phase θ_2.
P_3	initial amplitude of ring. CR feedback circuit. Phase θ_3.
$B_1(\omega)$	response curve voltage resonance LCR B'_1 const.
$B_2(\omega)$	response curve current resonance LCR B'_2 const.
$B_3(\omega)$	response curve CR feedback, B'_3 const.
B_0	frequency independent gain factor
α	positive feedback fraction

1. Introduction

The lightning discharge involves the rapid rearrangement of earth cloud charge systems during which currents of tens of thousands of amperes may flow through discharge channels many kilometres in length. During the discharge electromagnetic energy will be radiated at considerable power over a wide frequency band, but particularly in the audio frequency range.

The disturbance of the earth's electric field which these discharges produce at points hundreds, or thousands, of kilometres distant, are known as atmospherics and have been studied for many years, although for some time the nature of their origin was not firmly established. The method adopted has usually been to record the transient changes with as little distortion as possible, and the object to establish type classifications and study the dependence of waveform on propagating conditions and distance of the source.

An increasing body of information has meanwhile become available on the lightning discharge itself, and it is now possible to associate the origin of certain types of atmospheric with certain discharge processes. In particular the atmospheric in the form of a regular oscillatory train of a few half cycles, which is observed during daylight hours, has been related to radiation from the main return stroke of the discharge as modified by propagating conditions.

The present paper is concerned with atmospherics of this type, and a study is made of their amplitude spectra. The amplitude spectrum is a simpler property of a disturbance than its wave-form, and in the case of the atmospherics' spectrum, changes with increasing distance from the source gives, immediately, information on the attenuation of the propagating system as a function of frequency in a range where man-made sources are not available.

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To this end some theoretical results are first collected and put into a form suitable for application to the present problem. An attempt is then made to predict from available information on the lightning discharge and the propagating system, the amplitude spectrum of the type of atmospheric considered as a function of distance from the source. Finally, some work on recording statistical amplitude spectra experimentally is described, and the results obtained are shown to be consistent with those predicted if due allowances is made for scatter in source distance.

2. Excitation of Tuned Circuits by Aperiodic Impulses

We represent the aperiodic impulse by $\varphi(t)$ and impose upon it the physically significant conditions—

(1) $\varphi(t) = 0$ for $t<0$ and $\varphi(t)$ is always real for $t>0$

(2) $\int_0^\infty \varphi(t)\,dt$ is convergent.

According to Fourier's theorem we may then express $\varphi(t)$ in terms of a complex amplitude spectrum $A(\omega)$ by

$$\left.\begin{aligned} \varphi(t) &= \frac{1}{\pi}\mathscr{R}\int_0^\infty A(\omega)e^{-j\omega t}\,d\omega \\ \text{where } A(\omega) &= \int_0^\infty e^{j\omega t}\varphi(t)\,dt \end{aligned}\right\} \quad (1)$$

and $\mathscr{R}$ means " the real part of ".

The sign convention adopted is that of Burch and Bloemsma[1], which is different from that of Alexander[2].

$A(\omega)$ is complex, but the energy associated with the frequency range $\omega \rightarrow \omega + \delta\omega$ is $|A(\omega)|^2 d\omega$ and this physically significant quantity is independent of the sign convention adopted. $|A(\omega)|^2$ is termed the energy spectrum of the disturbance.

By equating expressions for the total energy of the disturbance derived from its time variation and its energy spectrum we obtain

$$\int_0^\infty |\varphi(t)|^2 dt = \frac{1}{\pi}\int_0^\infty |A(\omega)|^2 d\omega \quad \ldots (2)$$

Now consider the modification of the disturbance produced by an amplifier of arbitrary response curve. The properties of any network may be represented completely (that is allowing frequency dependence of both numerical amplification and phase shift) by a suitable complex function $B(\omega)$, and if we represent the output of such a network by $\varphi'(t)$ when the input is $\varphi(t)$ we have

$$\varphi'(t) = \frac{1}{\pi}\mathscr{R}\int_0^\infty B(\omega)A(\omega)e^{-j\omega t}\,d\omega \quad \ldots (3)$$

and from (2)

$$\int_0^\infty \left[\varphi'(t)\right]^2 dt = \frac{1}{\pi}\int_0^\infty |B(\omega)|^2\,|A(\omega)|^2 d\omega \quad \ldots (4)$$

To proceed further we confine attention to the case of a highly tuned system and an aperiodic disturbance. $A(\omega)$ is then a much more slowly varying function of ω than $B(\omega)$ and we may put equation (4) in the approximate form

$$\int_0^\infty \left[\varphi'(t)\right]^2 dt = \frac{|A(\omega_0)|^2}{\pi}\int_0^\infty |B(\omega)|^2 d\omega \quad \ldots (5)$$

where ω_0 is the resonant angular frequency of the tuned system.

This equation shows that, in principle, $A(\omega_0)$ for a disturbance $\varphi(t)$ may be measured by determining the response $\varphi'(t)$ of a system highly tuned to the angular frequency ω_0 when $\varphi(t)$ is impressed upon it. In a practical case $B(\omega)$ and $\varphi'(t)$ may be known analytically, and tedious numerical integration can be avoided. Three particular cases are considered further.

2.1. *Voltage Resonance in an LCR Network*

The circulation of charge in the series combination of inductance L, capacitance C and resistance R is governed by the differential equation

$$L\frac{d^2q}{dt^2} + R\frac{dq}{dt} + \frac{q}{c} = v(t) \quad \ldots (6)$$

The expression for the gain of such a circuit when used as a voltage amplifier is

$$|B_1(\omega)| = \frac{B'_1}{\left\{\left(\frac{1}{LC} - \omega^2\right)^2 + \frac{R^2\omega^2}{L^2}\right\}^{\frac{1}{2}}}$$

where B'_1 is a dimensional constant.

Then $$\int_0^\infty |B_1(\omega)|^2 \, d\omega = \frac{\pi L^2 C B_1'^2}{2R} \quad \ldots (7)$$

The response of this circuit $\varphi'(t)$ to an aperiodic disturbance comprises a free and a forced oscillation, and when the circuit is sharply tuned the free oscillation contributes practically the whole of $\int_0^\infty |\varphi'(t)|^2 dt$. Moreover, although we do not know its initial amplitude P_1, we know that its form is :

$$\varphi'(t) = P_1 e^{-Rt/2L} \sin\left\{(1/LC - R^2/L^2)^{\frac{1}{2}} t + \theta_1\right\}$$

where θ_1 is a phase term,

so that $$\int_0^\infty |\varphi'(t)|^2 dt = \frac{LP_1^2}{2R}\left[1 - \frac{R^2C}{4L}\cos 2\theta_1 - \left(\frac{R^2C}{4L} - \frac{R^4C^2}{16L^2}\right)\sin 2\theta_1\right]$$

Since we are interested in systems having Q values of several hundred, and in this case

$$Q^2 = \frac{L}{R^2C}$$

we see that the terms in $\cos 2\theta_1$ and $\sin 2\theta_1$ may be neglected in comparison with 1 and

$$\int_0^\infty |\varphi'(t)|^2 dt = \frac{LP_1^2}{2R} \quad \ldots (8)$$

From equations (7), (8), (5)

$$A(\omega_0) = \frac{\omega P_1}{B_1'} \quad \ldots (9)$$

2.2. *Current Resonance in an LCR Network*

For current resonance we obtain an expression

$$|B_2(\omega)|^2 = \frac{B_2'^2\omega^2}{\left(\frac{1}{LC} - \omega^2\right)^2 + \left(\frac{R^2\omega^2}{L^2}\right)}$$

where B'_2 is a constant.

So that $$\int_0^\infty |B_2(\omega)|^2 d\omega = \frac{B_2'^2 \pi L}{2R} \quad \ldots (10)$$

We may describe the free oscillations of the current in the circuit by an expression of the form

$$P_2 e^{-Rt/2L} \sin\left\{\left(\frac{1}{LC} - \frac{R^2}{L^2}\right)^{\frac{1}{2}} t + \theta_2\right\}$$

As before we may write approximately

$$\int_0^\infty |\varphi'(t)|^2 dt = \frac{P_2^2 L}{2R} \quad \ldots (11)$$

and hence from equations (11), (10) and (5)

$$|A(\omega_0)| = \frac{P_2}{B_2'} \quad \ldots (12)$$

2.3. *Resonance in RC Feedback Amplifier*

Several types of selective amplifier depend upon feedback through capacitance-resistance networks. It will be demonstrated in a particular case that their behaviour is similar to that of the simple LCR network, though the analysis cannot be based upon a differential equation of obvious physical significance.

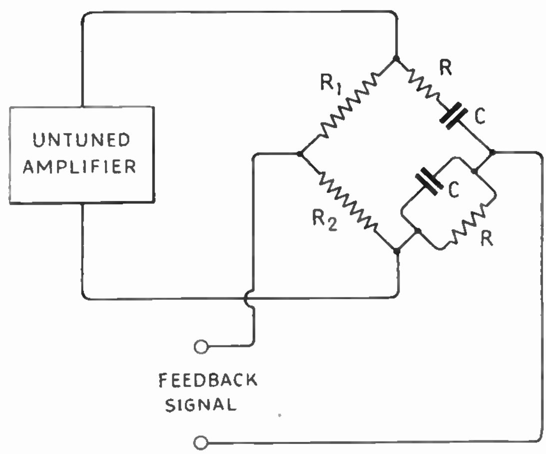

Fig. 1.—RC feedback amplifier

If the output of an untuned amplifier is connected across a Robinson frequency bridge, and the out-of-balance signal is fed back to the input, the combination constitutes a tuned amplifier, both the selectivity and resonant frequency of which can be varied by altering the bridge constants. (See Fig. 1.)

We may conveniently consider B_0 as the frequency independent numerical gain of the untuned amplifier as modified by feedback from the R_1R_2 resistor chain. The overall numerical gain $B_3(\omega)$ is then given by

$$B_3(\omega) = \frac{B_0}{1 - \alpha B_0}$$

where α is the positive feedback fraction.

$$\alpha = \frac{1}{3 + j\left(\omega CR - \dfrac{1}{\omega CR}\right)}$$

Write $\quad \omega_0 = \dfrac{1}{CR} \qquad \beta = \left(\dfrac{\omega}{\omega_0} - \dfrac{\omega_0}{\omega}\right)$

Then $\quad |B_3(\omega)|^2 = \dfrac{B_0{}^2(9 + \beta^2)}{(3 - B_0)^2 + \beta^2}$

Resonance occurs for $\beta = 0$ i.e. for $\omega = \omega_0$

For high selectivity B_0 must be slightly less than 3, and in practice this condition is obtained by varying the bridge arm ratio R_1/R_2.

We have

$$\frac{|B_3(\omega)|^2}{B_0{}^2} - 1 = \frac{B_0(6 - B_0)\omega_0{}^2\,\omega^2}{\omega^4 + \left\{(3 - B_0)^2 - 2\right\}\omega_0{}^2\omega^2 + \omega_0{}^4}$$

which is similar in form to the response curve for current resonance. In order that

$$\int_0^\infty \left\{\frac{B_3(\omega)^2}{B_0{}^2} - 1\right\} d\omega \simeq \int_0^\infty \frac{|B_3(\omega)|^2}{B_0{}^2}\, d\omega$$

the tuned stage must be used in series with one which cuts off the high frequencies and which has, in fact, a total pass band small compared with $\dfrac{9\omega_0 Q}{8\pi}$

We then have

$$\int_0^\infty \frac{|B_3(\omega)|^2}{B_0{}^2}\, d\omega = \frac{\pi B_0\omega_0(6 - B_0)}{2(3 - B_0)} \qquad \ldots (14)$$

We may show, for example, by Laplace Transform analysis, that the response of this circuit to an aperiodic disturbance consists of a free and forced oscillation, and that as before the former may be represented by

$$P_3\, e^{-\delta t} \sin(\gamma t + \theta_3)$$

where

$$\delta = \frac{(3 - B_0)}{2CR} \quad \text{and} \quad \gamma = \sqrt{\frac{(B_0 - 1)(5 - B_0)}{2CR}}$$

So that $\displaystyle\int_0^\infty |\varphi'(t)|^2\, dt = \frac{P_3{}^2 CR}{2(3 - B_0)} \qquad \ldots (15)$

Hence from (14) and (5)

$$A(\omega_0) = \frac{P_3}{3\omega_0} \quad \text{since } B_0 \simeq 3 \qquad \ldots (16)$$

2.4. *Calibration of Tuned System for Transient Disturbance Analysis*

It is apparent from the foregoing analysis, that if the Q of the system is plotted against amplification at resonance, a straight line is obtained. The gradient of this straight line Q/(amplification at resonance) divided by the resonant angular frequency of the circuit is then the ratio of the Fourier component of the amplitude spectrum to the initial amplitude of the ring.

3. The Amplitude Spectrum of the Disturbance Radiated by a Lightning Discharge

Studies of the lightning discharge have been made with moving cameras giving a time dispersion of the luminous phenomena,[3, 4] and these have been correlated with measurements of field change observed within a few kilometres of the flash.[5] In terms of this work the following physical picture emerges of the discharge between cloud and ground.

A single flash consists of one or more strokes, the relative frequency of occurrence diminishing as the number of strokes increases. The first stroke of a flash, which is somewhat different from succeeding ones, is initiated by an invisible pilot streamer which travels from cloud to ground at a speed of about 10^7 cm/sec, establishing an ionized path through the virgin air. Every few hundred microseconds a luminous dart leader travels with a speed of about 10^9 cm/sec from the cloud to the head of the pilot streamer, and the process continues until the ground is reached.

When the cloud–ground gap has been bridged by a column of ionized air, an intensely luminous discharge travels from ground to cloud, the rate of extension of the luminous channel decreasing as the cloud is approached from an initial value of about 10^{10} cm/sec. The luminosity of the channel, once established, dies away with a characteristic decay time.

After a quiescent period of some tens of milliseconds the second stroke of the flash occurs. If, as is usually the case, the ionization of the channel persists over this period, the second stroke is initiated by a single luminous dart leader travelling from cloud to ground without interruption. As before, the main stroke travels from ground to cloud immediately the dart leader reaches the ground. When the luminosity of the main stroke has decayed there is a quiescent period before subsequent strokes, if any, follow the general pattern of the second.

As a result of a lightning stroke a net charge is transferred from earth to cloud so that there is a net change produced in the earth's electric field. Following the work of Appleton and Chapman[6] and of Schonland,[5] the luminous effects can be correlated with the structure of this field change in some detail. The field change may be divided into three main parts of comparable net amplitude called by Appleton and Chapman the a, b, c portions. For the first stroke of a flash the "a" portion is of some milliseconds duration and has superimposed upon it a marked high frequency fine structure corresponding to the visual complexity of the pilot leader and dart-leader process. For second and subsequent strokes, however, this "a" portion field change is more rapid and largely free from fine structure as might be expected from the simpler visual phenomena observed.

The second or "b" portion of the electric field change is of similar amplitude to the first but takes place in less than 100 microseconds and shows little fine structure. This corresponds to the bridging of the cloud–ground gap by the main return stroke and for this component neither the visual nor the field change records show large differences between the first and subsequent strokes of a flash. The last third of the net field change occurs in a few milliseconds without marked fine structure and corresponds to the dying away of the luminosity of the main lightning channel.

This general picture of the lightning flash is now well established though marked departures from it in individual cases have been reported. One or more parts of the discharge may be missing, additional fine structure may occur on waveform or luminosity records, or disturbances may be recorded of an entirely different type, perhaps originating in intercloud discharges.

During the main return stroke, however, the current flowing reaches its maximum and the electrical moment of the system varies most rapidly with time. It is, therefore, from this portion that energy will be radiated at the greatest total power, though on particular frequency channels more disturbance may be produced by the "a" or "c" portions of the discharge. The relatively simple nature of the "b" portion variation makes it more amenable to description mathematically, and it is atmospherics radiated from this portion only which will now be considered.

We must describe the source quantitatively in terms of a current amplitude and its time variation. For this purpose we draw upon the statistical analysis of Bruce and Golde[7], who considered many of the characteristics of the lightning flash and showed the scatter in the numerical results describing them as reported in the literature.

They described the main return stroke current by

$$I = I_0\,(e^{-\alpha t} - e^{-\beta t})$$

and allotted to the constants the values

$$I_0 = 20 \text{ kiloamperes};$$
$$\alpha = 4{\cdot}4 \times 10^4 \text{ sec}^{-1};$$
$$\beta = 4{\cdot}6 \times 10^5 \text{ sec}^{-1}$$

The discharge is extending with a velocity V given by

$$V = V_0\, e^{-\gamma t}$$

where $V_0 = 8 \times 10^9$ cm/sec ; $\gamma = 3 \times 10^4$ sec^{-1}

Combining these, Bruce and Golde obtained the following expression for the time differential of the electrical moment of the system.

$$\frac{dM}{dt} = \frac{2 I_0 V_0}{\gamma}(e^{-\alpha t} - e^{-\beta t})(1 - e^{-\gamma t})$$

The electrical field variation normal to the equatorial plane of a dipole and a distance r from it in free space is given by

$$E = \underset{\text{electrostatic term}}{\frac{M}{r^3}} + \underset{\text{induction term}}{\frac{dM/dt}{cr^2}} + \underset{\text{radiation term}}{\frac{d^2M/dt^2}{c^2 r}}$$

The three components vary with different powers of r, and hence assume different relative importance at different distances, the electrostatic term dominating when r is small and the radiation term when r is large.

We can apply this equation to radiation from the lightning discharge since we have an

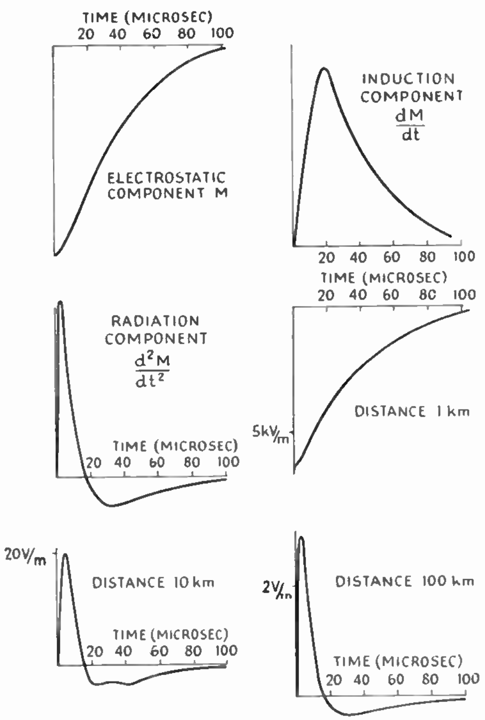

Fig. 2.—Field changes produced by a lightning discharge

analytical expression for the variation of electric moment with time.[8] The result of doing so in particular cases is shown in Fig. 2, where the evolution of the waveform with increasing distance from a purely electrostatic to a purely radiation type is demonstrated. If this evolution is compared with the experimental one published by Appleton and Chapman it is apparent that *(a)* the evolution is too rapid, i.e. the waveform becomes of pure radiation type at too small values of r and *(b)* the amplitude of the waveform decays too rapidly. Both these effects may reasonably be attributed to the difference between the actual propagating conditions and the free space and elementary dipole conditions assumed in the theory.

From the analytical expression for the field change at different distances we may calculate the amplitude spectra and in Fig. 3 this is done for the waveform records shown in Fig. 2. We see that in the radiation spectrum the maximum occurs at about 10kc/s. The analytical expression assumed may be used to predict the amplitude spectrum over any frequency range. Beyond the audio range, however, the result is of little value since the occurrence of a small amount of fine structure on the main waveform, the presence of which is not allowed for by the analytical expression adopted, will completely invalidate the prediction of the energy present at most ultrasonic frequencies.

The argument developed so far has assumed propagation in free space from a Bruce and Golde main return stroke source. The actual conditions differ in that propagation takes place between a curved conducting earth and the ionosphere through an imperfect dielectric. The conditions are thus complex, but a simple theory can readily allow for simultaneous transmission by ground ray and various sky ray paths.

It has been noted by various workers that the quasi - oscillatory waveform received under various conditions could be accounted for on just such a basis, and that from the time intervals separating successive pulses values could be obtained for both the height of the reflecting

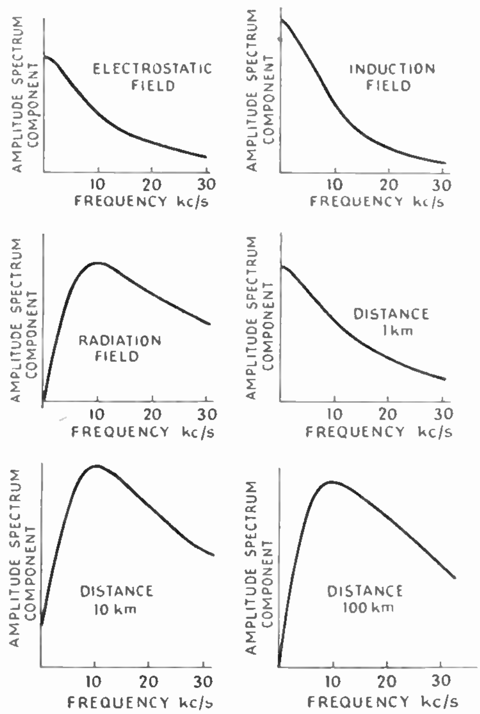

Fig. 3.—Amplitude spectra: field change from the lightning discharge

layer and the distance of the source. There is, however, some doubt as to whether phase change occurs at reflection or not[9, 10].

The energy spectrum of the resultant disturbance is calculated from that of the primary pulse as follows :—

If $\varphi(t)$ is the primary pulse

$\varphi_r(t)$ is the resultant disturbance after transmission by ground ray and sky ray paths.

$$\varphi_r(t) = \varphi(t) + R_1\varphi(t-\tau_1) + R_2\varphi(t-\tau_2) + \ldots \text{etc.}$$

where $\varphi(t)$ is repeated after an interval τ_1 at a relative magnitude R_1, after an interval τ_2 at a relative magnitude R_2 etc.

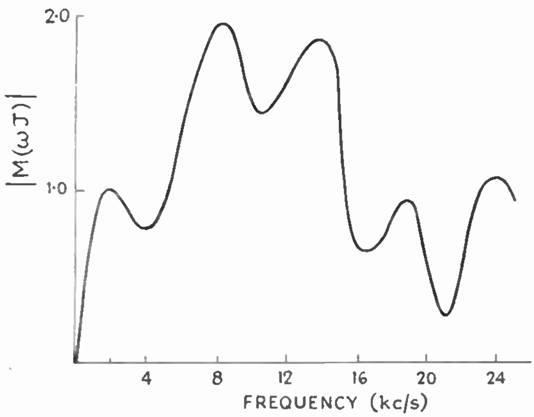

Fig. 4.—Modification of the amplitude spectrum of atmospherics by multi-path transmission

If $A(\omega)$ is the amplitude spectrum corresponding to $\varphi(t)$

$A_r(\omega)$ is the amplitude spectrum corresponding to $\varphi_r(t)$

$$A_r(\omega) = A(\omega)\left\{1 + R_1 e^{-j\omega\tau_1} + R_2 e^{-j\omega\tau_2} + \ldots\right.$$
$$= A(\omega)M(\omega\tau)$$

where $M(\omega\tau)$ may be regarded as a complex attenuation coefficient. $M(\omega\tau)$ is ponderous analytically, but it may be represented graphically in typical cases. For example, Fig. 4 shows $M(\omega\tau)$ calculated for a source distance of 750 km, a reflection height of 75 km and with $R_1 = -0{\cdot}7$ and $R_2 = -0{\cdot}4$, the negative sign corresponding to changes of phase for reflections at both the ionosphere and the earth. In practice interference effects will be further modified by differences in the frequency dependence of attenuation by ground ray and sky ray paths.

If we consider the average $M(\omega\tau)$ for atmospherics originating from distributed sources, the fine structure shown in Fig. 4 will disappear at progressively lower frequencies as source scatter increases. Whilst, therefore, the amplitude spectrum of an isolated waveform will feature as many troughs and peaks as shown in Fig. 4, a result derived from statistical analysis will, in general, give a smoother curve.

4. Some Experimental Observations

There are only one or two reports of experimental investigations into the amplitude spectrum of atmospherics.[11, 12]. Two basic methods are possible. The atmospheric may either be recorded with an untuned amplifier and the record subsequently analysed, or it may be recorded simultaneously with a number of tuned systems. In the present case the latter approach has been adopted and the results interpreted to give average spectra derived from a number of atmospherics.

4.1. *Experimental Technique*

The time of occurrence of atmospherics is not under the control of the observer, and the larger ones of interest occur in random sequence with a large number of others for which the average frequency of occurrence varies in some inverse manner with their amplitude. Any technique for photographing the atmospherics when presented on a cathode ray tube must contend with the difficulty that long exposure of a photographic film to a time-base sweep showing only background noise will nevertheless cause fogging of the entire frame by stray light.

To overcome this difficulty the tube is usually blacked out, until the leading edge of the atmospheric inself trips suitable circuits which turn on the brilliancy and apply a single sweep time base voltage to the deflecting plates. The presence of the background noise makes it necessary to prescribe a " gate " level, impulses below which do not trigger the tube. It is not difficult to make the waveform loss arising from trigger delay small compared with that lost below the " gate " level.

In the present investigation three cathode ray tubes were photographed simultaneously, one showing atmospherics after untuned amplification, and the other two recording the disturbance produced in selective receivers tuned to different frequencies in the range 1–25 kc/s and having Q values of a few hundred. These receivers were of the type discussed in the theory, based on feedback through a Robinson frequency bridge.

A separate triggering unit was used which worked directly on variations in the earth's field and was independent of gain changes on the recording channels.

4.2. *Results*

On a number of occasions it was found possible during the course of two or three hours to record several hundred atmospherics. Of these some 50 per cent were retained after examination of the untuned channel record as being of " main return stroke " type. The tuned channel records showed characteristic forced and free oscillations and from the initial amplitude of the latter the amplitude spectrum component of the disturbance at the resonant frequency of the circuit could be inferred. Throughout the period of recording twelve channels were used, two at a time, and hence an average amplitude spectrum could be derived.

There were usually 20 to 30 records showing responses on one channel. Because of this limited number a spectrum feature is not considered as real unless defined by several points.

Whilst the gross number of records taken can be increased by lowering the gate level or lengthening the time of recording, these practices are not without objection. Increasing the time is undesirable when propagating conditions and source distribution are all varying, whilst reduction of gate level increases the range from which atmospherics can be received.

Results are quoted for two cases in one of which the principal storm centre according to the Meteorological Office was about 1,000 miles distant, whilst in the other the sources were much nearer (100–200 miles) and more scattered. In both cases a family of curves was plotted showing, for each channel, the frequency of occurrence of different spectrum components as a function of their magnitude. In these plots the ordinate was the percentage of records showing spectrum components equal to or less than the corresponding abscissa expressed in volts per metre second.

If a line is drawn across such a family of curves parallel to the x axis at some value y_0 per cent it will cut them at spectrum component values $x_{\omega_1}, x_{\omega_2}, x_{\omega_3}$ µV/metre sec. The spectrum corresponding to y_0 per cent is then obtained by plotting x_{ω_1} against ω_1, x_{ω_2} against ω_2 etc. The largest records are weighted more heavily if $y_0 = 80$ per cent than if $y_0 = 40$ per cent.

Amplitude spectra obtained in this way for values of 80 per cent, 60 per cent and 40 per cent are shown in Fig. 5.

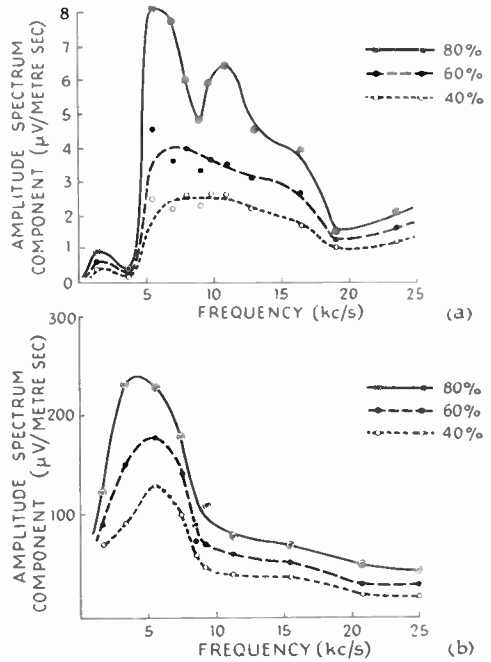

Fig. 5.—Statistical amplitude spectra

For the distant source each curve shows a trough at 3·5 kc/s which corresponds to a wavelength somewhat greater than the earth ionosphere separation. The 40 per cent curve is not otherwise markedly different from that corresponding to radiation from the Bruce and Golde flash. The 80 per cent curve shows a well defined trough about 9 kc/s which is difficult to explain except by assuming a subsidiary storm centre at a range of about 750 km for which ground ray and 2nd sky ray interference would give such an effect. (See Fig. 4.) The 6·5 kc/s trough observed by Chapman and Matthews for an atmospheric from 1000 km can be explained in a similar way.

In the case of the nearer storm centres the relatively greater importance of the lower frequencies shows the influence of electrostatic and induction components of the field change.

5. Conclusion

The experimental observations of statistical amplitude spectra of atmospherics are in general agreement with predictions from a Bruce and Golde main return stroke lightning source. The induction and electrostatic terms, however, are found to decay less rapidly, relative to the radiation term, than is the case for propagation in free space ; and where the originating storm centre is compact, ionospheric reflection superimposes marked fine structure on the amplitude spectrum of the unattenuated ground pulse.

6. Acknowledgments

The author's interest in this subject was aroused when working under Dr. F. W. Chapman, at King's College, London, where the experimental work was done. He would like to acknowledge the assistance and encouragement he received from Dr. Chapman at this time.

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ELECTRICAL PULSE COMMUNICATION SYSTEMS

3. Transmission and Reception Problems in Pulse Systems*

by

Professor R. Filipowsky, Dr.Ing. (Member)†

SUMMARY

Pulse systems offer the advantage of time-division multiplexing. Synchronous and asynchronous systems are compared and many further methods are mentioned. A novel method of reducing redundancy by functional multiplexing (CODEP) is disclosed. Special signals like synchronization pulses, channel markers, code switching signals, etc., have to be added to the combined pulse train prior to the final carrier modulation process. When comparing the various possible combinations of pulse modulation, multiplexing and carrier modulation to find the most efficient system, the limitations in power radiation have to be considered, average power or peak power limitation being the most usual specifications.

Noise and disturbance characterize the transmission medium. Passive and active interceptions may also influence the selection of an optimum system. Passive interceptions include the typical radar reflections, but also ionospheric recording, fault location, etc. Active interceptions comprise all types of repeaters. Regenerative repeating is the most important method for the elimination of cumulative disturbances over long distance communication systems.

Important problems arise at the receiver where noise and information carrying signals arrive in a closely mixed form. Apart from special detection methods there are several operations which may be adopted in the frequency domain (comb filter) and time domain (gating, integration, correlation) to improve the separation of signals from noise. Error detection and correction systems, together with prediction methods, may further improve the system quality.

Brief reference is made to the various destinations terminating any pulse communication system. Future possibilities are mentioned.

11. Multiplexing and Wave Formation

11.1. *Multiplexing Principle*

Pulse systems are most useful in multichannel arrangements. Combining several channels in such a way that they can use the same transmission medium without mutual interference is called *multiplexing* (M in Fig. 7).‡ If only two channels are involved, we speak of *diplexing* as distinct from *duplexing* (R), an operation which allows simultaneous operation over a transmission medium in two different directions.

Continuing our systematics, we may refer to Fig. 2 for general guidance. We have now to combine the output of various signal formers S, to get a so-called train T. Fig. 7 lists the detailed procedure and the various possible steps involved in the train and wave formation and transmission. In "information language," we have to form a "train space" by putting together several channel spaces. As the co-ordinates remain the same, and as normally no entropy matching is involved, we consider multiplexing a mere geometrical problem in the spaces concerned. The value of the train space cannot be smaller than the sum of the volumes of all the channel spaces. In practical systems it will be always larger. The signal cells are not perfectly limited prisms, as the elementary signals extend either in the time co-ordinate or in the frequency co-ordinate beyond the minimum limit. A certain amount of overlapping between adjacent channel spaces is therefore unavoidable. The effect is known as "*interchannel crosstalk*."

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‡ Previous parts of this paper were published in the *Journal* in September 1955 (Vol. 15, No. 9, pp. 451-467) and October 1955 (Vol. 15, No. 10, pp. 483-504). For convenience of cross reference, sections and illustrations are numbered consecutively throughout the paper, while Figs. 5 and 7 are repeated in this part.

The primary consideration in multiplex systems is to secure sufficient "crosstalk attenuation" (60 db being considered as perfect). It may cause a designer to favour either *frequency division* (*FD*) (M 1), where the channel spaces join along planes parallel to t_A-plane, or *time division* (*TD*) (M 4), with the channel spaces joining along planes parallel to the f_A-plane. Filter technique has advanced to such an extent that in FD-systems there is no difficulty in separating the channels again at the receiver, provided they are not irrevocably intermixed by cross-modulation due to non-linearities in the medium. Such cross-modulation can take place between any channels in an FD system, and its magnitude depends on the instantaneous value of the superposed signals of all channels. This, however, is the result of a statistical distribution, and occasionally there may be an extremely large peak value. The crosstalk problem is simpler in TD-systems. The peak value is never larger than the peak signal of a single channel. Crosstalk of high frequency components is only possible in the time-co-ordinate, and as such concerns merely the neighbouring channels in the time sequence, not all the channels. There is also in TD systems a possibility for crosstalk between all channels due to restricted low frequency response. This effect increases in p.w.m. with decreasing modulating frequency, whereas in p.p.m. it is independent of frequency[1]; p.a.m. is more liable to low frequency crosstalk due to non-linearities, than p.w.m. The best system is again p.p.m., if we consider the basic modulation systems only. Binary p.c.m, of course, will be perfectly free from crosstalk, as long as noise plus crosstalk components are sufficiently below the threshold of the system. P.d.m. systems are relatively immune to crosstalk, as the information is carried only by the time-difference between two pulses and not by the absolute position of a single pulse.

Any remaining crosstalk due to overlapping of adjacent channel spaces in TD systems may be removed at the receiver, by adding a fraction of the disturbing channel signals in opposite phase to the crosstalk components.[2] Special systems have been proposed to reduce bandwidth by permitting a certain amount of crosstalk, which is then removed at the receiver. (Narrow band, time division multiplex).[3]

11.2. *Multiplexing Methods*

Multiplexing is a difficult problem due to crosstalk and many other factors, but *scaling rules* may be applied when calculating the specifications of the multiplex system from the known data of the single channel system.[4]

If N identical channels are multiplexed in time-division to form an ideal system, we require as minimum conditions:

(*a*) a total bandwidth of N times the channel band-width,

(*b*) a time not larger than one N-th of the pulse repetition period as the maximum time available for any individual pulse,

(*c*) a peak power of each pulse equal to N times the peak-power in the single channel case,

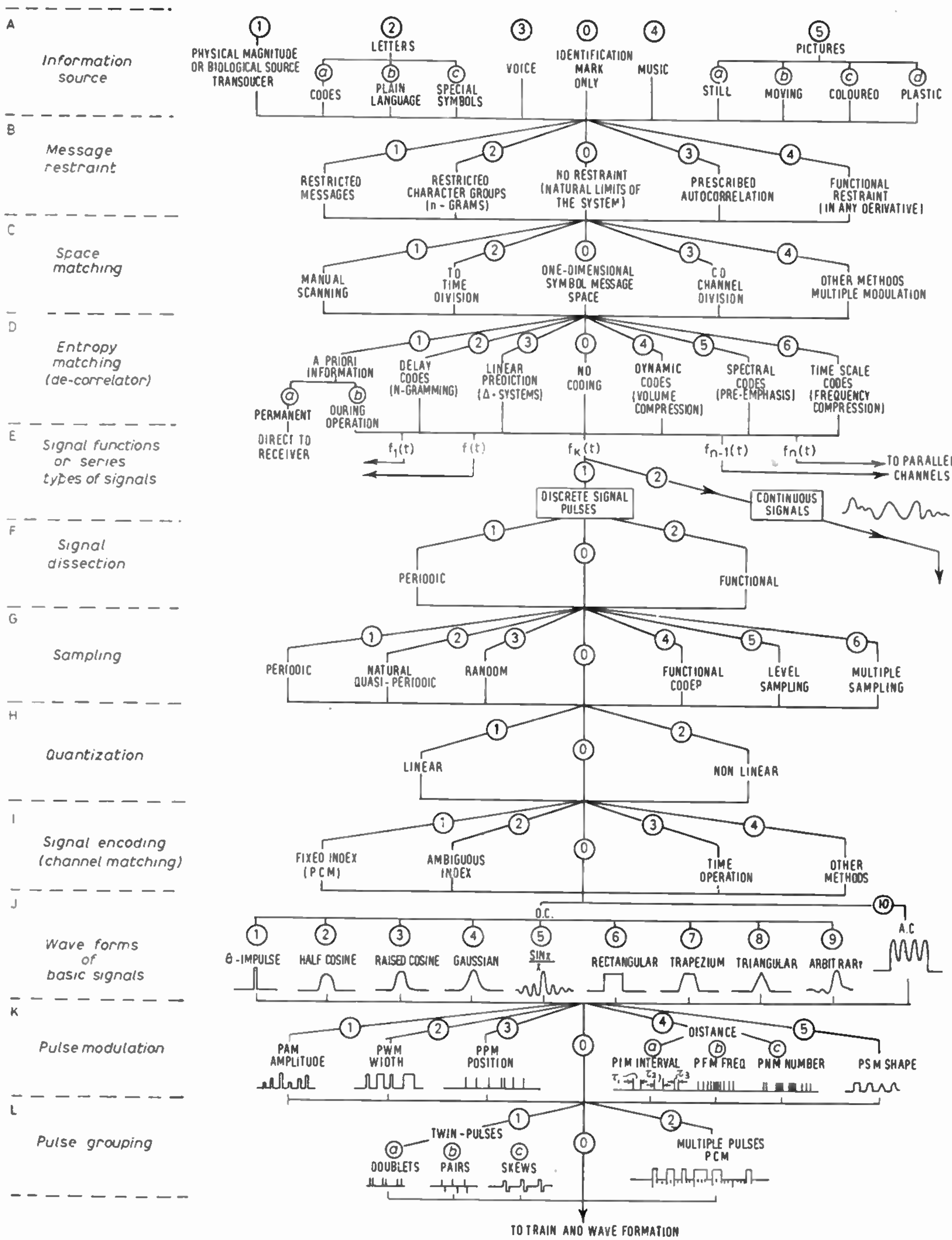
(*d*) an average transmitter power of N times the previous value.

Comparing FD methods of multiplexing with TD methods, the advantages of TD in respect to interchannel crosstalk have been stated. The great flexibility of TD systems permits addition or subtraction of channels at intermediate repeater points and the insertion of regenerative repeaters (U2, Fig 7) operating on all channels simultaneously, if all channels are quantized. Channels of widely different characteristics can easily be combined in TD. The circuit technique particularly in binary systems, does not require high accuracy. Cheap elementary circuits of perfectly identical construction (using printed circuit techniques) can be applied. A disadvantage is the need for synchronization in the conventional systems (M 4a), particularly the fact that a failure of the synchronizing part will completely upset all the channels simultaneously.

Frequency division methods (M 1a), on the other hand, are based on a well-established technique and on several decades of practical experience. Pulse systems have not entered this field, as they are inherently associated with time-division. FD is still superior to TD in systems with a large number of channels. Subgroups can operate on clearly separable sub-carriers, and filters can separate whole groups.

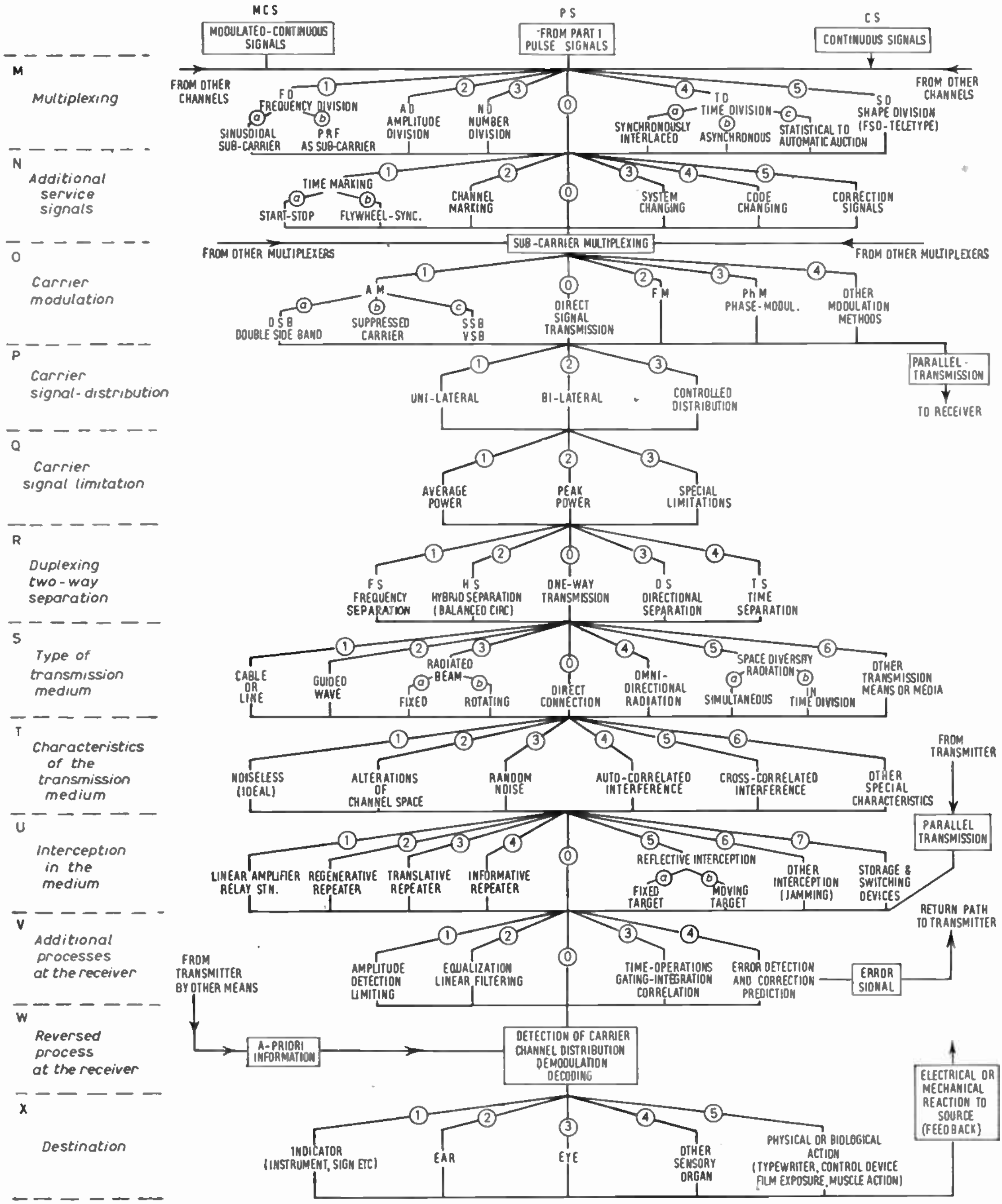
A special method of FD operating with pulses as sub-carriers, is listed under M 1b, Fig. 7 and is designated *"PRF as subcarrier."* This is primarily for mobile networks. Any one of the pulse-modulation methods may be used, but

Fig. 5.—Pulse Communication Systems. I—Message and Signal.*



* Illustration repeated from Part 2, published in October, 1955.

Fig. 7.—Pulse Communication Systems. II—Carrier and Transmission.*



each of the small number of transmitters must have a separate p.r.f. (pulse repetition frequency), which may be for speech transmission in the range of 7,000 to 12,000 c/s. All transmitters operate in the same medium and with the same radio-carrier frequency. The receivers are gate-operated and each station is tuned to one of the p.r.f.'s only, rejecting all other pulses, and such of the correctly timed pulses as are mutilated by disturbing signals from any other of the non-synchronous channels. A holding circuit keeps every sample until a new one is accepted.

Amplitude division (AD) (M 2) follows the same pattern as multiple-sampling (G 6), with the only difference that the several samples combined in one amplitude are now taken from different channels. The number of amplitude levels required in this case increases rapidly with the number of quantum steps per channel character in the individual channels. AD is therefore only useful with a practically noiseless medium (direct transmission) and with individual channels of binary to quaternary characters.[5]

Number Division (ND) (M 3) is closely related to AD and actually constitutes the more general principle. It requires quantized samples from the individual channels, each represented by one number. These n different numbers may all be limited to values smaller than m and are then combined to a new number (x), which corresponds uniquely to the series of all the individual numbers, (x_1; x_2; x_3; x_n), for example by the following equation:

$$x = x_1\, m^{n-1} + x_2\, m^{n-2} + \ldots \quad + x_{n-1}\, m + x_n$$

This single number is now transmitted by any of the known means, for example in n-ary form. If it uses a single character, we have the above mentioned AD system. If it uses binary digits, we have a method nearly identical with a TD-p.c.m. system, but with the slight advantage that no synchronization is required.[6]

Synchronous time division (M 4a) is the conventional method, where all channels have the same p.r.f. and where their individual pulses are well interlaced so that no two pulses may overlap and no free intervals remain between the epochs required for the modulated pulses. In p.a.m. the pulses may be densely packed and can follow each other at nearly Nyquist rate (2 W) if for example, six elementary signals are used (J. 5). P.w.m. and p.p.m. have to provide for sufficient time in each individual channel interval to secure the maximum required modulation percentage.

The scaling rules (see above) show that multiplexing in this form cannot reduce the total bandwidth below the sum of the individual channel bandwidths. On the contrary, the practical need for a synchronization signal increases the total bandwidth. It is interesting to note that the synchronizing pulse does not convey any information, as it is 100 per cent. predictable. An optimum system should therefore work without such an aid, and ND is a practicable system of this type.

Operational details about conventional pulse multiplex systems are excellently summarized in the four articles by J. E. Flood.[7]

An example of *asynchronous time division* (M 4b) was given in connection with random sampling (G 3). Originally intended for simultaneous operation over a given medium, the same principle may also be used for multichannel systems. The pulses are generated independently by the single channels and each channel repeats the samples at random, with certain upper and lower limits of pulse repetition rate. Naturally it cannot be avoided that samples arrive at the multiplexer simultaneously. In cases of very low traffic this may be the exception rather than the rule. The resulting occasional mutilation of samples may be allowed, and such samples may be rejected in both receivers. In cases of stronger traffic, provision may be made for instantaneous delay circuits which arrange the samples in an "orthogonal" time series by delaying some of them temporarily, so that no two of them may enter the multiplexed channels simultaneously. The delay, of course, must be a small fraction of the sampling interval.

It is evident that the receiver is not in a position to distinguish from which channel a particular sample originated. A "channel mark" (N 2) must be added to each individual sample to enable the receiver channel distributor to direct it to the right destination. The channel mark will preferably indicate the destination and not the origin. The system will then act as an automatic switching system as well. The transmitter at the beginning of an individual channel must be fitted with means of generating all possible destination marks.

Closely related to asynchronous time division, but much more efficient, is a *statistical TD-method* (M 4c). This may be termed an *automatic auction* device and is to be used in connection with functional sampling (G 4). The latter method tries to match the instantaneous sampling rate to the instantaneous complexity of the source. Multiplexing such systems causes asynchronous interlacing. This in turn requires channel marks and it is therefore immaterial at which instant a sample is transmitted, since the receiver can always direct it correctly. It thus appears logical to arrange for a central commanding circuit, which compares the instantaneous complexities of the channels to be multiplexed and selects for sampling in each transmission interval just that one of the individual channels, which has the highest complexity of all, i.e. which is most in need of transmitting a sample. This can best be done by letting each individual circuit "pay" a unit charge into a capacitor, whenever its waveform crosses a quantum limit in either direction. The "auction circuit" compares the levels of all capacitors and establishes connection to the "highest bidder," to which the next transmission interval is allocated. The capacitor is immediately discharged and the channel concerned must again start "bidding." If its complexity is high, it will raise the bids quickly and will soon be again in a position to pay the "highest rate" for a transmission interval. If its complexity is low it will take a longer time, but ultimately it will have accumulated sufficient charge to secure the highest bid, and will get its chance. Thus the valuable channel capacity is literally auctioned. The "opening bid" will be low if the traffic density is low, yet no transmission interval will remain idle. All channels will benefit equally from the larger capacity available and excellent quality will be provided for all channels. If the traffic density is high, the maximum bid will automatically be raised and, should many channels simultaneously be of high complexity, the quality may be poor. The statistical distribution of many speech channels will however render such an occurrence rare.

A practical "automatic auction circuit" is fully described in the British Patent Specification of this CODEP-system.[8] It provides for entropy matching without delay and at the same time permits the highest flexibility to the multiplexer. A new channel may be added at any time without sudden overloading of the system, merely causing a temporary decrease in quality if the number of busy channels exceeds a certain limit. Any channel may have any bandwidth within the given limits, and will automatically be supplied with sufficient channel capacity to handle its instantaneous rate of information. Thus channels of completely different specifications (telegraphy, telephony, facsimile, even television if required) may be connected to any input without discrimination. Any channel which is idle for even a fraction of a second, will be automatically disconnected, but will be instantaneously supplied with channel capacity whenever information is available for transmission. No separate switching is required for this purpose. The system can handle without discrimination any pulse modulation method.

The drawbacks of the system are of a minor nature. Additional channel capacity is required for the channel marks to be associated with with each sample. A certain amount of quantization noise (sampling noise) may be unavoidable due to the irregular sampling rate. Some special circuits must be added at the transmitting end (complexity measuring devices and automatic auction circuits) and at the receiving end (smoothing circuits, such as predictors or low-pass filters with instantaneously variable cut-off frequency). The sampling circuits should be locally combined with the multiplexer, thus excluding the possibility of automatic auction of mobile stations.

Shape division (SD) is listed as the last multiplexing method (M 5). It is applicable wherever an elementary signal has several parameters which may be modulated independently. Rectangular pulses may be modulated by one channel in their amplitude, by another channel in their width. A.c. pulses may be modulated independently by six channels in their six different parameters listed under J 10 (see also Fig. 10). The practical difficulties will be enormous, as no demodulating device will be sensitive to one modulating method alone. Crosstalk will inevitably result and costly equipment will be required to reduce it within tolerable limits. No such systems have been published so far, and no theoretical gain is claimed over more simple systems.

One interesting system however should not be ignored. It is designated "Frequency shift diplex" teletypewriter operation[9]; a.c. pulses are used to code the mark and space intervals of two channels simultaneously. Four voice frequencies (400, 800, 1,200, 1,400 c/s) represent the four possible combinations: space-space, space-mark, mark-space, mark-mark. Thus we have a system carrying the information of two channels on a single signal character. This system clearly comes under shape division.

Systems modulating the same carrier (pulse or sine-wave) by different channels in different methods are called "hybrid systems" by V. D. Landon. They also form part of this SD-group.

11.3. *Additional Service Signals*

Synchronous multiplexing systems require synchronization, which may be of two types. The *start-stop method* (N 1a), known from Baudot multiplex telegraphy and from standard television systems, involves instantaneous timing for each cycle of operation. The time marker signals (synchronization signals) should be less sensitive to disturbances and noise than the information signals. The failure of the synchronization signals will upset the whole system, whereas a failure of the information signals will effect a single channel only. *Flywheel synchronization* (N 1b), well known from television circuit technique[10] is preferable, as it is not immediately upset by the failure of a single synchronization signal.

Channel marking signals (N 2) are essential elements of non-synchronous TD systems, as stressed under M 4 b and c.

System changing signals (N 3) may occasionally be required, when specifications of the system have to be altered instantaneously, for example the p.r.f. when switching from speech to music.

Code changing signals (N 4) will be remembered as essential devices for the communication of new *a-priori* information (D 1b) to the receiver during operation. The use of these is conventional practice in telegraph or teletypewriter systems, when switching from letter symbols to figures or other symbols.

Correction signals (N 5) are required in both transmission directions. As *error signals* they may be returned from the receiver to the transmitter when an error has been traced by the help of an error detecting code. The transmitter will then use the next opportunity (a small interval in the transmission) to repeat the message part reported as a failure. For this purpose delay and storage of the information is required at both ends, and a certain correction signal must precede the repetition of the failed message part. Both the error signal and the correction signal may be required to convey a certain amount of information about the exact position of the error. In some systems the received information is instantaneously returned to the sender on a duplex path for checking. Whenever an error is traced, transmission is interrupted and repetition is inserted after a correction signal has informed the receiver.

All these service signals may convey information, if so required, by any of the modulation methods listed in Section 9.5. Because of their most important character but rare appearance, they are usually encoded in binary form (p.c.m.).

11.4. *Carrier Modulation*

In Fig. 7 carrier modulation is preceded by the symbolic specification for *sub-carrier multiplexing*. The whole process of multiplexing as specified in Sect. 9.2 can be executed in two (or more) steps, forming first small groups of channels, each group having a sinusoidal or pulsive "intermediate-subcarrier." Several such groups are then multiplexed to a common "multiplex train" which in turn modulates a final carrier to convey the total combined signal over the medium.

The final *carrier modulation process* (O) involves all the conventional methods of modulating a sinusoidal wave. *Amplitude modulation* (O 1) can assume the form of *double sideband modulation* (DSB) or *single sideband modulation* (SSB). In systems with large bandwidth and very low minimum signal frequency (Television) it may not be possible to separate the two side-bands without frequency distortions. *Vestigial side-band systems* may then replace s.s.b. and still retain most of the advantages of the latter. In all a.m. systems we may *suppress or reduce the carrier frequency* to economize output power, denying the system, however, the possibility of transmitting a d.c. component correctly. *Frequency modulation* (O 2) and *phase modulation* (O 3) offer similar advantages with continuous modulating signals.[4] The optimum choice of a carrier modulation system depends mainly on all other specifications of a pulse system and the considerations become involved where there are several modulating levels, as in a multichannel system with intermediate sub-carriers.

A useful notation to designate such multiple-

multiplex systems has been used by V. D. Landon.[11] PPM-FM-AM means for example, that several pulse-position modulated multiplex trains are each frequency-modulating an intermediate sub-carrier, and several such groups are amplitude-modulating an r.f. carrier. XX (YY) in place of any two letters indicates that any modulation system may be used. Thus PXX-YY means all TD-multiplex systems, as it is tacitly assumed that pulse-systems will use only time-division, and continuous signal systems will use only FD-multiplexing. Landon investigated all practical *combinations of modulation systems*, considering bandwidth economy, signal-to-noise ratio, cross-modulation and interference characteristics.

He indicates the following order of merit:

(*a*) *Wide-band gain* (the ability to improve signal/noise ratio, when bandwidth is increased: PAM-FM (slow); SS-PM; PAM(±)-FM; PWM-FM and PPM-AM.

(*b*) *Minimum received average signal power required for a specified signal/noise* ratio: (1) for large number of channels: PAM-FM (slow); PNM-FM; PAM-PCM-FM; PNM-AM; PAM-PCM-AM; (2) for small number of channels PPM-AM is best, if sufficient bandwidth can be provided.

(*c*) *Cross-modulation* and *interchannel cross-talk* must not appear in an "ideal" system and the extent to which they can be avoided in "practical" systems depends on the circuit complications involved.

(*d*) *Susceptibility to interference and selective fading* are equally dependent upon the skill and techniques used in making the equipment. Detailed discussion is given by Landon of several special features in favour of the one or the other group of systems.

The conclusions derived from the above investigation correspond to practical development. Apart from PAM-FM (slow), which seems to be handicapped by the lack of an electronic frequency multiplier ("time chopper") we find the first-ranking systems well established in practice. SS-PM is at least closely related to the usual practice in frequency division multiplex systems, and PPM-AM is the most frequently used pulse system. PAM(±)-FM is one of the more popular systems in telemetering and telecontrol practice. The (±) sign indicates bi-lateral modulation (P 2). Only PNM does not fill its theoretically justifiable place, probably due to practical difficulties in the associated circuit technique and to shortcomings in system characteristics, such as transient response.

One important system, favoured by C. W. Earp[12] and by E. M. Deloraine[13], who called it "time-sharing f.m." or "pulsed f.m." respectively, was not included in Landon's analysis. There is also no evidence to the author of any practical use having been made so far of this system, regardless of its theoretical super-efficiency. The pulse width, pulse height *and* pulse position are kept constant, and the carrier frequency of the r.f. pulses is varied in accordance with the instantaneous value of the modulating signal of any one of the channels. It thus constitutes in our systematics a system using p.s.m. (K 5) of the frequency of a.c. pulses, multiplexing several channels in TD but omitting any further carrier modulation. The sine-wave frequency of the a.c. pulses may be selected to fall into the proposed radio-frequency range, thus directly constituting the r.f. carrier.

This pulsed f.m. system is similar to the PAM(±)FM system, which figures high in Landon's list. The main difference is the condition of the output wave within the small intervals between the pulses. In truly pulsed f.m. there is no transmission between adjacent pulses. In PAM(±)FM there is full-power transmission of the carrier at the centre frequency of the band whenever the p.a.m. amplitude is zero, but *also* between the pulses.

One advantage of this system is the fact that adjacent pulses may be joined directly. When the intervals disappear, PAM(±)FM and pulsed f.m. become identical. Small differences may still remain in the sequence in which the operations are performed—multiplexing of p.a.m. trains first and common modulation thereafter (J 6, K 1, . . . M 4a, O 2), or p.s.m. of a.c. pulses first and multiplexing thereafter (J 10, K 1, . . . M 4a, O). A system similar to the second type, but with final amplitude modulation of a radio carrier, has been used by the author with practical success for jamming-proof symbol transmission (1943/44).

Another application of PAM-FM is the efficient frequency-shift telegraphy system, which is generally applied as a single channel system.[14, 15]

11.5. *Radiation Limitations*

Limitations are naturally imposed on the amount to which a carrier can be modulated. The elongation of the *modulating signal* (P) may be *unilateral or bilateral* in respect to zero signal value. In a.m. systems 50 per cent. rest carrier is required when bi-lateral modulation distribution is prescribed (P 2). Unilateral distribution (P 1) will be found in all systems with "keying" principle. P.p.m. or radar systems allow the well known gain in peak power, when small pulses are used in a PXM-AM system. Frequency and phase modulation systems do not offer any additional advantage when the signal is restricted to unilateral distribution.

Another *limitation* concerns the *radiated power* (Q). The design of the r.f. power output stages will be responsible for the restriction imposed on the free use of the modulator. With systems having a large ratio of pulse repetition period to pulse duration, the maximum tolerable *peak power* (Q 2) will usually be the limiting factor (radar).

In systems with closely packed pulses (multichannel p.m.-systems) the *average power* (Q 1) may be the limitation. A third possibility which may be listed under (Q 3) is peak signal limitation, either in the modulator or as maximum voltage or current limitation in the output circuit. The calculation of the optimum performance of a system requires consideration of the limitations imposed in a given case.[16]

12. The Transmission Medium

12.1. *Characteristics of the Medium*

The modulated wave or the unmodulated single or multiple channel signals now enter the transmission medium. Their information content may be affected in the medium by three factors:

(*a*) By *distortion* due to imperfect matching of the signal space to the channel space, or due to temporary alterations of these characteristics. Such alterations may be regular features of a certain medium and may affect any of the three dimensions of the channel space (T 2). Fading, for example, will affect the amplitude co-ordinate, detuning may affect the frequency co-ordinate and delay-distortion (phase distortions) affect the time-coordinate.

(*b*) By *noise or disturbances* present in the channel.

(*c*) By *interference* from other transmissions using the same medium. We speak of cross-talk, but must exclude inter-channel cross-talk between channels of the same multiplex systems (see Sect. 9.1). Crosstalk in a medium is the mutual interference of various transmissions which do not form a multiplex system, but must still be separated by similar means, as mentioned under level R in Fig. 7.

12.2. *Duplexing*

Duplexing is a means of allowing several transmissions to occupy the same medium (R). It refers particularly to two transmissions running in opposite directions, providing for a closed communication loop between two conversion partners (subscribers) allowing them to speak and reply without delay. Telephony, teletype and two-way "phono-vision" require duplexing.[17]

Frequency separation (FS) (R 1) is the usual method in radio communication. Different carrier frequencies with associated sidebands are assigned to the two opposite transmission directions, receiver and transmitter at each terminal being tuned to different radio channels. The main disadvantage of this system is the difficulty of avoiding cross-modulation between any two duplex channels when allocating frequencies for multi-station networks.

Hybrid separation (HS) (R 2) is the rule in line communications, where the incoming and the outgoing transmissions are separated by balanced networks. The large difference between transmitted and received power (140 db) makes this method inapplicable to radio transmission.

Directional separation (DS) (R 3) takes the place of hybrid separation in continuous microwave transmission for point-to-point radio relay links, in addition to the more usual frequency-separation method. In the centimetre wave range it is easy to achieve highly directional radiation and beaming is a convenient method of concentrating the radiated energy towards the receiver.

Time separation (S) (R 4) is the normal procedure in manual radio telegraphy where the opposite station has to listen until the sender gives an "over" signal, to indicate that he is now waiting for the other side to speak. Both stations must exchange receiver and transmitter connections to the aerial, and then the duplex route is established. This provides an easy method of duplexing, with only one channel required for both directions. Perfect separation is automatically secured. The disadvantage is the lack of "break-in" facilities. The latter may be provided by voice operated relays or more efficient methods. If full duplex operation (independent transmissions in both directions) is required, we may employ *sub-audio switching* or *supersonic switching*. The former method involves necessarily heavy distortions and requires shortest transmission time between the two stations. Nevertheless some authors claim good speech intelligibility with such systems.[18, 19] Supersonic switching involves extension of the channel bandwidth to at least twice its normal value, and automatic delay adjusting for mobile systems. (A distance of 150 km between receiver and transmitter results in a delay of 1 m sec between stations.) To avoid delay matching, asynchronous supersonic switching has been experimentally tested. (P.p.m. system with 1·5 μsec. pulse length and 10 kc/s p.r.f.; the receivers keep the last sample stored and are blocked during the radiation of the local transmitter.) The system is of particular use in a radio network.[17]

12.3. *Types of Transmission Media*

The various types of transmission media (S) range from a directly closed loop (calculators, servo-systems) (SD) to water and earth as very special cases (S 6).

Lines and cables (S 1) are the oldest media, and pulse transmissions are rarely conveyed by these means. If they are employed, good transient response (small phase distortion) must be secured.

Guided waves (S 2) over transmission lines or wave guides may easily act as carriers for pulse communications. The problems are similar to those of ordinary continuous wave transmission.

Beam radiations between fixed points (S 3a) are the common medium in radio relay links, whereas rotating beams (S 3b) are frequently used for radio location in homing beacons. Surveillance radar, instrument landing systems (ILS), early warning radar and many other location systems use specially shaped beams for scanning a determined area by e.m. waves.

The above systems may use the gain of a directional antenna system to increase the signal-to-noise ratio at the receiver, or to reduce the transmitter power when compared with an *omnidirectional radiation system* (S 4). The latter, however, generally requires all-round radiation only in the horizontal plane. The vertical radiation diagram may have maximum radiation in the horizontal plane and little radiation under any larger elevation.

Space diversity radiation (S 5) with *simultaneous* transmission (S 5a) from two transmitters located at a certain distance (base line) is well known from continuous wave "mapping" location systems. If the two or more transmitters operate in *time-division* (S 5b), they may form part of pulse location systems (Loran). Space diversity radiation is sometimes also used in long distance short-wave transmissions; space diversity reception is however usual in overseas communication systems.

12.4. *Noise and Interference*

Noise (T 3) is the most important characteristic of a transmission channel to be specified. It has been mentioned that the improved protection against noise attainable with certain pulse-communication systems is the most important argument in their favour. Naturally, the system engineer must familiarize himself with all aspects of the noise problem. Out of the vast literature available we may mention the papers by Rice[20], Jelonek[21], Macdonald[22], and Middleton[23], which are particularly relevant to pulse communication systems.

Noise, in the restricted sense, involves all disturbances in the medium and channel due to unavoidable natural sources (thermal agitation, fluctuation etc.). In a wider sense noise is understood to comprise all signal components appearing in the output of the receiver which were not present at the input of the channel. This is the definition preferred in general information theory. Here, however, the first meaning seems more appropriate and any additional components may be termed disturbances or interference.

Auto-correlated interference (T 4) originates from the same source as the wanted signal. It may, for example, be an echo due to mismatch in lines, or to reflection in the medium. Strong auto-correlation may render the discrimination of interference from the original signal difficult, but may on the other hand offer the possibility of recombining echos with the original by delay circuits, thus increasing the signal-to-noise ratio.

Cross-correlated interference (T 5) originates from similar sources as the wanted signal, but will not be as closely related to it as auto-correlated interference. Crosstalk from neighbouring channels or from other services sharing the medium is the usual cause of this type of interference. Re-combination with the original is now impossible and discrimination is difficult due to the closely related nature of signal and disturbances. Such interference is the worst and must be reduced as much as possible at the sources, i.e. all possibilities for crosstalk must be traced, and their influence balanced out.

12.5. *Passive Interception within the Medium*

Any major alteration of the wave (or train) containing the information signals may be termed interception in the medium. Such interception may be passive, for example merely the reflection of the wave (alteration of the propagation direction), or active, for example the complete absorption of the wave and its improved (amplified) reproduction.

The former is more important for location systems, the latter for point-to-point communication systems.

Reflective interception (U 5) is of a passive nature. The reflecting object may be fixed in space (U 5a) or moving (U 5b). In the latter case it may be desirable to distinguish it from nearby fixed objects (M.T.I.-moving target indication). Radar systems are based on this type of interception.

12.6. *Repeatering*

(*a*) *Linear repeaters* (*amplifiers*) (U 1) are widely used on lines and cables. With the invention of the first telegraph, the desire arose to extend the range of operation by repeating (regenerating) the signal at intermediate points, whenever excessive deterioration had been caused by the unavoidable attenuation and distortion along the line. Non-linear devices such as telegraph relays were the first solution, and electron tubes were primarily invented as a means of increasing the speed of such relays and of linearizing their operation for use with telephony signals. Modern telegraphic or telephonic repeaters have reached such perfection that they are to be submerged in the ocean with the first transatlantic telephone cable at present under construction in U.K., U.S.A. and Canada. Linear repeaters for RD-multiplex systems have to meet more stringent design specifications regarding low harmonic distortions (Sect. 9.1). Linear repeatering over long distances, involving several hundreds of repeaters, has the drawback of accumulating noise, distortions and interference without any chance of removing them again from the useful signals.

(*b*) *Regenerative repeaters* (U 2) are those which do not merely amplify the incoming signals, but restore their original waveform, which of course has to be standardized and must be identified by *a-priori* information to the repeater. Quantization is unavoidable and signal-to-noise ratio at the input of the repeater should be 12 to 16 db above the minimum calculated from the formula for the ideal channel capacity, which is based on the average signal and noise power. Due to the non-linear repeater action, incorrect signals of perfect shape may be produced whenever an incidental noise peak reaches into the adjacent quantum level. Thus the repeater occasionally increases the effect of disturbances instead of removing it. To reduce this error probability regenerative repeaters must be inserted at more frequent intervals than linear ones, in order to approach the possibility of perfectly noiseless transmission. Black[24] gives a table of the required "safety margin" to secure a sufficiently low error probability. For 1 in 10^{18} average relative frequency of error we require 14·1 db signal power in excess of the theoretical non-surpassable ideal.

Regenerative repeaters may have the simplest circuits when used for binary repeatering and this is the main reason for the high importance of p.c.m. systems.

(*c*) Translative repeaters (U 3) are becoming increasingly important with the establishment of more internationally interconnected networks and the rapid growth of automatization of telecommunication traffic. There are at present two problems of the greatest interest:

(1) *Code translators* for converting telegraphic signals encoded in one telegraphic code (morse code for example) into signals using another code (5 or 7 digit cable code).[25]

(2) *Television standard converters* to enable the interconnection of television networks using different standards.[26]

(*d*) *Informative repeaters* (U 4) forward different or additional information in the output signals. Beacon-responders, i.f.f. systems, and responding range measuring systems are some examples. Other applications may come in trunk telephony systems using "marked samples" (see M 4c), where individual samples may reach the destination on different routes through a cross-connected national network. On arrival they must be rearranged in their original sequence. Any sample using a longer by-pass route must be impressed with a mark on special repeaters on this route, specifying the delay involved in using the by-pass. A similar procedure is followed in digital computers, where special signals are added to encoded intermediate results, marking the calculating method employed or the process to be followed.

(*e*) *Jamming* (U 6) is another form of intentional interception which may impose particularly difficult design requirements on the system engineer trying to establish "jamming proof" communication systems. The approach will be a gradual one according to the amount and type of jamming to be expected. C.w. jamming causes severe beat sounds in a.m., but may be ineffective against keyed pulse transmissions. If positive jamming is likely, special pulse waveforms must be selected to allow an efficient discrimination at the receiver. The severest form of jamming is done by random a.c. pulses keying an f.m. carrier. All parameters of the a.c. pulses, particularly their "voice-frequency," may be changed at random. In such a case the problem is a statistical one. There is definitely no possibility of error-less discrimination. Whatever signal waveform we may select, the jammer will occasionally imitate it. The answer is an error-detecting cum error-correcting system.

The climax of jamming skill will be an automatically operating system tracer, which will observe the transmission, analyse it, and set the jamming device to produce instantaneously the opposite transmission. This may be answered by "hide and seek"—by increasing the redundancy, making the jammer tap without knowing whether a particular feature of the transmission is carrying information or is just "mimicry," or by changing frequently, and if possible at random, the system, the message code and signal code. In the latter case, system changing signals (N 3), code changing signals (N 4), and correction signals (N 5), will be needed to inform the receiver of the changes which took place at the transmitter. Parallel transmission or repeated transmission may secure a final escape from any form of jamming, but the equipment will by this stage have become of intolerable size and weight. All the skill of the system engineer is required to compromise between such diverging needs.

12.7. *Storage Devices*

Storage devices form an essential part in many pulse communication systems. They are required for most coding processes and for all types of error detection and correction systems. Apart from such applications, where storage is part of the normal system operation, there are many uses for long time storage by means such as disc and tape recording. Electrostatic or ultrasonic memories are for shorter delay times, and delay lines cover the range of extremely short times. All these devices are special embodiments concerning the equipment engineer rather than the system engineer. Full information on storage means may be collected from the papers by Hollander[27] and Schröter[28].

13. Reception of Pulse Signals

13.1. *The Work of N. Wiener*[29]

Wiener's work has so far been mentioned twice in this paper: as the theoretical basis of statistical theory of communications as a whole (Sect. 2.1), and as the main contribution to the specific problem of correlation and prediction in electrical communications (Sect. 5.7). It found practical application in the evaluation of the characteristics of sources and in message encoding.

Prediction was mentioned as an important tool for reducing redundancy by transmitting an error signal in place of the actual instantaneous symbol (Sect. 8.5). At that stage we are absolutely certain about the symbols produced and selected by the source of information. Uncertainty exists only about the future; there is a probability distribution for the choice of the next and the further symbols, but past and present are certain.

This is not so at the receiver: here nothing is certain, neither the past nor the instantaneously received signal, and certainly not the future of the signal. But in any useful communication system we require a rather high probability for the received symbol being actually the one which has been transmitted. Statistical theory is essential when dealing with systems for further reduction of the error occurrence. Two methods have been indicated by Wiener: filtering and prediction. The first one attempts to separate the signal from additional disturbances. We have only the mixture available, additively or otherwise, of the transmitted time series (signal) with other time series (noise and disturbances). It is intended to ascertain what the message would have been without the contamination by the other time series. The second method may be attempted in addition to the first or separately from it. We wish to know what the uncontaminated time series will do in the future, but we are also interested in finding what the uncontaminated time series has done in the past.

We spoke of the channel space containing the message in three-dimensional form as having the three co-ordinates amplitude, frequency and time. Special operations may be performed in each of them at the receiver with the aim of improving the detectability of signals in the presence of noise.[30, 31]

13.2. *Amplitude Detection Methods*

Operations in the amplitude domain are particularly useful with any system of time-modulation (including f.m.). *Limiters* are used to cut-off the noise and to pass only the signal, which must be larger than the noise peaks (Sect. 12.6b). When the average signal power is less than 13 db above the average noise power, disturbances "break through" the limiter to an increasing amount. Systems operating below this "*improvement threshold*" cannot make full use of the theoretical possibility of exchanging bandwidth for noise improvement. Comparing the theoretical characteristics of several transmission systems when operating in a noisy channel requires full consideration of their respective improvement thresholds.[32]

"*Single pulse detection*" may serve as the classical standard for comparing several detector and limiter combinations. Statistical operations cannot help in this case, as a single pulse has neither past nor future nor any statistical symbol distribution if it is a binary digit only. A single pulse may be recognised against a noisy background by "*linear detection,*" the usual method of envelope rectification, or by "*coherent detection.*"[33] The latter system makes use of the fact that even a single pulse, when appearing as the envelope of a radio frequency carrier, has several parameters, which may be useful as an indication of its presence. The most important ones are the frequency *and* phase of the carrier. Whereas the linear detector recognizes a signal only by the increase of amplitude above noise, the coherent detector compares frequency and phase of the incoming signal with a local standard: *Homodyne* and *Synchrodyne* detectors, two approximations to the coherent detector principle, derive the local oscillation from the incoming noisy signal. N. F. Barber[34] and D. G. Tucker[35] investigated a truly coherent detector system, branching the input into two parallel paths using separate modulators in each branch. The two local oscillators operate synchronously but in phase quadrature and are not affected by the incoming noise. Neither exact synchronization of the local oscillator with the carrier frequency of the incoming signal nor exact phase stability is required in such a system, yet we may retain the usual advantages of a coherent detector.

"*Periodic pulse train detection*" is not fundamentally different, but may be made more efficient by additional operations in the frequency and/or time domain.

13.3. *Operations in the Frequency Domain*

Since the first days of radio, linear filters have been used to select the wanted signal from other similar signals and from disturbances or noise. They cut out those parts from the signal space which are most likely used by the wanted signal and allow them to pass to the detector. Parts of the signal space which are less likely to be used by the wanted signal, but which may certainly be occupied by noise energy are prevented from reaching the detector.

Single pulses are characterized by continuous spectra, and the corresponding part of the signal space (channel space) to be cut out by the filter will be a continuous space. The problem is to find the "*optimum filter*" which will lift the signal in its output to the highest level above the noise. In 1943 North developed a theory[36] for the design of filters which produce the highest signal-to-noise ratio at the detector, when used in predetection stages. He found that in the case of white, additive Gaussian noise the filter response should be the conjugate of the Fourier transform of the pulse-signal to be detected. These results are embodied in "matched" filters.[37] North's theory has recently been extended to the case of non-white noise and finite observation time.[38] These investigations are based on average values of signal and noise. Wherever a receiving threshold is applied (non-linear problem), it is more important to use "the probability that the noise component of the receiver exceeds a preassigned tolerance value at least once during some finite period of observation" as joint criteria for the optimization of the filter response.[39]

Periodic pulses are characterized by line spectra and thus offer a striking opportunity to use "comb filters" for their separation from noise. Such filters are multiple band-pass filters which pass the p.r.f. and all its multiples in separate pass-bands, but attenuate all frequencies between these selected narrow bands[40] (Fig. 11). These filters, of course, may be used in post-detection circuits. They dissect the signal space into many small slices parallel to the time-amplitude plane, the p.r.f. and its harmonic being the parameters.

A similar device is the *String Filter* proposed and demonstrated by F. Vilbig.[41]

13.4. *Operations in the Time Domain*

Both the signal and the noise are represented by time series or time functions. Operations in the time domain will naturally be based on the statistical difference between these two series. Such operations require a sufficiently high level and are usually applied in post-detection stages. There the waveform will be limited to a highest frequency equal to the bandwidth (W) of the channel, with WT samples describing the wave-form completely over an epoch T. Taking these samples at the receiver, the problem remains as whether a particular sample originated from the time series of signal plus noise or from the time series of noise only.

Two statistical errors are possible: noise may be misinterpreted as a signal (false alarm) or a signal may be mistaken for noise (non-detection). The best system is one which in the long run will stabilize the probability of false alarm and will minimize the probability of non-detection.[42]

The analogous operation to the comb filter, exercised in the time domain is the *periodic gating* process. In cases where periodic pulse trains are specified by a small ratio of pulse width to pulse period it is practical to cut out all epochs from the signal space which do not contain the signal. This gives slices parallel to the frequency-amplitude plane at fixed time intervals, acting as the parameters. Many pulse systems, particularly binary systems, are rendered practically "self-gating" by the employment of double-sided slicers (limiters) in the amplitude domain (Sect. 13.2). Gating processes may also be non-periodic. The gate is then actuated by an "opening signal", preceding the actual pulse. As the opening signal does not carry any other information, except the order "to open the gate", it may be of the simplest nature, say a binary digit. Its function as "door opener" may, however, be combined with a "key" task, allowing it to open only one particular gate out of several available in the channel. It then becomes additionally a "marker pulse" as discussed in Sect. 11.3.

In systems with inconstant p.r.f. (level G 3, 4, 5 in Fig. 5 and M 4b, c in Fig. 7) it is advisable to "hold" each sample until the next has arrived and is accepted as genuine information (error detecting system). Even in error detecting systems with constant p.r.f. *holding circuits* are highly useful to reduce the effect of "lost" samples.

Integration of successive pulses is an efficient method of improving the signal-to-noise ratio in cases where the pulses are not individually modulated (radar). The signal power increases

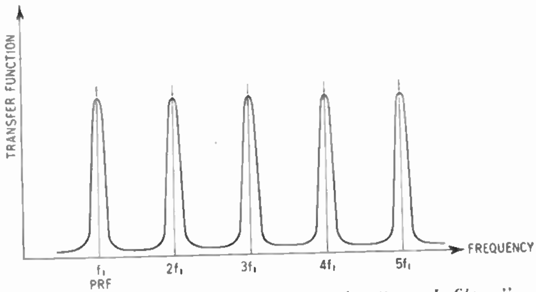


Fig. 11.—Frequency response of a "comb filter."

linearly with the number of pulses involved in the process, whereas the noise power increases only with the square root of the number of pulses.[31] Practical devices operating as such "comb filters in the time domain" have been proposed by N. Wiener and Y. W. Lee on the one hand and by G. D. Robertson on the other (feedback filter using a delay line).[43]

Correlation detectors (non-linear filters) consist of a bank of tuned circuits and linear detectors, each followed by a threshold trigger device (blocking oscillator or multivibrator). These *LC* circuits (or narrow band-pass filters) are tuned to the range of possible frequencies of the incoming signal.[42] The system would be identical with a comb-filter were it not for the non-linear trigger devices. The correlation detector is therefore a combination of operations in the frequency and amplitude domain. Due to the time delay necessarily or purposely involved, their combined effect results in an operation in the time domain. It may best be considered as a statistical operation in the whole channel (signal) space.

The *exhaustion method* is borrowed from astronomy and meteorology and involves optical storage. W. Meyer Eppler[44] demonstrated results of three different display procedures, when using an inexpensive cathode-ray oscilloscope in connection with this method. It is similar to the two previous methods, and is mainly applicable to periodic pulse trains which are either unmodulated or carry only extremely low modulating frequencies (radar). It has the advantage of simplicity, but optical storage may be a severe drawback wherever instantaneous evaluation is required.

Crosstalk compensation at the receiving end is the practical method of realizing the theoretical possibility of reducing the bandwidth in order to improve the signal-to-noise ratio. The extreme case of a noiseless channel should theoretically permit the transmission of a given amount of information in a given time with nearly zero bandwidth. Such an attempt would immediately extend all signals along the time-scale. Maintaining the same rate of transmitting the symbols will unavoidably result in overlapping characters in the time domain. MacColl proposed[45] a scheme which adds to each symbol character at the receiving end a delay of one character epoch in opposite phase, thus cancelling the "after effects" of each character.[46]

13.5. *Pulse Demodulation*

After pre-detection filtering, carrier detection, and post-detection operations on the channel space, all that is possible to separate the wanted signal from noise disturbances may have been done.

The remaining steps in the total communication system will generally be the reversed operations of the transmitting end.

The purpose of pulse demodulation is to reconvert the channel space from its modulated form, in which it was suitable for multiplex transmission, to its basic form of a single channel space.

Special circuits are required for each pulse modulation system such as pulse-amplitude demodulators, pulse-width demodulators, pulse position demodulators, etc.

Pulse height analysers are essentially p.a.m. demodulators, frequently combined with switching and counting circuits (for example in coincidence circuits). They have a huge field of application in observation systems, registering natural pulse trains, for example, those emerging from cosmic ray cloud chambers.

Pulse width discriminators have similar important applications outside the normal demodulation methods.

13.6. *Signal Decoding*

Signal decoding is an essential operation to restore the signal space from the channel space, if signal encoding is used at the transmitter.

Error detection will normally be inserted after carrier detection, but may be inserted either before or after decoding and demodulation. It will be recalled that error detection systems require a certain redundancy (extra channel capacity) to allow the insertion of characteristics into the signal, train or wave, which make it dffiicult for disturbances to imitate the actual information or its symbols, but at the same time allow the receiver to recognise mutilated symbols as such and to reject them rather than accept wrong symbols. The full *a-priori* information available to the receiver may be used for error detection. Broadly speaking all the operations discussed in Section 13.2-.4 may be considered as part of the error detection procedure, as they use the *a-priori* information about the carrier, wave-modulation system, spectra of the signals and their statistics, to reduce the error probability. This can be further reduced by considering in detail the individual probability of certain channel symbols being imitated or mutilated by noise or disturbances of given statistical structure. All such operations may be carried out without purposely introducing additional redundancy at the transmitter.

The most efficient error detecting methods however, intentionally introduce special new characteristics, thus increasing redundancy, but reducing equivocation (Sect. 2.7). So far only discrete signals of binary type have been dealt with in the form of a general theory.[47] "Single error detection" is done by one additional "dummy digit", added to any number n of ordinary bits, carrying the information. To reduce the probability of occurrence of a second error in the same group of n bits, the number n should not be too large. In a binary p.c.m. system each character of say seven bits (character elements) may be taken as one group ($n=7$), and an eighth bit may be added as an error-detecting dummy. At the transmitter all the bits of one character which are in a particular position, say position 1, are counted. If this number is even, the dummy is left in zero position. If the number is odd, the dummy is given position 1, thus making the total number of 1's in any case an even number (even parity check). This rule is known to the receiver as *a-priori* information and a mere counting of the 1's in each group will permit the detection of a single error, as any error either reduces or increases the number of 1's in a group making it an odd number. A second error will make the check number even again, so that two consecutive errors will remain undetected in this simple system. More complicated systems for multiple error detection on a counting basis may be envisaged.

Error detection is useful, as it will warn us against accepting a wrong symbol. In any communication systems we can afford to reject that sample and wait for the next. If a holding system is used to keep the last correctly received sample until a new one is accepted, the disturbance experienced due to the missing sample is generally small compared with the effect likely to be produced by an erroneous sample. However, in teletype service and still more in computers, we cannot afford to lose a single sample.

Error correction methods are essential in this case. They may be of the repetitive type or of the self-correcting type. The first method is as old as human communication. If the received message does not make sense, the repetition of all or part of it is requested. An error signal conveys this request over the return channel to the transmitter which then repeats the doubtful part, taking it from some storage device which regularly holds any transmitted message until it is accepted as correct by the receiver. *Self-correcting methods* are preferable, particularly when there is no need for a return channel under normal service conditions. They require however much more redundancy. The fundamental ideal is to add k dummy pulses to each group of m information pulses, and to use them for a series of parity checks over certain selected character elements. The result of a check will be marked 0 if parity is certified. It will be marked 1 if the check fails and traces an error. R. W. Hamming disclosed a key for the selection of elements for each check. If all checks prove parity, the signal is correct. If there is a single error, some or all of the checks will fail. Their corresponding binary number of the 0's and 1's (for correctness or failure of a check) will directly indicate the position of the erroneous bit, which may then be corrected. The number of required dummy bits (k) is relatively large for a small number $n=m+k$ of all bits together. The maximum number of useful bits (m) may be calculated from:

$$2^m \leq \frac{2^n}{n+1}$$

For an error counting code with 7 information handling bits ($m=7$), four dummy bits are required, or a total of 11 bits.

Adding one more dummy bit will make it possible to correct a single error (as above) and in addition to detect a second error.

Similar codes are known in telegraphic systems as "*protected codes*."[48] The six-unit code is a single error detection code using the international alphabet No. 2 for the first five digits and having the sixth digit as dummy for a parity check. Three types of seven unit codes are listed, which offer better protection than the six unit code. They are followed by a discussion of an eight unit code. A ten-unit code is based on easy conversion from the five unit code,[49] each element of the latter being immediately followed by an element of reverse sign. None of these codes is self-correcting as stipulated by Hamming. One reason for this is most probably the fact that at least four dummies are required to make a five unit code self-correcting. Thus only a nine-or ten unit code could use this advantage. Other disadvantages may be the complicated correcting mechanism required and the difficulties encountered in translating a standard 5 unit code into a 9 or 10 unit self-correcting code. Repetitive error correcting methods, on the other hand, are frequently used in telegraphic practice (automatic RQ/BQ signals, automatic ZBY and Verdan or full repetition systems).

Prediction methods at the receiving end are necessarily required when some sort of delta-message encoding (D 3) is used. The error signal only is transmitted but the basic value, which must be corrected by the error signal, may be derived by (linear) prediction. The error signal must be executed in identical form at the transmitter and the receiver in accordance with the available *a-priori* instructions. Prediction could be combined with error detecting and correcting systems, but equipment cost is likely to limit such methods to simpler types.

13.7. *Channel Separation*

Channel separation and distribution of the signals in a multichannel system are the reverse operations to the multiplexing action at the transmitter. In asynchronous TD systems some channel mark detecting device is required, which will operate a switch for directing each sample to the correct channel.

14. Destination

Full knowledge of the "*resolving power*" of the destination in each co-ordinate (domain) of the information space is a paramount requirement for the designer of an efficient communication system.

Transducers are required to perform the reverse operation of the input transducers in use at the transmitter. They make the transmitter information available in the required physical form. It has been stressed that the purpose of several communication systems (sensory prosthesis) is the reproduction of information in another physical stage than that in which it was picked up. Simultaneous evaluation is occasionally preferable in different forms, for example instrumental indication with oscillographic facilities.

The human observer is an important factor in most communication systems. He is contrasted to the "ideal observer," who always will make the best possible use of a series of available observations, when deciding for example whether or not a signal is present in a noisy background.[50, 51] The efficiency of the human observer depends on psychological and physiological factors which reduce it below that of the ideal observer, but which on the other hand permit the toleration of certain shortcomings of a system designed for human observers only.

The physiological factors concern primarily the ear and the eye, as the most important senses for the perception of information. The psychological factor is prevalent when evaluating the pragmatic aspect of the information content, where mental associations, memory, education, etc. are of importance. This does not directly concern the communication engineer, but has at least some bearing on the conduct of *speech intelligibility* tests.[52]

The *human ear* as the ultimate component has been investigated by many scientific workers in divergent fields. Recent results have been summarised by Huggins[53] and Fletcher.[52]

Television systems including recent colour systems are entirely based on the shortcomings of the human eye (compare for example: Trendelenburg: Der Gesichtssinn, Berlin 1943). *Visual displays* for the received information must match their characteristics (brightness, colour, persistence, repetition frequency, contrast, etc.) to the requirements of the eye.[54]

15. Complete Communication Systems

In Section 1 an attempt was made to classify all pulse communication systems broadly into four major groups. Further sections listed all known specifications and operations associated with such systems. Naturally, many of them may be omitted from any particular system. The practical importance of certain system characteristics will now be considered, and the present and probable future applications of pulse communication systems will be surveyed.

Location systems are almost exclusively pulse-operated. C.w-radar systems cannot successfully compete with the accuracy of pulse systems in time measuring devices. Their high efficiency, due to the high peak power obtainable with small average power, is another advantage of pulsed radar and other pulsed location systems. Only d.f.-systems rely on c.w.-operation.

Computing systems offer well-balanced ranges of application to the two basic computer methods. *Analogue computers* (continuous signals) are preferred in small scale units of restricted accuracy, but the *digital* type (pulsive operation) is now being developed as small units. Most of the additional operations such as error detection and correction are only applicable in pulse systems.

Conventional communication systems are taking gradual advantage of the many new ideas for efficient information transmission, as expressed in the previous paragraphs. *Teletype-transmissions* show an increasing use of certain types of error detecting devices. Error correction is so far restricted to the repetitive system, but self-correcting methods may follow as soon as they have proved their practicability under service conditions.

Multichannel telephony still favours the well established FD-systems as used for wire and cable transmission, but the TD-systems, first introduced during the last war for certain field applications, are likely to predominate in the future. Their potential advantages are: relative freedom from non-linear distortions, possibility of regenerative repeatering, ease of coding procedures including error correction, and certain practical advantages for branching off single channels at intermediate points in a network. At present it appears that the need for synchronization and for elaborate switching devices handling multiplexing and channel distribution stands in the way of a speedy introduction of pulse systems in telephony. Possibilities of avoiding synchronization and at the same time of gaining the advantage of reducing redundancy by statistical TD multiplexing have been discussed in Section 11.2 under M 4c. It is likely that such methods will in time offer all the striking advantages of pulse systems without their present shortcomings.

Some enthusiasts may imagine a *future ideal trunking network*, where each sample of a message will consist of three groups of binary digits, the first being a "destination mark," giving the call number of the wanted subscriber, the second being the quantized and coded amplitude value of the sample, and the third group being an assembly of dummy digits for error detection and correction. Such a sample will switch its own way through the network and whenever it may be delayed by waiting a few milliseconds to find a free channel or when it has to run over a longer by-pass route, further digits may be added to indicate the total delay. The receiver has to rearrange the samples prior to encoding and demodulation according to their individual delay-time. The increased duration of each sample due to destination mark, error dummies and delay mark may be easily tolerated on account of the enormous saving in idle channel-time. No operators are required, nor time for building up connections; the channel is not held busy throughout the ringing time, nor when the required person is called to the telephone. Samples are only produced at the minimum instantaneous rate required for the reproduction of a small epoch of the speech wave-form. Sampling will stop completely during the least gap in the message, function, for example when the speaker is taking in breath. Immediately after each sample all channels and trunking cross connections are again freed for other samples. Any additional communication may be added to the system regardless of its nature. Teletype, telemetering, music, voice facsimile and even television could operate simultaneously and mixed over the same system, provided the bits could be made short enough (order of millimicroseconds). All repeaters could be of the regenerative type, capable of handling equally well all the above transmissions. They could be produced absolutely identical for all stations. No filters would be required, no single side-band modulators and demodulators; radio-relay station could be the simplest type.

Today such prospects sound fantastic, but considering that binary coding and switching are well established techniques, that delay circuits and regenerative repeaters are perfectly understood and that information theory has proved the high degree of redundancy available in voice communication, we may frankly state that the realisation of such an ideal system is a problem of organization and administration rather than a technical one.

Telemetering, telecontrol, and recently also certain *servosystems*, favour pulse operation in time-division as a highly efficient method.[55, 56] Synchronous multiplexing is at present the rule, but the large redundancy of many of these systems may lead to attempts to increase their efficiency by methods operating on the lines discussed above.

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53. W. H. Huggins, "A theory of hearing," pp. 363-380 in W. Jackson, *loc. cit.* (Ref. 16).
54. D. Levine, "Information cells on intensity modulated c.r.t. screens," *Proc. Inst. Radio Engrs*, **41**, pp. 1766-1768, Dec. 1953.
55. M. L. Greenough and C. G. Gordon, "Parachute-borne telemetering system," *Electronics*, **27**, pp. 148-151, June 1954.
56. A. J. Bayliss, "A ten-channel pulse code telemetering system," *Electronic Engng*, **24**, pp. 485-489, Nov. 1952.

. . . Radio Engineering Overseas

A selection of abstracts from European and Commonwealth journals received in the Library of the Institution. All papers are in the language of the country of origin of the journal unless otherwise stated. The Institution regrets that translations cannot be supplied.

621-52:621.039

Recent progress in reactor control.—J. WEILL. *Onde Electrique*, **35**, pp. 867-887, October 1955.

A review of modern techniques used in nuclear reactor control is given. Different types of detectors as well as amplifiers are described. Detailed considerations are given to an automatic control system which can be applied to different types of reactors.

621.315.212

Investigations of a wide-band coupler between co-axial and a wave guide. I. Theoretical; II. Measurement.—E. BELOHOUBEK, *Archiv der Elektrischen Ubertragung*, **9**, pp. 432-440 and 469-474, September and October 1955.

Part 1: The investigation starts from a wave-guide coupler in which the inner conductor of the coaxial line serves as antenna and the wave guide is closed at one point by a sliding short. On the basis of theoretical considerations it is outlined how the wideband character of the coupler depends on the individual matching parameters: length of the antenna, eccentricity of the antenna coupling unit, and distance of the shorting bar from the antenna. With suitable proportioning—the diameter of the antenna also proved to be critical—very wide bands, in particular for mid-band waves around $\lambda_0 = 1{\cdot}4a$ (a = width of wave guide) can be obtained. For bands near $\lambda_0 = a$ and $2a$, the attainable bandwidth decreases again heavily.

Part II: The coupling conditions between a wave guide of cross section 72.1 mm × 34 mm and a coaxial cable of the diameters $D_i = 7{\cdot}5$ mm, $D_a = 20$ mm are investigated with respect to the bandwidth attainable at different mid-band frequencies. The measuring arrangement used for this purpose was developed with great care so that even low reflection coefficients can be measured satisfactorily. The possible sources of error are outlined and their influence on the measuring accuracy estimated. The measured results, represented in graphic form, show wide agreement with the theoretically expected values. Bandwidths of 20% with a reflection coefficient of less than 2% could be realised at waves around 10 cm length.

621.37/8:539.185

Determination of the transport free mean path of thermal neutrons by measuring a complex length of diffusion.—V. RAIEVSKI and J. HOROVITZ.

Electronic equipment used in the measurement of the diffusion length in graphite.—M. LE BAUD. *Onde Electrique*, **35**, pp. 821-832 and 833-836, October 1955.

The neutron beam entering the graphite is modulated and the diffusion length is determined directly by phase comparison.

621.372.543.2:621.397.5

The transient state and television.—P. POINCELOT. *Onde Electrique*, **35**, pp. 764-767, August-September 1955.

The transient response of a dissipative-law pass-filter is considered. The filter consists of any number of sections matched by their image impedance, and the input signal follows an exponential law. The response of such structures corresponding to particular cases is compared with that of a television channel.

621.385.032.216

Thermionic emission of sintered mixtures of tungsten and alkali-earth carbonates.—G. MESNARD and R. UZAN. *Le Vide*, **10**, pp. 105-118, July-September 1955.

The interest and technique of preparation of cathodes formed with pressed and sintered mixtures of powdered tungsten and alkaline earth carbonates are at first pointed out. Thermionic properties of such mixtures are then studied with reference to chemical reactions between components. The effects of an addition of carbon or nickel and the influence of proportions are considered. Schottky effect and variations of cathode resistance produced by treatments are finally examined.

621.385.2

The rise of the noise temperature of a space charge limited diode.—H. W. KONIG. *Archiv der Elektrischen Ubertagung*, **9**, pp. 411-418, September 1955.

Under the assumption that the random noise is due to a velocity fluctuation (v-fluctuation) plus a current fluctuation (q-fluctuation) at the cathode (correlation factor $r+js$) the paper calculates the noise temperature for small deviations of the space charge factor ζ from the ideal value of unity. If the space charge factor is but a few tenths of a per cent. below unity, a considerable rise is already caused in the noise temperature with increasing anode voltage. The deciding part thereof is due to the q-fluctuation. In this way the observed noise temperature of many times the theoretical value can be interpreted.

Simple estimates reveal that exceedingly minute inhomogeneities in the structural conditions of the cathode may cause such a reduction by a few tenths of a per cent. below the ideal value of $\zeta = 1$. The analysis given in the paper offers the possibility of an experimental determination of the imaginary part s of the correlation factor.

621.392.5.091.012

Universal diagram for determination of absorption in standard electrical band-pass filters. L. GYERGYEK. *Elektrotehniski Vestnik*, **23**, pp. 242-246, July-August 1955.

Deals with the method of determining by diagram the absorption of electrical band-pass filters designed on the basis of image parameters. It is possible to determine the absorption of a band-pass filter accurately enough with little effort; furthermore, the method permits of the selection of parameters in order to obtain the requested minimum absorption within the prescribed range of frequencies. The diagram shows plainly the course of absorption against frequency and is therefore suited for practical application.

621.395.8:621.376.3

Constant amplitude modulation for telephony.—P. Deman. *Onde Electrique*, **35**, pp. 739-746, August-September 1955.

Considerations of transmission efficiency lead to the use of compression at the origin of a noisy telephone circuit and the existing knowledge of acoustics permits the definition of annoyance value due to the noise picked up during transmission. To obtain maximum protection against noise it is necessary to add pre-emphasis before transmission and to control the expansion at reception by an auxiliary channel known as the level channel. The use of a frequency modulated channel, by decreasing mutual interference when the speech and level channels use a common circuit, enables the theoretical limit to be more nearly approached.

621.396.11

Variations in the phase constant of ground.—M. Argirovic. *Onde Electrique*, **35**, pp. 687-691, July 1955.

The rule of the sum of the numerical distances, for different terrains, can be applied to calculate the equivalent phase constant. Curves calculated show the variation in field and equivalent phase constant with distance for different electric coil constants and for different frequencies.

621.396.11

Magneto-ionic triple splitting over Delhi.—S. N. Mitra. *J. Instn Telecom. Engrs*, **1**, pp. 124-129, September 1955.

The various possible causes for the occurrence of this rare phenomenon at the low geomagnetic latitude of Delhi are discussed. It has been indicated that the triple splitting is likely to be caused by the longitudinal propagation of the ordinary ray associated with an increase in the collisional frequency of the ionospheric layers.

621.396.11

Atmospheric noise interference to broadcasting in the 5 Mc/s band at Poona.—K. R. Phade, *J. Instn Telecom. Engrs*, **1**, pp. 136-146, September 1955.

The results during the hours 18.00 to 23,00 (I.S.T.) are utilized to compute the data on noise necessary for broadcasting services. These data are compared with the estimates of CRPL, Washington, and the measurements carried out on behalf of the Radio Research Board, London, at places in or near India. The discrepancies are explained and it is shown that both give lower values of noise. Noise levels are estimated from lightning discharge data and compared with measured values. There is close agreement between the estimates and the measured values. In all these comparisons, it is shown that such of the differences, as arise between the 3 Mc/s measurements reported elsewhere and the 5 Mc/s measurements, arise entirely from the differences in the propagation at the two frequencies.

621.396.665:621.396.97

Peak limiting amplifiers.—Ram Yadav. *J.Instn Telecom.Engrs*, **1**, pp. 147-154, September 1955.

The requirements for controlling the wide range of amplitude signals, encountered in broadcasting and other similar telecommunication services, are discussed. It has been found that a combination of manual and electronic control is necessary to keep the modulation high without overloading the system. The electronic control is carried out with the help of a variable gain amplifier, commonly known as a peak limiting amplifier. A limiting amplifier based on some desirable performance characteristics is described with particular reference to its dynamic behaviour.

621.396.97

Transmission properties of audio frequency programme circuits in the German long distance carrier frequency cables.—Von E. A. Pavel, *Fernmeldetechnische Zeitschrift*, **8**, pp. 455-461, August 1955.

The results collected from tests on a broadcast land-line of 2212 km length are reported. The land-line is a low frequency circuit along superimposed circuits on star quads in modern German carrier-frequency trunk cables.

621.397.5

Television for industrial, educational and scientific applications.—A. Fayard. *Onde Electrique*, **35**, pp. 641-644, July 1955.

The equipment for industrial television described comprises a vidicon camera and a control box; its design suits it to the widest applications. The scanning system is the same as that used for broadcasting so that commercial receivers can be used if necessary.

621.397.74

The European television network and "Eurovision 1954."—F. Kirschstein and H. Bodeker. *Fernmeldetechnische Zeitschrift*, **8**, pp. 419-425, August 1955.

Describes the network employed in the Eurovision transmissions and the way it is operated. Oscillograms are given for observed transmission faults, and photographs show the quality of the transmitted pictures. The present work for the improvement of international transmissions is discussed.

681.14:621.039

An analogue harmonic analyser.—J. Pottier. *Onde Electrique*, **35**, pp. 847-866, October 1955.

A non-destructive method for the evaluation of the purity of the materials employed in nuclear reactor construction, is described. The test sample is oscillated inside the reactor and a harmonic analysis of the time variation of the reactor power performed.

681.14:621.039

Analogue calculators as pile simulators.—P. Braffort. *Onde Electrique*, **35**, pp. 888-898, October 1955.

After a brief review of the fundamental principles of analogue calculating machines the author gives various diagrams relative to the kinetic equations of a nuclear reactor. A simulator used in the study of the automatic control of the Saclay pile is described.