SECTION 5: PRODUCTION FACILITIES

Planning and Costs of Broadcast Facilities

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The information to follow may be used as a guide for general planning and cost estimating of facilities. Areas covered include the following:

- Checklists of items to be considered in any facility planning,
- Systems for television facilities,
- Acoustic considerations,
- Cost estimating and typical cost figures,
- Planning for AM and FM radio facilities.

Many factors determine the initial budget, as well as final cost figures at the time of actual completion. Proper planning and project coordination by the owner's representative is mandatory if a well-built, cost-effective facility is to be produced, by the date required. After many years of experience, the writer has been involved with numerous large and small broadcast facility projects, all having one major thing in common: the fact that time begins to run out before you are really finished. This may not actually be so bad, as otherwise, what else could ever stop the additional requests and change-orders, given the ever changing nature of our business.

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GENERAL PLANNING CONSIDERATIONS

The early planning of a broadcast facility usually involves a number of factors such as: consideration of the market to be served; site selection; studio requirement; station programming and personnel; hours of operation; and available capital. First and foremost of the decisions to be reached is whether the studio and transmitter are to be combined under one roof or to be in separate locations. It is generally agreed that wherever practical it is most economical to combine the studio and transmitter. The initial equipment requirements are less, but more important is the fact that the day-to-day operating expenses are lower. With the plant "all under one roof" there are savings in heating, air conditioning, building maintenance, travel time, and personnel. A combined operation, however, is not always possible.

When a combined operation is not practical, the second approach is, of course, to operate the transmitter by remote control from the studio. By utilizing remote control, a transmitter site can be selected that is most advantageous from a coverage standpoint, and the studio can then be placed in the most convenient location. The building requirements at the transmitter can be the very minimum, requiring only space for the equipment, a small work area, and a small heating unit. The studio then would contain both the programming and business functions.

Codes

Virtually any construction or alteration of a building may be covered by one or more codes. In addition to general building codes, there are electrical, mechanical, plumbing and other specialized codes. National associations have proposed model codes; however, it is up to state and local jurisdictions to adopt whatever codes, with variations, which they find appropriate. Reputable architects, contractors and subcontractors are familiar with local codes affecting their areas and should be involved in the work. Zoning is another consideration which is dependent on the local jusridiction.

FACILITY PLANNING

The initial step in station planning is to develop an outline of requirements that will form the basis upon which future decisions are made. Such an outline for broadcast facility planning may be developed as follows:

Outline for Broadcasting Facility

1. Site Selection

a. Adequate space for immediate building needs plus anticipated expansion.
b. Adequate parking space for employees, guests, and studio audience (if latter is being considered).
c. Trucking access and adequate loading area with loading dock, if possible.
d. Accessibility from high speed or uncongested roads (mobile units, audience, convenience of employees).
e. Transmission tower space or line-of-sight for microwave.
f. Zoning: use of towers, antennas, identification or advertising signs.
g. Possible use of helicopters for news and traffic reporting.
h. Relation of site to environmental elements: (i) noise, (ii) weather, (iii) drainage.

2. Building Program (Space Arrangement)

a. Proper flow and separation of studios, related technical spaces, craft shops and storage, programming and engineering, talent and dressing, administrative offices and mobile units.
b. Entrance arrangements for employees, guests, VIPs, and audience.
c. Relation of studios to loading areas.
d. Relation of parking areas to employee and audience entrances.
e. Security: inside, outside.

3. Building Construction—General

a. Selection of structural system, taking into account local conditions, codes, availability of materials, flexibility for future changes, special loading conditions and spans, degree of fire resistance and effect on fire insurance rates. Careful study of fire code requirements.
b. Partitioning: types to afford relatively good sound isolation, yet flexible enough for ease of removal or relocation. Better to make rooms larger in beginning and subdivide later if necessary.
c. Ceilings—high degree of accessibility for multitude of communications wiring; good acoustical value; cleanability, ease of electronic repairs.
d. Wall materials—extreme durability in technical and production areas, due to frequent moving of equipment and supplies.
e. Floor materials—durable yet resistant to traffic sounds and vibration. Raised computer floors may be used in control, video tape, and other electronic areas.

(Note: Floor slabs for TV studios must be extremely level for proper camera movement.)

4. Engineering—General

a. Availability of incoming HV power service from power company
b. Separate unit substations for: (i) air-conditioning, air handling and other facility motor loads, (ii) general illumination, (iii) studio production lighting, (iv) technical loads.
c. Lighting in general—fluorescent, 100-foot candles in working areas, dimmer controlled incandescent in control rooms.
d. Air conditioning—heavy studio loads (TV), low velocity for sound control, stand-by AC for master control rooms, separate exhaust system (air purging) for TV studios.
e. Miscellaneous systems—telephone (technical and commercial), public address, watchman's tour, door security, closed circuit TV and fire alarm.

5. Budget Estimate

The following Checklist and Project Budget Form are designed for estimating the scope and cost of station construction.

PROJECT BUDGET FORM

Call Sign ______________________ Start Date ______________________

Service ______________________ Finish Date ______________________
(Radio or TV)

Type of Facilities: (Check as applies)

Office ____________ Studio ____________ ______________________
Street address of project

Transmitter ____________ Mobile ____________ ______________________
City, State and Zip

Item #	**Description**	**Estimate**	**Actual**
1.	Land purchase/lease	________	________
2.	Survey-property (and other)	________	________
3.	Title search/insurance	________	________
4.	Real estate broker/commission	________	________
5.	Architects/engineers—fees	________	________
6.	Permits and licenses (if separate)	________	________
7.	Consultants-acoustic structural, etc.	________	________
8.	Site preparation and demolition	________	________
9.	General construction and finish	________	________
10.	Optional construction items and finish	________	________
11.	Special construction and finish	________	________
12.	Furniture and fixtures	________	________
13.	Decoration—interior	________	________
14.	Landscape—exterior	________	________
15.	Special equipment—electronic (and other)	________	________
16.	Special equipment—installation of above	________	________
17.	Contingencies (including price increases)	________	________
18.	Other	________	________
19.	Other	________	________
20.	Other	________	________
	Totals	________	________

Prepared by ______________ Checked by ______________ Approved ______________

Suggested Checklist for Planning

ITEM
- Land
- Land tests
- Site clearing
- Architects' and engineers' fees
- Permits
- Special consulting fees (include interior decorator)

GENERAL CONSTRUCTION
- Architectural
- Electrical
- Mechanical
- Heating and Plumbing
- Special in-house cabling (TV, Music, PA)

INTERIOR FINISH
- Wall covering (fabric or paint)
- Floor covering
- Special studio treatment

FURNITURE AND FIXTURES
- Decorative office furniture
- Standard office furniture
- Office area built-ins
- Working area counters, cabinets and built-ins
- Draperies
- Art work

TELEPHONE
- Broadcast line facilities
- TWX or facsimile
- Type of switchboard (owned or leased)
- Office interconnection needs
- Number of private lines
- System—owned or leased

SECURITY
- Control of public (where received)
- Control of all points of entry
- Closed circuit TV systems, watchman's tours
- Electrified gates and doors
- Separation of 9:00-5:00 areas from 24-hr. areas
- Night lighting

MISCELLANEOUS ITEMS (likely to be overlooked):
- Copier outlet (special power hookup)
- Special ventilation for odor areas
- Building maintenance equipment closets
- Drinking fountains and vending machine areas
- Space for air conditioning subdistribution boxes, fans, etc.
- Special waste water treatment for film darkroom processing, if local ordinances dictate
- Cable connection to roof for radio and television antenna system and two-way antenna
- Possible microwave antenna mounts

6. Electrical

General

The following is a sample specification for the electrical system of a large television station. It is based on the National Electrical Code and is intended to show how such a specification may be set up. Elements may be adapted to other applications, including radio station.

It is important that all state and local electrical, fire and safety codes be observed.

Area Classification

All electrical installations, materials and equipment shall comply with the classification "General Purpose" except for hazardous areas which shall be designed for Class I, Group D, explosion-proof conditions.

Incoming Power Service and Metering

1. Incoming power service may be high voltage (4.16 kv or 13.8 kv) due to high load requirements.
2. Standby incoming service shall be provided with automatic transfer when normal service fails.
3. One point of primary metering shall be provided to obtain best possible utility rates.

Primary Distribution

1. Distribution within complex may be high voltage (4.16 kv or 13.2 kv) from a primary switchgear to unit substations located as close as possible to the center of the loads served.
2. Separate unit substations shall be provided for different type loads, as follows:
   With secondary 120/208 v to handle equipment and motor loads and all fluorescent lighting;
   With secondary 120/208 v to handle receptacle, incandescent lighting and small equipment loads;
   With secondary 120/208 v to handle studio production lighting only;
   With secondary 120/208 v to handle technical TV loads only.
3. Each unit substantion shall include components as follows:
   Primary compartment with Hv fused load break switch;
   Open dry type transformer with Hv primary delta and secondary 120/208 v or 277/480 v, 3 phase, 4 wire. Sound rating of transformer shall be best possible;
   Voltmeter, ammeter, and selector switches;
   Main secondary air circuit breaker;
   Moulded case feeder circuit breakers and spares.

Secondary Distribution

1. Power shall be extended from unit substations with cable and conduit to automatic circuit breaker panels and motor control centers.
2. Motor control centers shall be Class I, Type B, with combination magnetic, full voltage starting, circuit-breaker-type motor starters or circuit breakers only for 480 v, 3-phase operation. Each starter shall have 3 thermal overloads.
3. Power panels shall be designed for 480 v, 3-phase, 3-wire service. Panels shall be of the dead front type with automatic circuit breakers of ampere rating as required.
4. Panels for receptacle and incandescent lighting loads shall be designed for 120/208 v, 3-phase, 4-wire service. Panels shall be of the dead front type with automatic circuit breakers of ampere rating as required.
5. Lighting panels shall be designed for 277/480 v., 3-phase, 4-wire service. Panels shall be of the dead front type with 20 ampere automatic branch circuit breakers. Panels shall be similar to Westinghouse Type NH1B-4.

Conduit

1. Rigid steel conduit asphaltum painted shall be used when installed in concrete slabs, below grade and outdoors above grade.
2. Rigid aluminum conduit shall be used for exposed installation in mechanical equipment rooms, damp locations and locations where exposed to mechanical damage.
3. Rigid aluminum or steel conduit shall be used for all feeder and subfeeder runs.
4. Steel EMT with compression weathertight fittings shall be used for all other branch circuit wiring indoors and above grade.

Wire

1. HV cable shall be single conductor crosslinked polyethylene insulated and shielded.
2. Building wire shall be Type THW rated at 600 v-75 degrees C No. 12 AWG and smaller shall be solid copper. No. 8 AWG and larger shall be stranded.
3. Fixture wire shall be Type AF, 300 v insulation.
4. Minimum wire size shall be No. 12 AWG, except No. 14 AWG for control wires. Maximum wire size shall be No. 500 MCM.

Grounding

1. Electrical grounding shall be provided in accordance with the National Electric Code. Equipment enclosures, electrical service, transformer neutrals, outdoor lighting standards, and cable shielding shall be grounded.
2. Insulated bushings and double lock nuts shall be provided at all panel boards and pull boxes in feeder runs and pull boxes shall be bonded through with bare copper wire.
3. Separate technical equipment ground system shall be provided as required.

Switches, Wiring Devices,
Wall Plates and Special Enclosures

1. Single pole switches shall be 20 amperes, 120/277 v, ac, quiet type.
2. Duplex receptacle shall be 20 amperes, 125 v, 2 pole plus U-slot ground.
3. Special outlet to be provided as required.
4. All wall plates for switch, receptacle, and telephone outlets shall be .06 in. stainless steel.

Telephone System

1. Two incoming underground services are required, one for technical use and one for commercial use.
2. Equipment room for the technical service shall be located close to Master Control and there shall be a cable-tray tie between the equipment room and Master Control.
3. The commercial system shall be complete, consisting of conduits from equipment room outlying telephone closets and interconnecting panels and thence to the various outlets as required. All installations shall be in accordance with the requirements of the local telephone company.
4. Interconnecting panels shall be of steel with plywood backboard, full opening door, latch, cylinder lock and trim.
5. Telephone closets shall be furnished with plywood backboard.
6. Conduits shall be ¾ in. minimum.
7. Telephone outlets shall be 4 in. sq. with bushed hole cover plate.
8. Equipment instruments and wiring shall be by the telephone company.
9. Some recent trends are for the broadcast station to actually own the telephone equipment. Should this be true, Items 1 through 7 are still to be used as a guide.

Public Address System

1. A complete PA system consisting of amplifiers, loudspeakers, and microphone shall be provided.
2. Loudspeakers shall be located in corridors throughout the complex and calls shall originate from the telephone operator's desk.
3. System shall be zoned as required.

Fire Alarm System

1. The fire alarm system shall be closed circuit zoned, consisting of control cabinet, gongs,

manual stations and automatic fire detectors.
2. Manual stations shall be provided at each stair on each floor and at all ground level exterior doors.
3. Automatic thermal or smoke detector shall be provided in all areas except where sprinkler heads are installed.
4. Each sprinkler alarm valve shall indicate on the fire alarm panel zone annunciator as a separate zone when activated.

Video Cable Trays and Audio Signal Conduits

1. A system of cable trays and signal conduits originating from Master Control shall be provided to studio control rooms, studios, microwave rooms, electronic shop, program computers, etc.
2. In addition, a separate cable tray with antenna cables shall be provided from Master Control to all areas where antenna outlets would be required.

Studio Production Lighting System

1. Unit substation and dimmer board shall be located as close as possible to studio served.
2. Unit substation shall include the following:
   a. Electrically operated main circuit breaker to permit remote control from studio floor.
   b. Transformer with 6-2-1/2 percent taps, 3 above and 3 below rated primary voltage, to compensate for possible secondary voltage variations.
3. Other work shall be as follows:
   a. Wireaway with wiring from load size of dimmers to studio floor patch panel.
   b. Studio grid wireaways with load wiring to studio patch panel;
   c. Wiring under studio floor from studio floor "pockets" to studio patch panel;
   d. Control wiring from studio control console to dimmer board.
4. In sizing unit, substantions serving dimmer boards, a 50 percent demand factor may be applied to connect dimmer load.
5. An "on-air" studio warning light system be provided as required.

Security

1. The following security systems shall be provided:
   a. Supervision of all exterior doors on ground level, with control cabinet in guard room;
   b. Closed circuit TV cameras at key positions, with monitor in guard room;
   c. Manual nonwired watchman's tour stations located throughout complex;
   d. Electrically operated gates to control automobile traffic.

Emergency System

1. Power for the emergency system shall be provided with a water-cooled diesel generator set with generator voltage 277/480 v, 3-phase, 4-w. The installation shall include all accessories such as automatic transfer switch, output switchboard, battery starting set, oil storage tank, fuel pump, mufflers, vibration isolators, etc.
2. Generator set shall automatically sense power failure or 80 percent undervoltage, start engine, attain and maintain speed, and transfer emergency load. Manual override of start and transfer of load controls should be provided.
3. Provide local transformer with primary delta 480 v, 3-phase and secondary 120/208 v loads on emergency supply.
4. Loads on emergency supply shall include stair lights, exit signs, selected corridor lights, PA system and all technical lighting and heating, ventilating, and air conditioning loads required for transmission of live news programs and taped programs.

Lighting (277 v for Fluorescent, 120 v for Incandescent)

1. Lighting fixtures shall be completely installed with all required outlet boxes and accessories.
2. Lighting levels shall be in accordance with IES latest recommendations, with minimum 100 FC in working areas.
3. Fluorescent fixtures shall be used for general illumination. Fixtures shall be with 40 w RS lamps and HP factor ballast, with best sound rating. Fluorescent fixture types shall be as follows:
   a. Recessed with acrylic lens diffuser to be used in areas with hung ceiling;
   b. Surface or pendant mounted with wraparound acrylic lens diffuser to be used in stairs and other selected areas with exposed ceiling;
   c. Industrial RLM with porcelain reflector to be used in mechanical equipment rooms, storage rooms, etc.
   d. Executive offices and conference rooms shall be provided with dimmer-controlled incandescent lighting using recessed fixtures.
5. Selected walls and art work shall be illuminated with recessed ceiling-mounted incandescent wall-washing fixtures.
6. Make-up room mirrors shall be illuminated with special bracket wall-mounted fluorescent fixtures. Dressing room mirrors shall be illuminated with special wall-mounted strips with incandescent bare lamps.

Outside Lighting

1. Outside lighting shall include illumination of audience concourses, entrances, parking lots, signs, building exteriors, planters, etc.
2. Lighting levels for all outside area illumination shall be strictly in accordance with IES latest recommendations.

Miscellaneous

1. Wall-mounted clocks operating on 120 v shall be provided complete with outlet in designated areas.
2. Clock system for technical use with master clock in Master Control and indicating clocks in studio control rooms, offices, etc., shall be provided as required.
3. Local office intercommunication systems shall be provided as required.
4. Local sound systems shall be provided for large conference rooms.

Mechanical

General

Color television studios require special consideration in solving the many problems entailed in the mechanical design due to the high-lighting capacity, noise criteria, air distribution and emergency stand-by operation. Black and white, color television studios, and radio broadcasting facilities would require the same consideration, but some of the problems would not be as severe.

Design Conditions

1. The optimum summer and winter design conditions to be maintained by the air-conditioning system is 75 degree F dry bulb bulb and 50 percent relative humidity. However, these conditions should be checked (in respective area with the owner's) since they may require slightly different criteria.
2. The outside design conditions are dependent upon the geographical location from data established by the US Weather Bureau.

Air-Conditioning Loads

1. Studio and production lighting for color television constitute the major portion of the heat gain and can exceed 75 percent of the total cooling load requirement. The unit lighting load requirement in the production area of the studio can equal 50-60 watts per square foot of floor area and in many cases this load can occur in any part of the studio since the production area and audience accomodations are flexible.
2. Transmission and solar heat gains are minimal since the exterior walls are well insulated and windowless.
3. Another contribution to the cooling load results from occupancy heat gain and the fresh air load. The fresh air requirement should be based on either 15 CFM per person, or the equivalent of one air change of fresh air, whichever is greater. It is also important to check and insure that the fresh air quantities conform with all code requirements.

Method of Air Distribution and Noise Control

1. Proper air distribution and air movement are of critical importance. Systems should be designed so that within a zone of 12 ft. above floor level, an air movement of 25 fpm is not exceeded. Air velocities exceeding the 25 fpm can cause movement of performer's hair, clothing, and stage props, which are usually highly expendable and flimsily built of thin canvas and light plywood.
2. The air supply should be introduced at a level above the movable lighting grid system to prevent interference with the closely spaced batten strips. Low-level return grilles located at the perimeter of the studio, in principle, would be desirable. However, due to the nature of studio operation, the grille could be blocked off by the cyclorama curtain or by studio props, which would result in an ineffective return air system. It could also be a possible source of noise generation. Locating the return air outlets at a level above the air supply will tend to relieve the neutral zone before it can heat the ceiling and radiate heat downward. Proper location of return air grilles and maintaining low velocities will eliminate any possible "short circuiting."
3. Noise criteria are of utmost importance and unless proper consideration is given to this problem, it can result in a nonproductive studio. Noise level should be within a range of NC 20 to NC 25, so as not to interfere with studio performance, particularly during scenes where there is no conversation and no background sound effects. Duct velocities should be designed for approximately 400 fpm within 10 ft. of diffuser or register opening, 525 fpm within 10-30 ft. from opening, 700 fpm within 30-50 ft. of opening, and 800 fpm within 70-90 ft. of opening.
4. All ductwork (supply and return) should be acoustically lined for sound attenuation and the sound power level of all outlets should be carefully checked to insure that it does not exceed the decibel rating at the end of the duct run, otherwise it will become additive (logarithmic) and negate a portion of the acoustically treated ductwork.
5. All piping should be insulated and all ductwork should be externally insulated to

eliminate reflected sound in studios. Where ductwork and piping pass through walls or floor, the openings should be sealed with acoustical sound-deadening material. Ductwork and piping should be suspended from vibration isolators. Where ductwork and piping pass through studio walls, flexible pipe and flexible duct connections should be provided. Piping should be sized so that velocity of the medium transmitted is low enough to be inaudible.

6. Mechanical equipment should be located remotely from studio to eliminate transmission of sound and vibration. All equipment should be properly supported from vibration isolators. Sound traps for sound attenuation and flexible duct connections to prevent transmission of vibration should be provided for all air-handling apparatus.

Type of System and Control

1. Each studio should be served by its own air-handling apparatus and should consist of supply and return fans, filters, heating and cooling coils, sound traps and a purge exhaust system.
2. There are several schemes that can be utilized in the arrangement and selection of the component parts of the system, and this is somewhat dependent upon the economics, space conditions, and geographical location of the project. An "economizer" cycle utilizing 100 percent fresh air during intermediate season or a fixed percentage fresh air system can be used. The application of preheat coils is also dependent upon the percentage of fresh air and geographical location. Heating coils can be either steam or hot water type and cooling coils should be of the chilled water type. In areas where freezing outside temperatures are experienced, special conditions have to be considered to prevent possible "freeze-up" of preheating coils and chilled water coils.
3. A separate purge exhaust system should be provided to permit studio to be evacuated during periods when it is not in operation. During purge operation, the system should handle 100 percent outside air without attempting to maintain studio design conditions. Where an economizer-type cycle is provided, purging of the area can be accomplished by resetting controls to 100 percent outside air.
4. Control system should be arranged to control studio temperatures and where facilities are provided for audience participation, additional humidity control should be provided.
5. The installation of a Supervisory Data Center would provide operational supervision of the project and would include remote control for resetting of space temperatures and humidity, starting and stopping of air handling system, "read-out" of other pertinent air and water temperatures and alarm indication.
6. Pneumatic temperature control system should be provided with standby air compressor with automatic cut-in features.

Stand-By Operation

1. Due to the critical operation of the Master Control Room and Videotape Room, a separate air-handling system shall serve these areas with provisions for standby equipment in the event primary equipment fails. This can be accomplished by interconnecting the ductwork (properly dampered) with another air-handling system serving a noncritical area in the building (i.e., office areas), thereby permitting the Master Control Room and Videotape Room to be satisfied during an emergency period. Another desirable feature to be incorporated in the system is provision for handling 100 percent fresh air in the event refrigeration equipment becomes inoperative.
2. Multiple refrigeration equipment and boiler equipment should be provided so that in the event a single unit becomes inoperative, partial operation can maintain conditions in critical areas.
3. Compressed air system serving videotape machines should be provided with standby compressors to insure continuous operation. Compressors that are water cooled can operate off the chilled water system and arranged so that in the event there is a loss in chilled water pressure, the cooling system will automatically switch over to city water.
4. In the event of an electrical power failure, an emergency generator should start automatically to maintain operation of the boilers, heating pump, air-handling system serving the Master Control Room and Videotape Room, pneumatic temperature control air compressor and air compressor serving videotape machines.

ACOUSTICS

Studio Planning

When a microphone is opened in a studio or combo control room, it is expected to pick up the voice of the talent. But if it is also subjected to building rumble, office noise, air conditioning noise, hissing radiators, and unwanted reflections from the room walls; the resulting air signal, YOUR PRODUCT, can sound cheap and hollow. It can actually diminish intelligibility, and possibly invite listener fatigue in severe cases. There is no complete electronic fix for poor acoustics.

Creating the "ideal" studio acoustic is an expensive undertaking. It requires the expertise of an acoustical consultant and an architect, who are both familiar with studio construction. Retaining professionals who specialize in unrelated fields within their professions should be avoided, because of the special nature of broadcasting requirements. Good designers will often find it necessary to recommend types of construction which are two to five times as costly to execute than standard office construction. However, the resultant studios and control rooms will be well isolated from intrusive noise and vibrations, and will project a clean and crisp On-Air signal from live talent, free of unwanted resonances and echoes.

Costs of Acoustic Treatments

Many factors affecting the final costs of construction are actually under the station's control. First, determine whether your needs require the "ideal". A Top 40 station whose talent works only a few inches from a single microphone, relies far less on room acoustics for its live "sound" than does an All Talk format station requiring five open microphones to pick up guests at any variety of distances. Because of their wider frequency response and greater dynamic range, and the fact that their listeners may tune in on sophisticated high fidelity receivers, FM stations generally require more closely controlled studio acoustics than their AM counterparts. Remember that the more a station employs signal compression and limiting, the more evident will be the annoyance of poor studio acoustic characteristics. Conversely, compression on stations with better studio acoustics will be less noticeable during live broadcasts.

When planning new studios, significant cost reductions can be realized from selecting a site which is far removed from noise generating sources such as pumps and air conditioning equipment; street, rail and air traffic; and manufacturing equipment. Be sure to consider ceiling height. Large HVAC ductwork is required to keep air flow noise at a minimum; and if insufficient space exists for the simple installation of the ductwork, expensive alternatives will be required. Likewise, when roughing out the details of a floor plan, don't draw walls as single pencil lines. All partitions have some thickness which will take up floor space; and it is not uncommon for some acoustic partitions employing internal air spaces to exceed a 16" thickness. If space is not a premium, such constructions are generally more economical than thinner partitions with similar transmission characteristics. Absorptive surface treatments can easily add an additional 4" to 6" to each side of the wall profile.

The control of the transmission and reverberation of sound are two disciplines in acoustics which apply to broadcast environments. They are virtually independent and require completely different methods and materials in creating studios suitable for broadcast use. It is important that each be understood, since confusion between the two can result in aggravating an existing problem.

Controlling Sound Transmission

Sound Transmission results when a source of sound is carried from one area to another. The vehicle of transmission may be a direct path, a flanking path around an object, or a path carrying vibration. The most economical and practical means of limiting the transmission of noise usually requires the control or reduction of it at the source. Disturbances with low frequency components are the most difficult and expensive to contain. Completely enclosing the noise source in a highly absorptive room, and isolating its vibrations from the building structure will limit the ability of noise to intrude upon other areas. For example, if footfalls on a cement floor adjacent to or above the studio are causing problems, try carpeting those areas before attempting to increase the transmission loss of the studio construction.

Where treatment at the source is not practical, the most effective way of isolating an area against the intrusion noise is to employ floating construction. With floating construction as shown in Fig.1, the noise sensitive area is actually a room built within a room. The most important aspect of this type of construction is that mechanical contact between the inner room and outer room is kept to a minimum, with the construction of both rooms creating as complete and uninterrupted an envelope as possible. This is most commonly achieved by resiliently supporting the floor and suspending the ceiling of the inner room on springs or neoprene isolators. Hence the term "floating". Usually in radio studios the walls have no connection between them; but in television studios, where the ceilings are significantly higher, resilient sway brace connections between the inner and outer walls may be required to stiffen the construction, and prevent buckling.

Depending on the intensity and proximity of the noise source, and the degree of residual noise which can be tolerated, some or all of the aspects of floating construction can be employed in designing a broadcast studio or control room. The resiliently isolated floor and ceiling is necessary to prevent structure borne sound and vibration from entering the interior space. The double walls with the resulting air cavity between them are actually more effective than a single partition of the same mass. In order to preserve this advan-

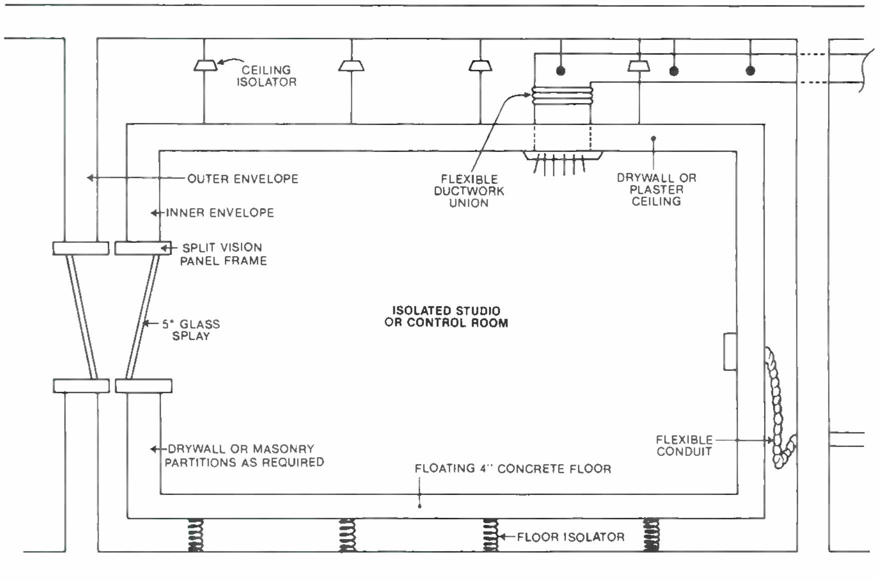


Fig. 1. Floating construction schematic showing mechanical isolation between inner and outer rooms.

tage, no service such as air conditioning ductwork, plumbing, or electrical conduit should be allowed to make a solid mechanical contact between the outer and inner rooms. The use of flexible connections within the cavities is extremely important.

When spring or neoprene isolators are employed for vibration isolation, the effectiveness of the isolator is measured by its nominal deflection, the relation of its natural period to that of the disturbing vibration, and the stiffness of the structure supporting it. It is important that isolators with the proper characteristics be specified. Isolators which are too lightly loaded will provide little isolation, while isolators which are too compliant may cause instability, or bottom out. If the isolation system exhibits a natural resonance equal to the disturbing frequency, the transmission of the disturbing frequency will be amplified.

One often overlooked source of vibration which may induce unwanted disturbances into adjacent areas is the control room monitor speaker system. Resiliently suspending the loudspeaker enclosure from the wall or ceiling will significantly reduce the structure borne transmission of sound from the monitor.

The materials used for the reduction of sound transmission in walls, ceilings and floors, are typically heavy, dense, and thick. Designing a barrier for maximum transmission loss would include practical considerations for the stiffness, resonance, mass, isolation, cost, and construction details of a partition. Typical partitions may be composed of brick or masonry block, poured concrete, lead, multiple layers of gypsum board supported by metal studs, or a combination of these. Regardless of the materials used, the most efficient barriers employ an integral airspace or cavities filled with a damping material such as fiberglass batts. Isolation at the lower frequencies becomes increasingly difficult and expensive, requiring stiffer, thicker and heavier construction than that needed for higher frequencies.

Perhaps the single most important aspect in the construction of acoustical barriers is sealing. Regardless of the materials used, the fit of the individual components with each other and the existing structure must be tight. All joints should be filled and sealed with a resilient, non-hardening caulk. The smallest openings can have disastrous results, and render otherwise expensive construction no better than a far cheaper counterpart. As an example, the TABLE 1 shows the reduction of the sound insulation (R) value of a hypothetical partition at some mid frequency, due to acoustic leakage through various size openings.

Once the design of the partitions has been decided, an entrance with the same sound insulation characteristic will be needed. No practical doors are commercially available which meet or exceed the R values of the most common partitions used in broadcasting. Therefore, a soundlock entrance scheme is required, which will attenuate sound through two doors and an intervening air space. The soundlock is a small vestibule between the studio and a hallway or other room requiring access to the studio. A small soundlock is detailed in Fig. 2. By entirely covering the walls of the soundlock between the studio and external doors with absorptive material, the efficiency of the soundlock will increase, decreasing the effect of opening one door. The use of a soundlock not only reduces the insulation requirement for each door, but also provides passage in and out of the studio or control room without exposing it to the full noise and disturbance of an adjoining area. Doors with good acoustic characteristics are available either in metal or wood, both of which feature a sound retardant core.

Like the partition requirements, the door must seal tightly within its frame. Lack of a good seal wastes whatever investment is necessary to procure a good sound rated door. Commercially available compression and/or magnetic type seals are recommended at the head and jambs of the door, while a mechanical drop seal and step saddle should be employed at its threshold. Some expensive pre-hung doors feature integral seals and can lift hinges, which actually lower the entire door into place tightly against the threshold saddle when it is closed. The greatest advantage of these doors results from the fact that they never have to be pushed shut to make a good seal all around the door. All other types require an oversize door check (pneumatic or hydraulic closer), and a latch to properly compress the seals when the door is closed.

In order to match the transmission loss characteristic of the partition, most installations will require double glazed vision panels. The two panes of glass should be of different thicknesses to minimize the coincidence of each pane's deficiencies. The glass should be mounted in a resilient neoprene or felt lined channel, and caulked tight. Some provision should be made for removing either piece of glass for replacement or cleaning. The glass should be mounted with a 5 degree

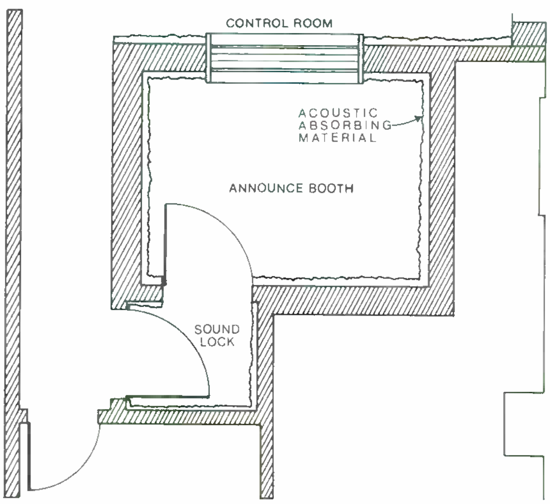


Fig. 2. Soundlock entry to a small announce booth.

outward splay at the top to reduce acoustic and visual reflections. Where a high degree of isolation is required, acoustic laminated glass should be employed. Each pane of this glass consists of alternating layers of plastic and glass; and it is lighter, less resonant, and a better acoustical barrier than standard plate glass.

Welded hollow metal frames are preferred over wood for both doors and windows for their stability. However, the void within them should be packed with either a cementitious or mineral fiber filler. It is important that the frames not connect the vertical partition sections in floating room or double wall construction.

In large cities where the cost of construction labor is high, small studios and control rooms may be economically built from modular components. The modular rooms are constructed from 4" thick, prefabricated hollow steel panels. A modular room usually includes a self supporting ceiling and an integral floor. The entire room is assembled on rails which can be isolated with springs or neoprene from the building structure. Most companies offering modular rooms also include double and triple partitions, doors, windows, conduit, ventilation ductwork, and a performance guarantee as available options. Because modular rooms can be disassembled and relocated, they are not usually considered leasehold improvements, and may offer certain financial benefits in addition to being somewhat portable.

Regardless of the design, the installation contractor must be familiar with the stringent requirements of acoustic construction. The efforts of the general construction, electrical and mechanical trades will have to be coordinated by the architect both in the design and execution of the studio project, to assure that no conflicts compromise the acoustic plan. The job should be inspected during construction by the acoustical consultant, to check for any potential leakage paths brought about by existing field conditions, and conformance to acoustical details.

If the station is to be situated in an office building, or other shared location, don't forget that other tenants may be annoyed by high level monitoring, tape rewind noises, and other sounds peculiar to broadcasters. So during the design phase, if the acoustic consultant recommends protection for adjacent building tenants, it may avoid a future embarrassment.

Controlling Reverberation Time

Separate and distinct from providing sound and vibration isolation, is that part of acoustics which governs how a room "sounds" when a sound originates in it. This is almost entirely a function of the room size, proportion and the ability of its contents to reflect, diffuse or absorb sound of differing frequencies. To complicate matters, all these parameters interact with each other in determining the reverberation characteristic of a room. The major measurable characteristics are a combination of the time that it takes a sound to decay 60 dB within a room, once the source of that sound is terminated (T 60), how the T 60's differ with frequency, how uniform the decay rate is, the ratio of early and late reflections, and the natural room resonances (modes). Room tuning is both an art and a science, which requires more research to fully understand. However divergent the philosophies involved, most consultants agree on several guidelines for radio and television studios.

The smaller and more symmetrical a room is, the more noticeable will be its undesirable resonances. This is why many television announce booths sound more like stuffy little phone booths. Avoid exceptionally long and narrow proportions, square rooms, rooms with concave walls, or rooms with a ceiling height equal to the height or width. Splayed walls and ceilings are dramatic, but necessary only as opposing surfaces which cannot be covered with mid and high frequency absorptive material, such as large vision panels and glass doors.

Controlling the T 60 of a room yields the most dramatic results. Most medium size radio studios having a T 60 of approximately 0.3 to 0.4 seconds from 100 to 6 k Hz yield a pleasant acoustic environment. Unlike the massive solids that are used for acoustic isolation, the most common absorptive materials are light and porous. The most common materials that are commercially available for absorbing sound are carpeting, acoustic tile ceilings, and fiberglass or polyurethane foam wall panels. The difficulty in using these materials in broadcast studios, is that they provide only mid and high frequency absorption. Exclusive and excessive use of these materials can cause a studio to become "boomy", by causing it to have too long a T 60 at low frequencies in proportion to the short T 60 at the higher frequencies which they can absorb.

An acoustic consultant can specify the design of resonant slot, hole, and panel absorbers, as well as extra thick mineral fiber materials to absorb low frequency sound. Boominess can also be decreased by adding a thick fiberglass blanket above a lay-in tile ceiling; and using commercial absorptive materials in their thickest available form. Applying 3 or 4 foot widths of these materials with 2 or 3 foot spacings between them may also help balance the T 60s at low and high frequencies. However, to avoid reflective echoes, no hard untreated surface should ever oppose another either parallel or at an acute angle to it.

In combination control rooms, mechanical equipment such as broadcast cartridge machines and reel to reel tape decks should be surrounded with as much absorptive material as practical. This will help absorb some of their mechanical sounds which might otherwise be reflected toward the host's microphone. Also, try to avoid placing the console microphone position too close to a vision panel. Whether omnidirectional or cardioid, any conventional microphone requires a free field behind it; and will color the sound it picks up from the front, if a reflective surface is present behind it.

Large television studios often derive their acoustic characteristic more from their sets and backdrops, than from any materials purposely installed for acoustic purposes. Since any portion of a studio wall may be exposed at one time or another, it's a good idea to cover the walls with absorptive mineral batts. The batts should be protected by a wire mesh, to keep them from disintegrating when props and sets are stored up against them. The ceiling, above the lighting grid, should also be heavily absorptive, especially since large portions of the floor will remain reflective. It is important to keep the T60 of the television studio quite low to minimize the transmission of camera and crew noises, and to permit greater talent to microphone distances without an "off-mike" quality. Hard, concave, acoustically reflective sets should be avoided, since any combination of these tends to reflect unwanted sound toward the talent microphones in front of them.

The most reliable rule of thumb in acoustics is that treating low frequency problems is always more difficult and expensive than mid and high frequency work. If the budget is tight, always assign top priority to sound transmission considerations. Once the facility is built, little can be done to make up for economies made in the basic construction, while considerably more flexibility for improving interior room acoustics will remain.

TABLE 1. Reduction in the sound insulation, (*R*) of a partition due to acoustic leakage through various size openings.

Example: Partition with R = 60 dB

SIZE OF OPENING	RESULTANT R
0%	60 dB
0.1%	30 dB
1.0%	20 dB
10.0%	10 dB
50.0%	3 dB

ESTIMATING COSTS

During the last few years cost increases have been been continuous and are unpredictable. They must be examined carefully by competent architects and construction engineers for the specific structure and area involved, if there is to be any accuracy in budgeting the project.

Two major factors contributing to cost overrun on projects are inadequate initial plans and the resultant change orders during construction. Also, there must be an owner's supervisor highly involved in the project on a daily basis.

In each case, a general contractor was retained and coordination was handled by staff engineering personnel.

Electrical costs include power to all equipment, but not wages paid to staff technicians who installed the broadcast equipment.

Architectural and consulting fees are for outside help only. No attempt was made to estimate.

TOTALS

New studios and offices (complete new structure)	$82 to $138/sq. ft.
Conversion for studios & offices (in existing structure	$40 to $65/sq. ft.
Transmitter building (new structure)	$61 to $98/sq. ft.

SEPARATE ELEMENTS

Demolition and site work	$1.80 to $3.00/sq. ft.
Structural	$9.00 to $14.00/sq. ft.
General construction	$18.00 to $50.00/sq. ft.
Heating, ventilating and air conditioner	$9.00 to $16.00/sq. ft.
Electrical	$5.40 to $18.00/sq. ft.
Plumbing	$1.35 to $4.00/sq. ft.
Arch & Engineering fees	$2.25 to $7.00/sq. ft.
Special acoustical	$2.70 to $5.00/sq. ft.
Woodwork and built-ins	$1.08 to $3.00/sq. ft.
Office furniture	$2.70 to $7.00/sq. ft.
Decoration	$3.60 to $7.00/sq. ft.

Note: Electronic equipment installation and wiring not covered by any of above. A good general rule is to allow 15 to 25 percent for this in addition to basic equipment cost. Variation is due to location, personnel, working rules, etc.

CONSTRUCTION EXAMPLES*

The examples that follow illustrate construction for TV facilities. The stations are in large cities, with construction being completed between 1980 to 1983.

**Note:* See Tables 2 and 3 at end of chapter for labor costs and conversion factors.

EXAMPLE A

BOSTON:

There were three basic requirements which the addition had to meet.

1. Provide Corporate and Television Division offices
2. Provide expansion for the WCVB-TV News.
3. Provide a studio and ancillary facilities capable of producing syndicated shows as well as increased local station production.

   The available area to build was limited since the property was divided into two types of zoning. This required the expansion to be planned for the area in which the zoning would allow such construction. We were able to meet the codes for parking, but the lack of Public Transportation presented a requirement of additional parking which has not totally been met.

   The timetable for the construction of the addition was dictated by the fact that the corporate and Television Division personnel were sharing space with the station. This, as well as wanting to enclose the building before winter set in, pretty much established the schedule of construction. For all practical purposes the construction took approximately 9 months not including engineering and construction drawings.

NOTE: See Fig. 3 & 4

EXAMPLE A (Cost)

New Building—Studio
25,000 sq. ft.

New England Area (Boston)

Demolition & Site	$ 111,000.	/$ 4.44
Structural	255,000.	/$10.20
General Const.	1,694,600.	/$67.68
Decoration	93,700.	/$ 3.75
Heating/Ventilating	315,000.	$12.60
Electrical	225,000.	/$ 9.00
Plumbing	119,000.	/$ 4.76
Architect, Elect. & Mechanical Engrg.	396,000.	/$15.84
Special Woodwork & Built-ins	55,000.	/$ 2.20
Special Acoustical (incl. ceilings)	172,300.	/$ 6.89
		$137.46

NOTE: See Fig. 3 & 4

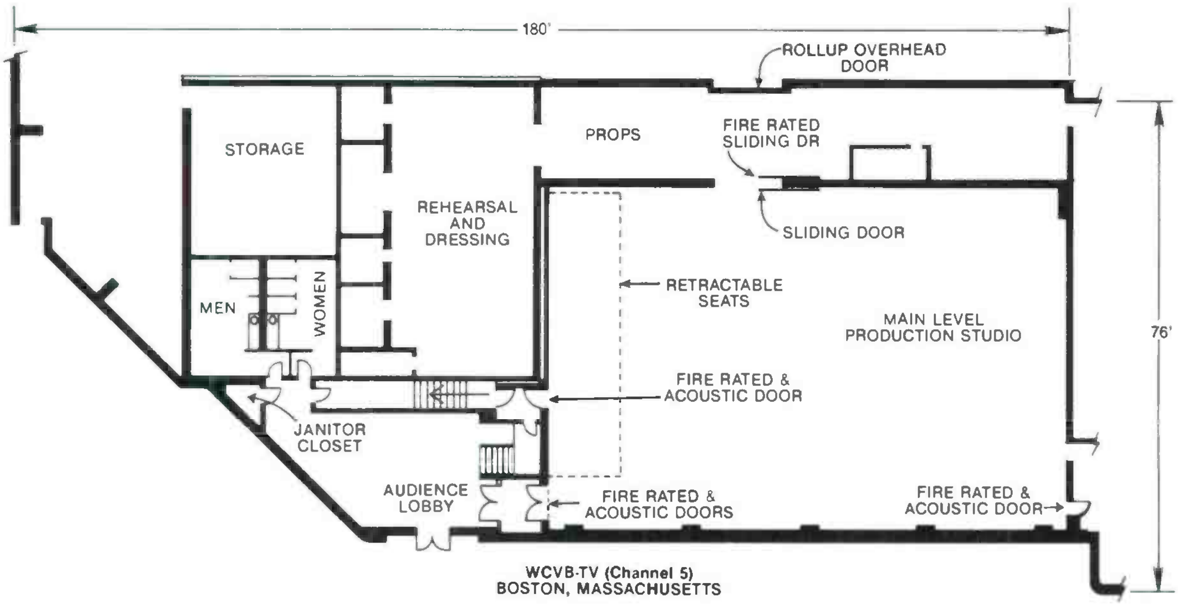
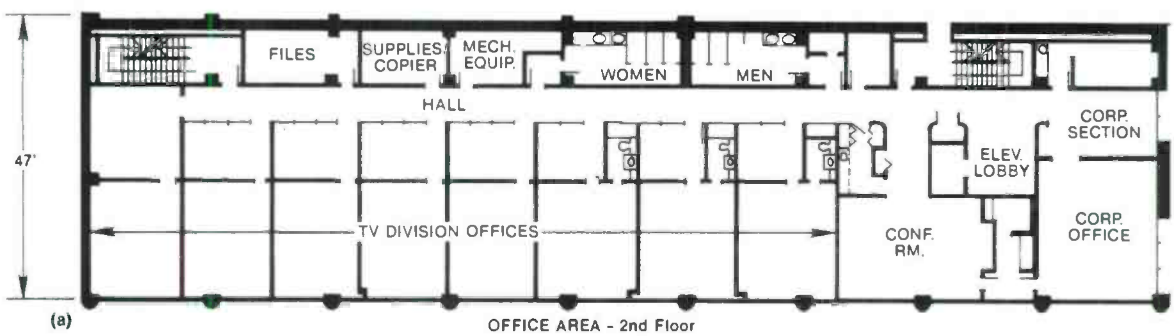


Fig. 3. Production studio addition.



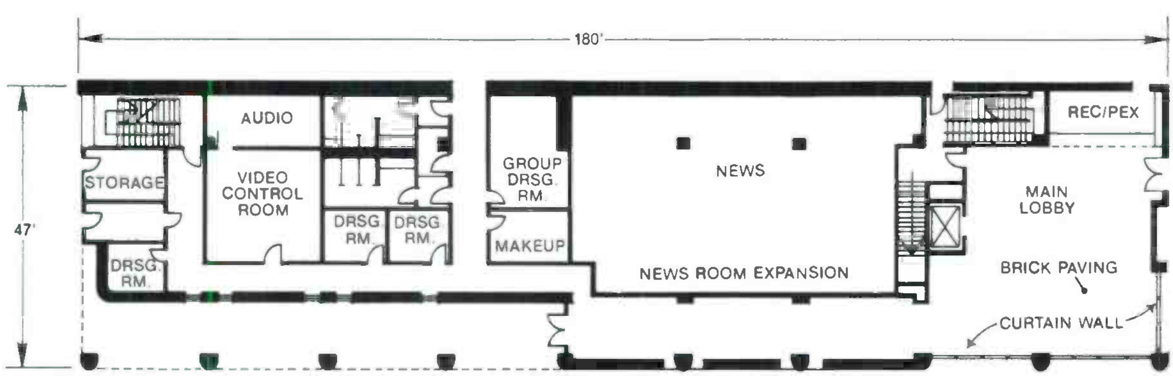


Fig. 4.

EXAMPLE B

HOUSTON:

In relocating the Houston transmitter, we set the following goals:

1. Relocate from downtown site which was being surrounded by buildings taller than our transmitting antenna.
2. Locate in same general area as the existing Houston TV stations.
3. Ideal was to have an antenna height of 2000 feet AMSL.
4. Provide a Principal City Coverage over the same area as the old transmitter site.

We found a site and did get FAA approval for a 2000 foot AMSL antenna height. However, the site did present some problems as it was located on top a salt dome. The salt dome in itself created enough concern that it could require a chapter of its own. Suffice it to say after some lengthy research and a design by a well-known structural engineering firm we erected the tower or should I say towers as there were two towers 100 feet apart. The foundation design was such that the tower foundations as well as the transmitter buildings were all a part of one foundation.

The transmitter and antenna were selected on the basis of a practical consumption of primary power and with transmission line loss and antenna gain would give 5 Megawatts radiated power.

NOTE: See Fig. 5 & 6

EXAMPLE B (Cost)

New Building—Transmitter
2,300 sq. ft.

Gulf Coast Area (Houston)

Demolition & Site		
Structural	$ 29,000.	/$12.72
General Const.	77,000.	/$33.77
Decoration	6,000.	/$ 2.63
Heating & Ventilating	15,000.	/$ 6.59
Electrical	67,898.	/$29.78*
Plumbing	10,000.	/$ 4.39
Architect, Elect. & Mechanical Engrg	14,730.	/$ 6.46
Special Woodwork & Built-ins	4,000.	/$ 1.75
Special Acoustical (incl. ceilings)		
		$98.09

NOTE: See Fig. 5 & 6

*Includes Auxiliary Generator, Pad and Roof

AM AND FM STUDIO FACILITIES

Building Planning

One of the prime requisites for a successful broadcast station is the careful layout of studio, production, and administrative areas to achieve maximum effectiveness of space and personnel. The following are four typical layouts depicting a small market minimum staff facility to an arrangement suitable for a large metropolitan operation employing a full complement of personnel. Each floor plan is handled differently according to the needs of different size stations.

Control room, studio, and production facilities for each station are in a centrally located CORE AREA. The suggested sizes of these areas should be considered as minimum from an operating standpoint with normal equipment complement. The layouts are presented as a guide for planning a modern, functional radio facility with considerations given to size of market, staff and programming requirements.

Plan One

With approximately 1,800 sq. ft., this floor plan provides adequate space for the small AM or FM station with a minimum staff. Since smaller staffs have several responsibilities, partitioned general office space is omitted in favor of a large news, transcription storage, and general-use area at the rear of the building.

The transmitter or workshop area is next to the control room, with a window recommended for a clear view of the transmitter meters. Alternate CORE AREA layouts are shown.

The building is of brick and plaster fascia, and includes a glass curtain wall. The building price will vary considerably depending on area construction costs; but a typical range is $60,000 to $75,000 (See Fig. 7).

Plan Two

In medium size stations, office space requirements for sales, promotion, and programming activities exceed the need for a substantially larger technical CORE AREA. This floor plan expands the "small station" layout to approximately 2,500 sq. ft., providing more room for the sales staff and clerical help, and an impressive office for the general manager. Studio and control room space is slightly larger in anticipation of more equipment and activities in these areas. The news director, transcription library, and chief engineer gain office space. Alternate CORE AREA layouts may be employed, and a few suggestions are indicated.

The building includes brick walls, weathering steel columns, and fascia, with dark glass entrance and glazing strips. Cost is approximately $65,000

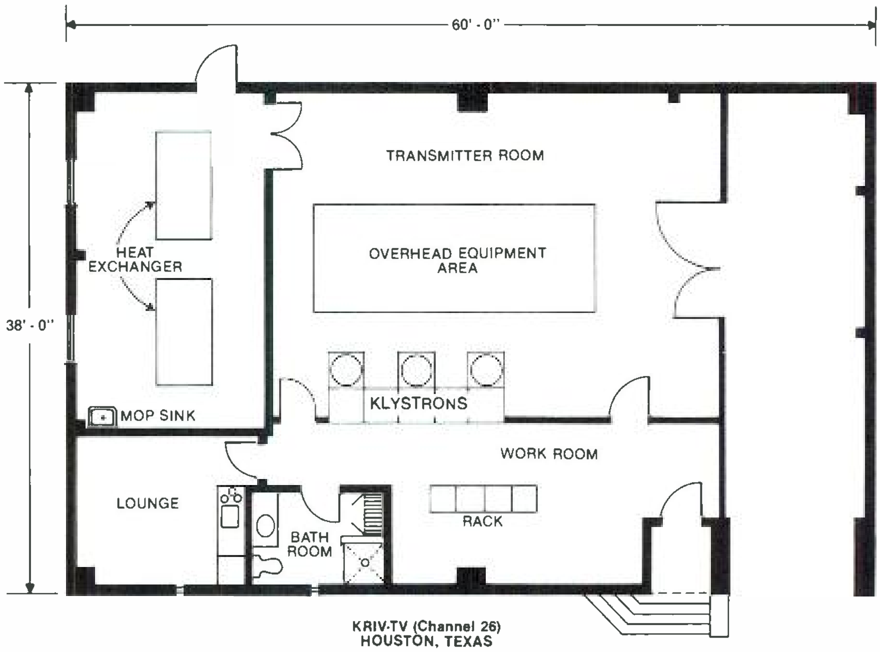


Fig. 5. Transmitter building.

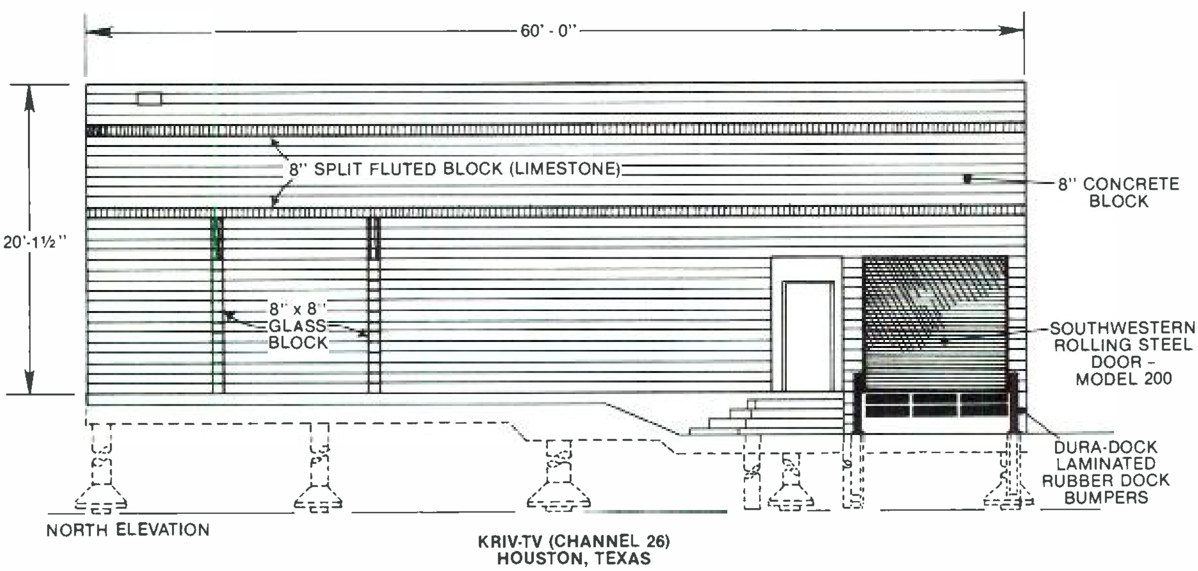


Fig. 6. Transmitter building.

to $90,000, but may vary considerably, depending on construction costs in your area. (See Fig. 8)

Plan Three

In Plan 3, the technical CORE AREA is adequate for two full-size control rooms, each with a large associated studio. Control rooms are separated by the transmitter, automation, or workshop area. This floor plan includes approximately 3,150 sq. ft. and is suggested for stations planning both AM and FM operations. Additional office space is also allocated for the larger staff in this station. See alternate CORE AREA floor plans for additional layout ideas. The mirror glass curtain wall building is set in a reflecting pool and costs approximately $90,000 to $135,000. Note: building cost may vary greatly, depending on the area in which it is built. (See Fig. 9)

Plan Four

This 4,300 sq. ft. studio/office complex is an impressive broadcast center. Of primary importance is the location of all control room and studio space in the center of the building, eliminating the problem of outside traffic noise in a metropolitan area.

Operating personnel are assigned to the rear office areas, and the news room is strategically located near the control rooms and an outside exit to the newsmobiles.

The building is of exposed concrete, with a dark glass curtain wall. Building price is approximately $100,000 to $150,000, but may vary greatly from area to area, depending on material and labor costs. (See Fig. 10)

CONTROL ROOM ANNOUNCE BOOTH DESIGN

Many present-day radio control rooms are not only used for this purpose but also act as announce booths where most of a station's announcing is carried on. In many cases the control room may be the only studio area in a station, and it will contain all the audio equipment of the station. Frequently, a transmitter and record library will also be located in the control room. This makes the problem of acoustics more complex but since radio sells with sound, good acoustics are a must when the best sound possible is desired.

Design Factors of combination control rooms are at best a compromise of the several design factors. In the design of this control room, consideration must be given to the following:

1. Location of the control room within the studio building;
2. Isolation: elimination of unwanted sound and noise, both internal and external;
3. Construction: dual wall, floating wall, single wall;
4. Reverberation control, elimination of unwanted reflections, and floor treatment;
5. Ventilation and air conditioning;
6. Size and arrangement of equipment;

Note: See section on ACOUSTICS in this chapter for more detail.

RADIO STUDIOS ON A LOW BUDGET

Considerations Involved in Building an AM or FM Station with Less Than an Optimum Budget

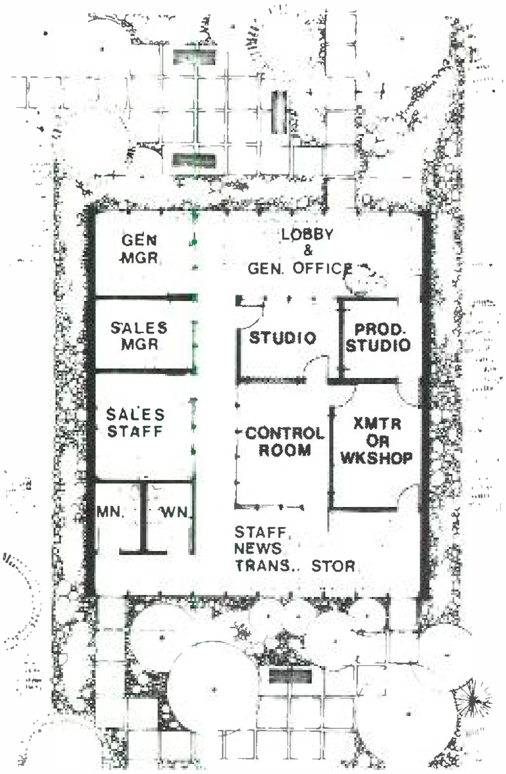
It would be easy to apply one set of standards for the construction of radio station facilities everywhere in the country. Unfortunately, the difference in cashflow of a 50,000-watt clear channel in a major market and a 250-watt daytime or Class A FM in a rural area dictate that the small market station is going to be quite different than its big city counterpart. In most cases, the selection of a studio site in a small market is dictated by what it costs to get the space. It is not unusual for space to be traded out in part or in full for advertising.

In many cases, the chief engineer will be presented with an existing suite of offices, a store front or even an older house that must be converted to a studio. The first thing to do in this situation is call a meeting of management, sales, and programming to see what they expect of the facility. If it is a typical small market station, it will fall within the following requirements:

1. Record shows: combination operation. Announcer playing records, taped commercials.
2. Direct airplay or recording beeper reports for later use in newscasts, farm reports, high school news.
3. Facility for picking up remotes from high school or college athletic events, church remotes.
4. Capacity of recording and dubbing commercials for later airplay.
5. Capacity for originating a live music program from your studio. (Quite often a church will use a studio to do a small service rather than invest in phone lines and remote equipment!)
6. Delayed programming such as a telephone talk, call in forum or swap shop show.
7. Remote off-air pickup of other AM and FM stations for rebroadcasting regional networks.

SMALL MARKET MINIMUM STAFF

Fig. 7. Small size AM station equipment list (Plan One).



Studio equipment (monophonic):

1	8-Mixer Console, mono
2	12" Turntable
2	Integrated Circuit Equalized Preamplifier, mono
2	12" Tone Arm
2	Stereo Cartridges (Outputs connect in parallel for monaural operation)
3	8'' Loudspeaker
3	Speaker Matching Transformer
3	Wall cabinets for Speaker
1	Cardioid Microphone
2	Dynamic Omnidirectional Microphones
1	Boom Stand
1	Desk Stand
1	Clamp-on Mike Stand
4	Wall Receptacles for Microphone
4	Microphone Plugs
4	Connectors
100'	2-Conductor #20 Microphone Cable, jacketed
500'	Shielded Miniature Audio Cable
1	Headphone
1	Phone Plug for Headphone
2	Studio Clocks

Studio equipment options (stereophonic)

1	Stereo Console
2	Stereo Equalized Preamplifier

Remote Broadcast Equipment:

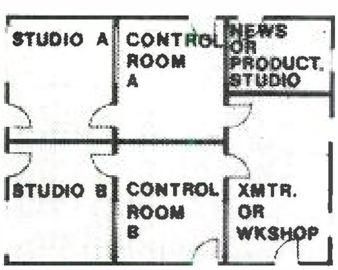
1	4-mixer Transistor Amplifier, Less Batteries
1	Battery Kit
1	Headphone
1	Plug for Headphone
1	Dynamic Microphone with 18 Ft. Cable
1	Plug for Microphone

Tape Recording Equipment:

1	Half-Track Portable Tape Recorder, 7½'' per sec.
2	Cartridge Playback, mono
1	Recording Amplifier for above, mono
24	Cartridges, 40 Sec.
24	Cartridges, 70 Sec.

Remote Control Equipment for Unattended Operation:

1	Remote Control System for unattended operation
1	RF Amplifier with Antenna
1	Rack Cabinet



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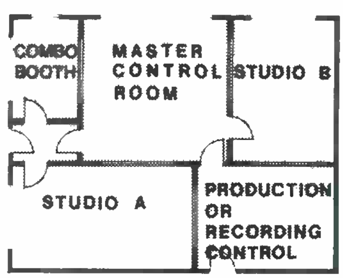

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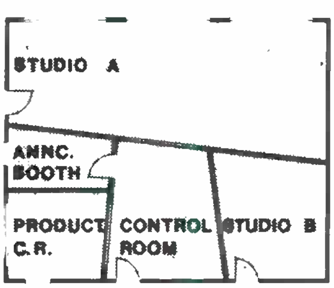

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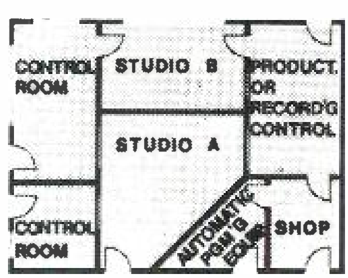


D

alternate "core area" plans

MEDIUM MARKET NORMAL STAFF

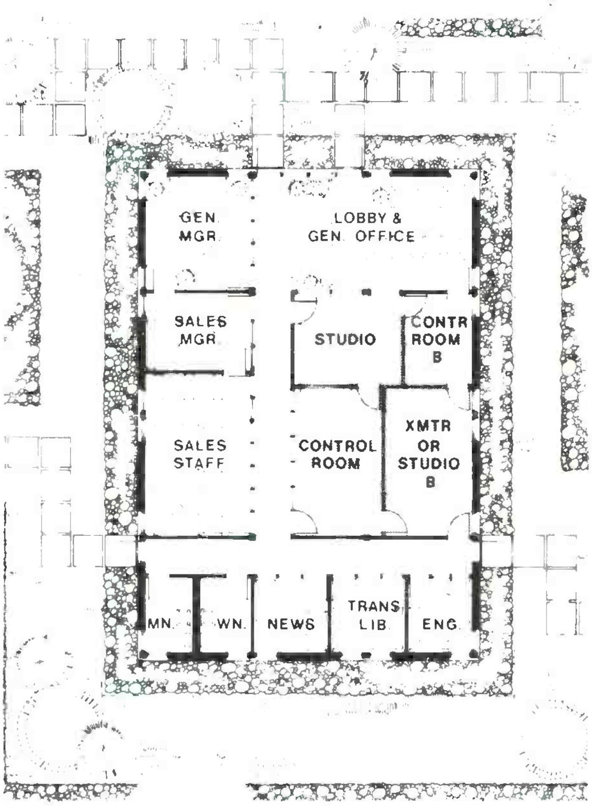


Fig. 8. Medium size AM station equipment list (Plan Two).

Studio Equipment:

1	8-Mixer Console, Monaural
2	12'' Turntables
2	Integrated Circuit Turntable Preamplifier
2	12'' Tone Arm
2	Stereo Cartridge (Outputs connected in parallel for monaural operation)
3	8'' PM Type Loudspeaker
3	Speaker Matching Transformers
3	Wall Cabinets for Speakers
1	Cardioid Microphone
2	Dynamic Omnidirectional Microphones
1	Boom Stand
1	Desk Stand
1	Clamp-on Mike Stand
4	Wall Receptacles for Microphone
4	Microphone Plugs
4	Connectors
100'	2-Conductor #20 Microphone Cable, Jacketed
500'	Shielded Miniature Audio Cable
1	Headphone
1	Phone Plug for Headphone
2	Studio Clocks.

Remote Broadcast Equipment:

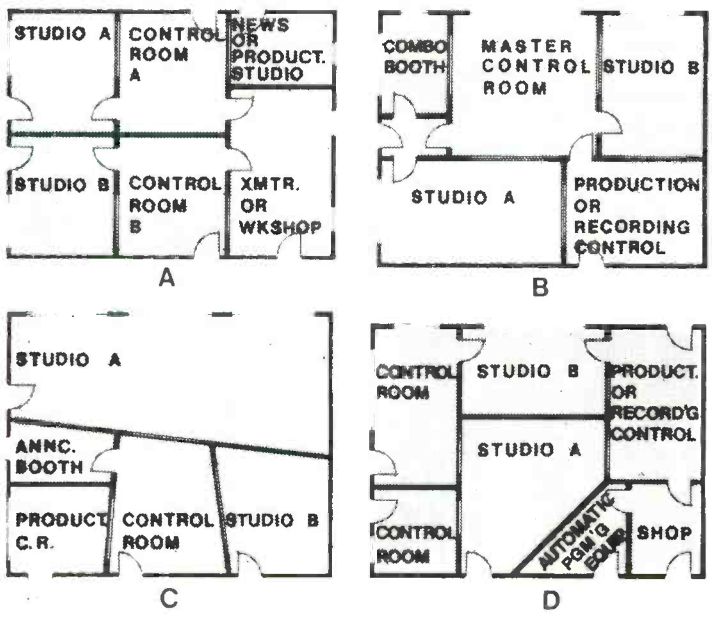
1	4-Mixer Transistor Amplifier, less batteries
1	Headphones
1	Phone plug
1	Omnidirectional Microphone

Tape Recording Equipment:

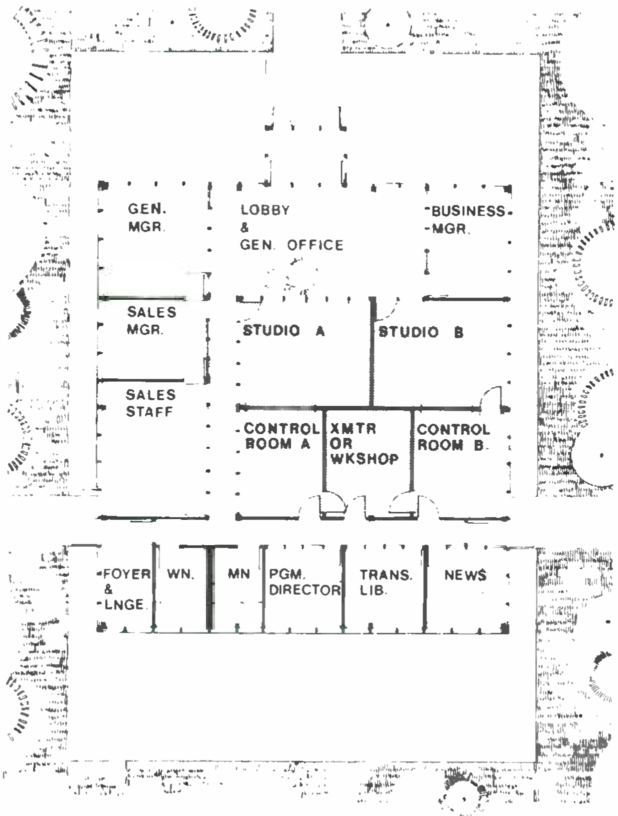
1	Half-track Portable Tape Recorder, 7½ in. per sec.
1	Cartridge Playback Unit and Recording Amplifier, one tone, desk mount
1	Cartridge Playback Unit, Desk Mount
24	40 Sec. Cartridges
24	70 Sec. Cartridges

Remote Control Equipment for Unattended Operation:

1	Remote Control System for unattended operation of transmitter
1	RF Amplifier with antenna
1	Rack Cabinet.



alternate "core area" plans



AM-FM OR DUAL CONTROL ROOM OPERATIONS

Fig. 9. AM/FM dual control room equipment list (Plan Three).

Transmitter Audio Equipment:

1	Rack Cabinet
1	AM Solid State Limiting Amplifier
1	FM Solid State Limiting Amplifier

Monitors:

1	AM Modulation Monitor, Solid State
1	EBS Monitor
1	FM Modulation Monitor

Studio Equipment

1	8-Mixer Console, Monaural
2	12'' Turntables
2	Integrated Circuit Turntable Preamplifier
2	12'' Tone Arm
2	Stereo Cartridge (Outputs connected in parallel for monaural operation)
3	8'' PM Type Loudspeaker
3	Speaker Matching Transformer
3	Wall Cabinets for Speakers
1	Cardioid Microphone
2	Dynamic Omnidirectional Microphones
1	Boom Stand
1	Desk Stand
1	Clamp-on Mike Stand
4	Wall Receptacles for Microphone
4	Microphone Plugs
4	Connectors
100'	2-Conductor #20 Microphone Cable, jacketed
500'	Shielded Miniature Audio Cable
1	Headphone
1	Headphone Plug
2	Studio Clocks.

Remote Broadcast Equipment

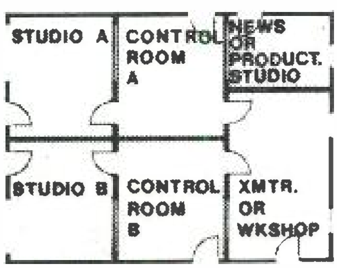
1	4-Mixer Transistor Amplifier, less batteries
1	Headphones
1	Phone Plug
1	Dynamic Omnidirectional Microphone

Tape Recording Equipment

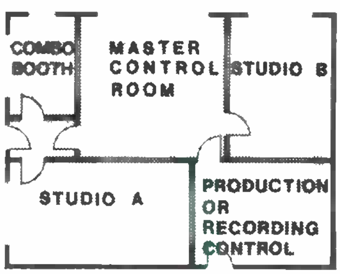
1	Half-track Portable Tape Recorder, 7½ in. per sec.
1	Cartridge Playback Unit and Recording Amplifier, one tone, desk mount
1	80 Cartridge Playback Unit, Desk Mount
24	40 Sec. Cartridges
24	70 Sec. Cartridges

Remote Control Equipment for Unattended Operation

1	Remote Control System for unattended operation of transmitter.



A



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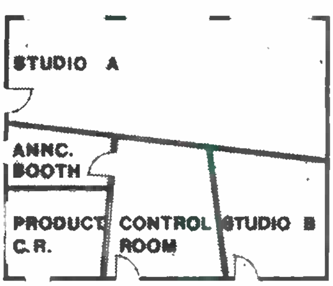

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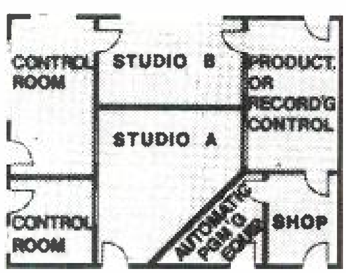


D

alternate "core area" plans

METROPOLITAN MARKET FULL STAFF

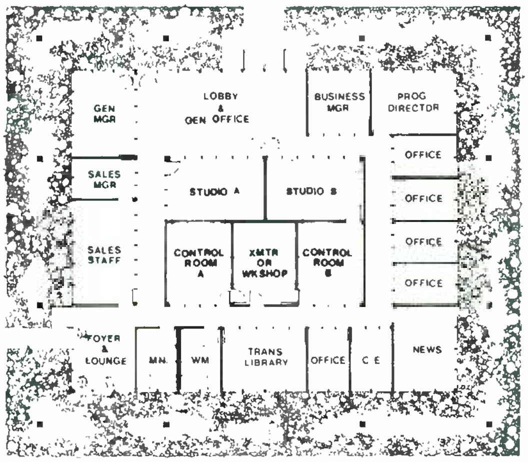


Fig. 10. Metropolitan market full staff (Plan Four).

Transmitter Audio Equipment (monophonic)
- 1 Rack Cabinet
- 1 FM Solid State Limiter

Monitors
- 1 FM Frequency Measuring Unit
- 1 FM Modulation Monitor
- 1 EBS Monitor

Stereo Options
- 1 Stereo Generator
- 2 FM Solid State Limiters, matched
- 1 Stereo Modulation Monitor
- 1 19 kHz Pilot Frequency Comparator

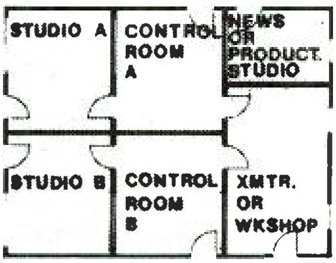
Studio Equipment (monophonic)
- 1 8-Mixer Console, mono
- 2 12'' Turntable
- 2 Equalized Preamplifier
- 2 12'' Tone Arm
- 2 Stereo Cartridge (Outputs connect in parallel for monaural operation)
- 3 8'' Loudspeaker
- 3 Speaker Matching Transformer
- 3 Wall Cabinets for Speaker
- 1 Cardioid Microphone
- 2 Dynamic Omnidirectional Microphones
- 1 Boom Stand
- 1 Desk Stand
- 1 Clamp-on Mike Stand
- 4 Wall Receptacle for Microphones
- 4 Microphone Plugs
- 4 Connectors
- 100' 2-Conductor #20 Microphone Cable, jacketed
- 500' Shielded Miniature Audio Cable
- 1 Headphone
- 1 Phone Plug for Headphone
- 2 Studio Clock

Studio Equipment Stereo Options
- 1 Stereo Console
- 2 Stereo I.C. Equalized Preamplifier

Remote Broadcast Equipment
- 1 4-mixer Transistor Amplifier, Less Batteries
- 1 Battery Kit
- 1 Headphone
- 1 Plug for Headphone
- 1 Dynamic Microphone with 18 Ft. Cable
- 1 Plug for Microphone

Tape Recording Equipment
- 1 Half-Track Portable Tape Recorder, 7½'' per sec.
- 2 Cartridge Playback, mono
- 1 Recording Amplifier for above, mono
- 24 Cartridges, 40 Sec.
- 24 Cartridges, 70 Sec.

Remote Control Equipment for Unattended Operation
- 1 Remote Control System for unattended operation
- 1 RF Amplifier with Antenna
- 1 7 Rack Cabinet



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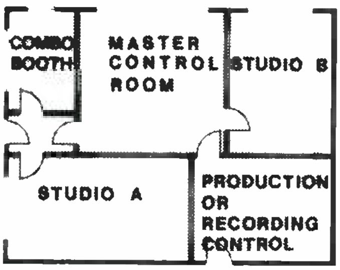

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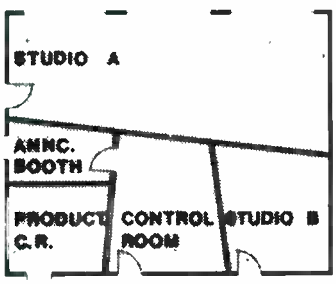

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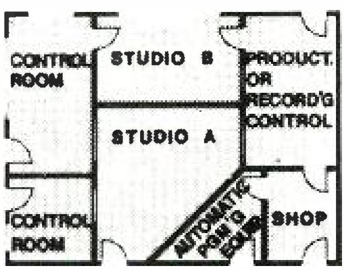

D

alternate "core area" plans

8. Remote pickup of mobile units and portable transmitters for news actualities and other events.

If the station is a new or growing operation and cannot afford the equipment to do all of the above, at least consider what functions may be required at a future date. For this reason plan now for what might be needed in two years.

Physical Layout

This will depend largely on what is available. The bare bones minimum control room known to be used was 6 ft. wide by 7 ft. deep. There was barely enough room for an operator, two turntables, cartridge machines, and a console. It worked.

The operation was well constructed, well maintained and well utilized. The technical quality of the programming was as good as many large operations in major markets, so it can be accomplished with small space.

When laying out studio location, do the following:

1. Draw a sketch of the available floor space and existing walls.
2. Make a number of copies of this floor plan and start drawing in studios, offices, reception areas and the like. Make three or four.
3. The first items you want to place are the main studio and control room. In smaller operations these are one and the same. Naturally they must be quiet, so keep them away from noisy areas such as underneath heavily traveled stairways, air conditioners, and front windows opening on busy streets. The control room floor should be very solid. The most desirable material to use if you are building a studio is reinforced concrete. If you are stuck with a wooden floor try and select a location that does not bounce. People walking on such a floor will cause the floor to move. This motion is transmitted to the turntable cabinet to the tone arm where it is picked up and sent over the air. In extreme cases, a heavy person can cause the needle to jump out of the groove. Sandbags or bricks in the base of the turntable cabinet will sometimes improve a bad situation.

   Another item that will determine placement of the control room is visibility of the meters on the transmitter and its associated monitoring equipment. FCC rules and regulations have recently changed regarding this requirement and undoubtedly will change in the future. Check the rules before you build. If you are fortunate enough to be putting in your own walls, stay away from designing perfectly square rooms. A room that is 8 by 8 by 8 ft. is going to have a very definite resonance. Strive for 6 by 8 by 10 ft. or dimensions in this proportion. Sound locks while desirable are not essential. Doors of solid construction with weatherstripping can be used effectively.
4. Measure some typical office furniture and cut out cardboard desks, file cabinets, counter tops and record cabinets to scale. Put in your furniture and see how it fits. Is there room for people to move around? Did you leave room for doors to open and close? Remember not to have doors open out into busy hallways where people will walk into them as they pass by. Doors come in all sizes. The average door in most stations is 3 ft. wide. If you get into a space problem this can be reduced by 6 inches or even a foot. But remember that you will be moving furniture and equipment in and out of that door and you should provide room for it to go in and out and be turned once it's in the room. The narrowest hallways should be no less than 3 ft. 6 in., preferably 4 ft. Also be aware that hallways take up space. For every square foot of hallway you eliminate you get an equal amount of space that can be used for something productive.

Equipment

In a low-budget situation remember this rule of thumb, "Is what I am about to buy going to pay for itself and make the station money?"

The purchase of a $200 directional microphone to do a one time remote broadcast when being paid only $100 for the whole job is poor business.

Use this logic when equipping the station. As mentioned earlier, create a list of those functions you will have to perform, spend only what is needed to get on the air and function. However, do plan for the future. Design around a console with enough inputs to accommodate these anticipated needs. A four potentiometer board is not enough for most operations. Even the smallest stations should have six or eight pots. A four pot board will be satisfactory today but two years later it will not. If the station is a one studio operation, how will the console be replaced while on the air? It makes good sense to purchase the proper console in the beginning.

Do not forget the patch panel. There is a growing tendency among engineers today to eliminate this important switching center. All high level inputs and outputs should appear here. Even console inputs should show up. This also applies to all recorders, tuners, and other audio sources. This practice makes for a very versatile operation and can save the embarrassment of dead air time if the console should fail. If this happens,

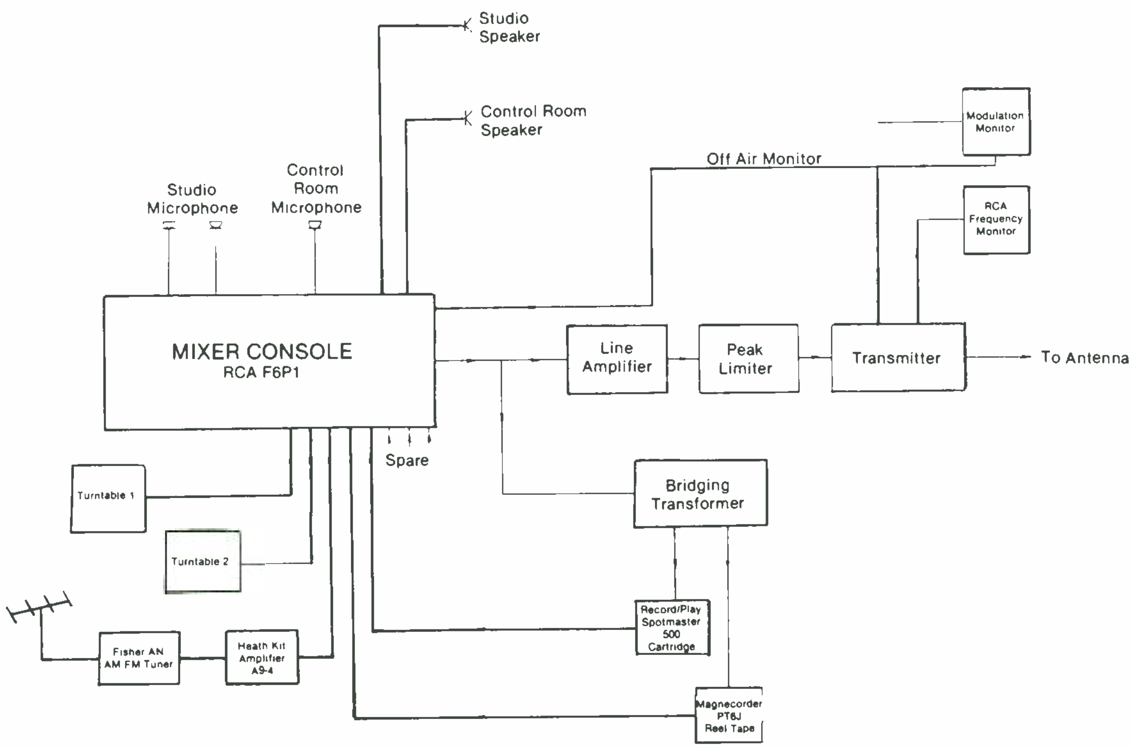

Fig. 11. Audio flow diagram from station WBME-AM.

the recorder can be patched directly to the transmitter, bypassing the defective elements in the system.

Fig. 11 shows the audio flow in a typical small market radio station with the bare essentials. This was installed at WBME-AM in Belfast, Maine. It should be considered a minimum installation. One cartridge tape machine, one reel tape, audio processing equipment and a transmitter. This system can be expanded to include other program sources as the station's needs grow.

Figs. 12 and 13 show the audio flow and physical layout of a Class C FM station in Topeka, Kansas. Note the versatility here. The studio can be operated through the automation system as a source. In the event of an emergency, the announcer can remotely connect the output of the console to the STL, bypassing the automation system. In the event of an STL failure, the audio can be patched directly to the transmitter via the remote control system line.

In selecting equipment for a small station, one may consider used equipment. U.S. made broadcast equipment is built to last. It is not unusual to see transmitters in regular use after 30 years. The same can apply for consoles, amplifiers, and some other audio equipment. However, if you purchase something this old, the chances are that the manufacturer will no longer be able to supply spare parts. Most equipment suppliers will maintain spares for 20 to 25 years. Do not get the impression that spare parts are no longer made, they just are not stocked by the equipment manufacturer. In many cases, it will be necessary to seek out the item wanted from the original component manufacturer.

Special items such as modulation and plate transformers can be rebuilt or fabricated; it can be costly but it can be done.

When buying a piece of used equipment, try and get the previous owner to throw in his spare tubes and other parts. Also obtain a history of the unit and where parts can be obtained. It is also advisable to find other owners of the same type of unit.

As mentioned previously in some instances, the licensee either has an available structure or wishes to take advantage of an existing structure to use as the nucleus of his broadcase facility. In such cases, the existing structure is usually a house or a desirable piece of property that can be easily expanded or converted into the studio/ transmitter building.

An excellent example of this concept is station WNHV, White River Junction, Vermont, that expanded upon an existing building to develop a very efficient and workable broadcast facility. Fig. 14 depicts the changes which were made to the existing building with the resulting added floor space.

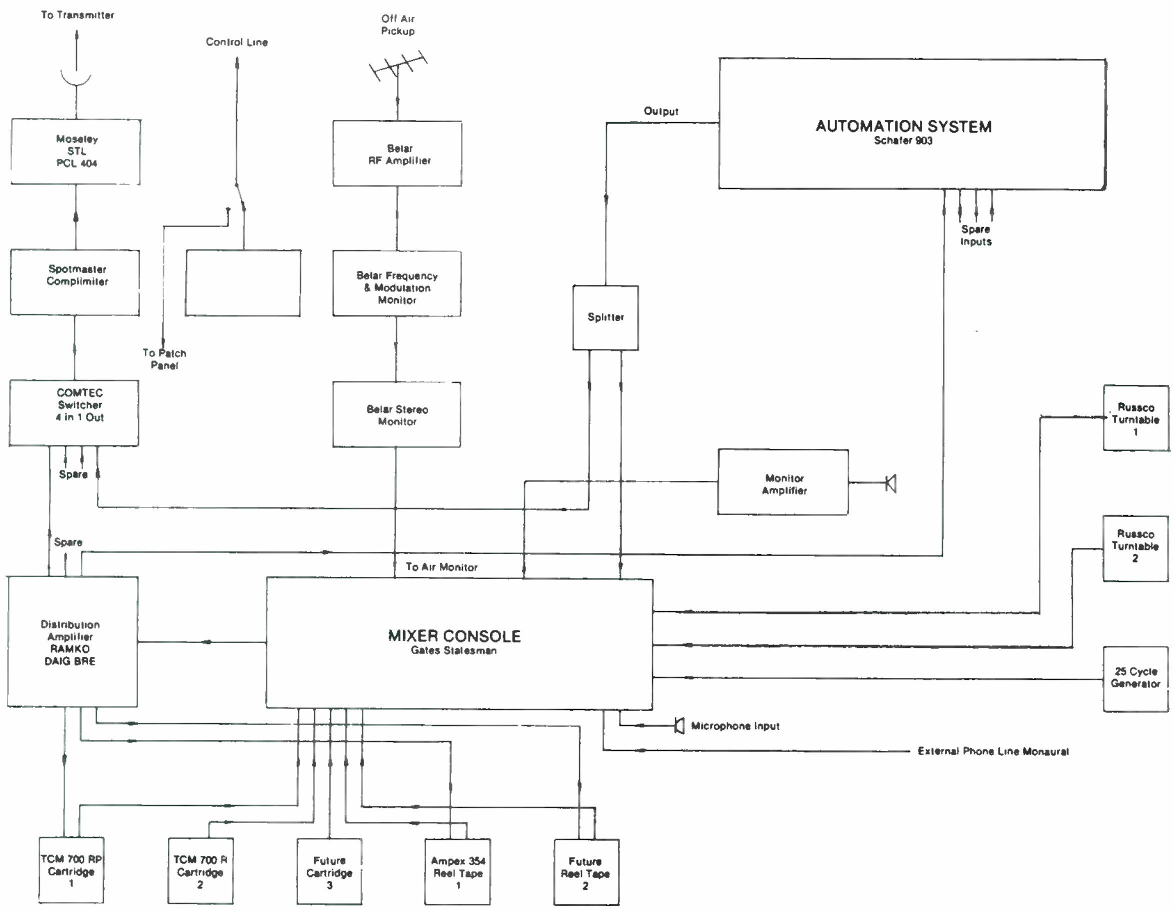

Fig. 12. Audio flow diagram from station KTPK-FM.

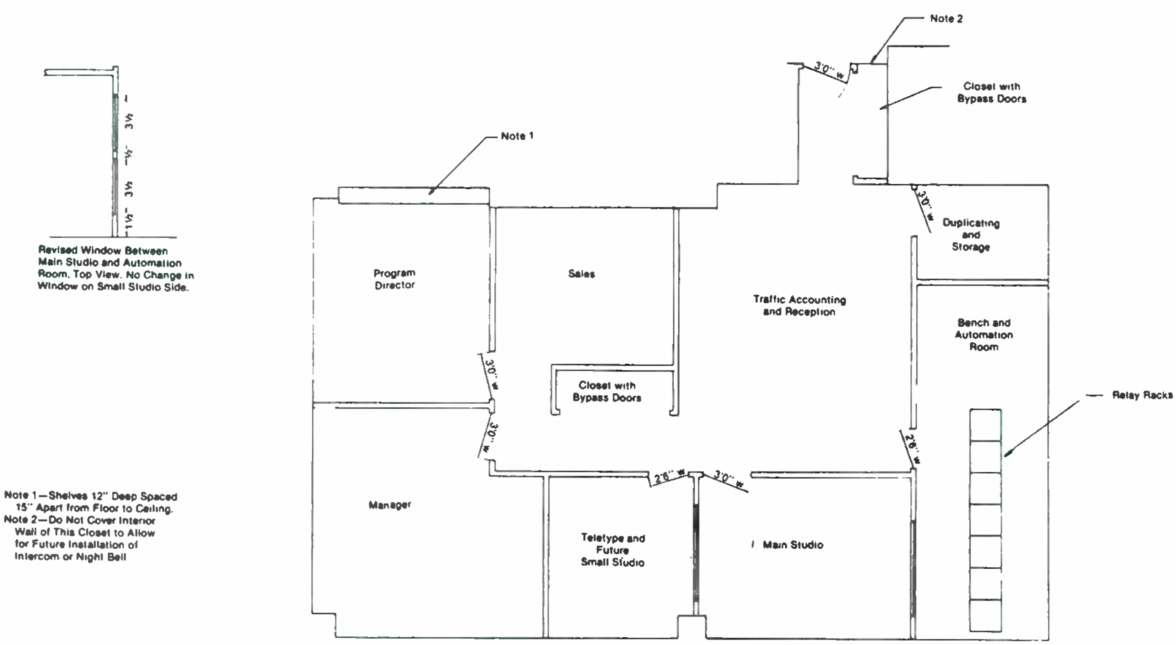

Fig. 13. Studio layout for station KTPK-FM, Topeka, Kansas.

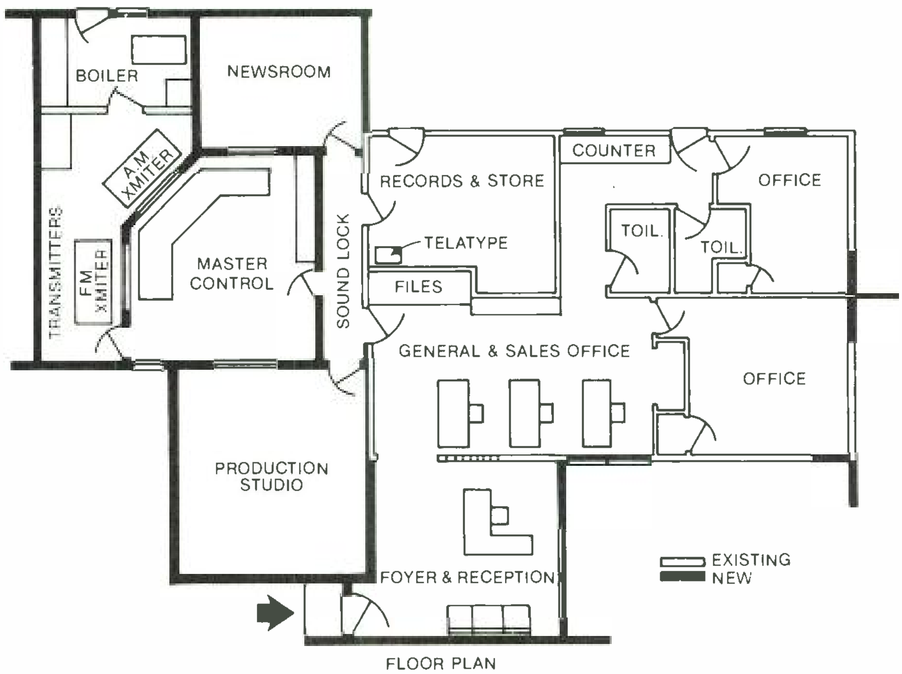

Fig. 14. Additions and alterations to WNHV radio station, White River Junction, Vermont.

The nucleus of this building was a nondescript, small house which was ultimately camouflaged behind additions and beneath new materials. Much attention was given to sound conduction, natural and artificial lighting, and controlled ventilation. Aluminum was used extensively throughout in conformance to the licensees needs. The modification added a total of 1,120 sq. ft., compared to 864 sq. ft. in the existing structure. Cost in 1984 would be between $20,000 and $35,000.

**TABLE 2. Industrial and commercial division*—
hourly labor costs.**

The hourly labor costs shown in the column headed "Hourly Cost Including Subcontractor's 30% Markup" have been used to compute estimates in the "Labor" column on pages 212 to 457 of this book. All figures are in U.S. dollars per hour.

"Hourly Wage and Benefits" includes the wage, welfare, pension, vacation, apprentice and other mandatory contributions. The "Employer's Burden" is the cost to the contractor for Unemployment Insurance (F.U.T.A.), Social Security and Medicare (F.I.C.A.), state unemployment insurance, worker's compensation insurance and liability insurance. Tax and insurance expense included in these labor-hour costs are itemized in the sections beginning on pages 103 and 191.

These hourly labor costs will apply within a few percent on many jobs. But wages may be much higher or lower on the job you are estimating. If the hourly cost on this page is not accurate, divide your known or estimated cost per hour into the listed cost per hour. The result is your adjustment for any figure in the "Labor" column for that trade.

For example, the hourly cost (including 30% subcontractor's markup) for a carpenter is given as \$32.08. Assume your carpentry subcontractor can be expected to charge 10% more, \$35.29 per hour. Your adjustment is plus 10%. On page 285 the cost of laying 3/8" CD plywood roof sheathing is listed at \$278. Your cost would be 10% more, \$305.80

If you don't know what wage rates will apply on the job being estimated, use the table that begins on the following page. It shows estimated union wages in over 100 U.S. cities in percent of the wage listed below.

Trade	Hourly Wage and Benefits	Typical Employer's Burden (%)	Employer's Burden ($) Per Hour	Hourly Cost	Hourly Cost Including Subcontractor's 30% Markup
Air Tool Operator	16.64	25.8	4.29	20.93	27.21
Asbestos Worker	20.05	27.4	5.49	25.54	33.20
Boilermaker	21.98	20.9	4.59	26.57	34.54
Bricklayer	19.32	22.6	4.37	23.69	30.80
Bricklayer's Tender	14.61	22.6	3.30	17.91	23.28
Building Laborers	15.57	23.2	3.61	19.18	24.93
Carpenters	19.56	26.2	5.12	24.68	32.08
Cement Masons	19.11	20.2	3.86	22.97	29.86
Crane Operators	19.99	25.2	5.04	25.03	32.54
Drywall Installers	20.25	21.2	4.29	24.54	31.90
Drywall Tapers	17.14	21.2	3.63	20.77	27.00
Electricians	22.43	18.7	4.19	26.62	34.61
Elevator Constructors	21.24	18.2	3.87	25.11	32.64
Floor Layers	22.26	20.0	4.45	26.71	34.72
Glaziers	19.52	24.9	4.86	24.38	31.69
Iron Workers (Structural)	20.89	36.8	7.69	28.58	37.15
Lathers	19.42	20.6	4.00	23.42	30.45
Marble Setters	23.30	22.6	5.27	28.57	37.14
Millwrights	20.10	26.2	5.27	25.37	32.98
Mosaic & Terrazo Workers	17.73	20.0	3.55	21.28	27.66
Painters	19.08	23.0	4.39	23.47	30.51
Pile Drivers	18.73	31.6	5.92	24.65	32.05
Pipefitters	22.11	19.9	4.40	26.51	34.46
Plasterers	18.97	25.8	4.89	23.86	31.02
Plasterer's Helper	14.28	25.8	3.68	17.96	23.35
Plumbers	21.64	19.9	4.31	25.95	33.74
Reinforcing Ironworkers	20.53	21.5	4.41	24.94	32.42
Roofers, (Composition)	18.12	39.0	7.07	25.19	32.75
Sheet Metal Workers	21.93	21.0	4.60	26.53	34.49
Sprinkler Fitters	21.70	19.9	4.32	26.02	33.83
Tractor Operators	19.87	20.0	3.97	23.84	30.99
Truck Drivers	15.98	26.2	4.19	20.17	26.22

TABLE 3.

LONG MEASURE

12 inches	1 foot
3 feet	1 yard
5-1/2 yards	1 rod
40 rods	1 furlong
8 furlongs	1 sta. mile
3 miles	1 league

SQUARE MEASURE

1 sq. centimeter	0.1550 sq. in.
1 sq. decimeter	0.1076 square feet
1 sq. meter	1.196 sq. yd.
1 acre	3.954 sq. rods
1 hectare	2.47 acres
1 sq. kilometer	0.386 sq. mile
1 sq. inch	6.452 sq. centimeters
1 sq. foot	9.2903 sq. decimeters
1 sq. yard	0.8361 square meter
1 square rod	0.259 acre
1 acre	0.4047 hectare
1 sq. mile	2.59 sq. kilometers
144 sq. inches	1 sq. foot
9 square feet	1 square yard
30-1/4 sq. yds.	1 square rod
40 sq. rods	1 rood
4 roods	1 acre
640 acres	1 square mile

SURVEYOR'S MEASURE

7.92 inches	1 link
25 links	1 rod
4 rods	1 chain
10 sq. chains or 160 sq. rods	1 acre
640 acres	1 square mile
36 sq. mi. or 6 mi. square	1 township

CUBIC MEASURE

1.728 cubic inches	1 cubic foot
128 cubic feet	1 cord wood
27 cubic feet	1 cubic yard
40 cubic feet	1 ton shpg.
2,150.42 cu. in.	1 standard bushel
268.8 cu. in.	1 standard gallon dry
231 cu. in.	1 standard gallon liquid
1 cubic foot	about 4/5 of a bushel
1 Perch	A mass 16-1/2 ft. long, 1 ft. high and 1-1/2 ft. wide, containing 24-2/3 cu.ft.

APPROXIMATE METRIC EQUIVALENT

1 decimeter	4 inches
1 meter	1.1 yards
1 kilometer	5/8 of mile
1 hectare	2-1/2 acres
1 stere, or cu. meter	1/4 of a cord
1 liter	1.06 qt. liquid or 0.9 qt. dry
1 hektoliter	2.8 bushels
1 kilogram	2.2 pounds
1 metric ton	2,200 pounds

METRIC EQUIVALENTS — LINEAR MEASURE

1 centimeter	0.3937 in.
1 decimeter	3.937 in. or 0.328 ft.
1 meter	39.37 in. or 1.0936 yards
1 dekameter	1.9884 rods
1 kilometer	0.62137 mile
1 inch	2.54 centimeters
1 foot	3.048 decimeters
1 yard	0.9144 meter
1 rod	0.5028 dekameter
1 mile	1.6093 kilometers

DECIMAL EQUIVALENTS OF FRACTIONS

1/64	.0156	33/64	.5156
1/32	.0312	17/32	.5312
3/64	.0468	35/64	.5468
1/16	.0625	9/16	.5625
5/64	.0781	37/64	.5781
3/32	.0937	19/32	.5937
7/64	.1093	39/64	.6093
1/8	.125	5/8	.625
9/64	.1406	41/64	.6406
5/32	.1562	21/32	.6562
11/64	.1718	43/64	.6718
3/16	.1875	11/16	.6875
13/64	.2031	45/64	.7031
7/32	.2187	23/32	.7187
15/64	.2343	47/64	.7343
1/4	.250	3/4	.750
17/64	.2656	49/64	.7656
9/32	.2812	25/32	.7812
19/64	.2968	51/64	.7968
5/16	.3125	13/16	.8125
21/64	.3281	53/64	.8281
11/32	.3437	27/32	.8437
23/64	.3593	55/64	.8593
3/8	.375	7/8	.875
25/64	.3906	57/64	.8906
13/32	.4062	29/32	.9062
27/64	.4218	59/64	.9218
7/16	.4375	15/16	.9375
29/64	.4531	61/64	.9531
15/32	.4687	31/32	.9687
31/64	.4843	63/64	.9843
1/2	.500	1	1.00

CONVERSION FACTORS

(°C × 9/5) + 32 = °F.
(°F − 32) × 5/9 = °C.
Liter × 1.05671 = U.S. quarts
Quarts × .946333 = liters
Liters × 61.025 = cu. in.
Gallons × 231 = cu. in.
Kilograms × 2.2046 = lbs
Lbs × 453.59 = grams
Ounces (avdp.) × 28.35 = grams
Kilowatts × 1.341 = horsepower
Horsepower × 746 = watts
1 atmosphere = 33.899 ft. of water at 39.1°F
1 atmosphere = 760 mm of mercury
1 atmosphere = 14.7 lbs per sq. in.
1 cu. ft. water = 62.37 lbs @ 60°F
1 cu. in. water = 0.036 lbs @ 60°F
Cu. meters × 35.314 = cu. ft.
Cu. ft. × 0.02832 = cu. meters
Centistokes × density = Centipoises
Pounds/gal. at 20°C = Specific gravity at 20/20°C × 8.3216
1 cm = 0.3937 in.
1 in. = 2.540 cm

ACREAGE AND AREAS
SQUARE TRACTS OF LAND

Acres	One Side Square Tract	Area
1/10	66.0 lin. ft.	4,356 sq. ft.
1/8	73.8 lin. ft.	5,445 sq. ft.
1/6	85.2 lin. ft.	7,260 sq. ft.
1/4	104.4 lin. ft.	10,890 sq. ft.
1/3	120.5 lin. ft.	14,520 sq. ft.
1/2	147.6 lin. ft.	21,780 sq. ft.
3/4	180.8 lin. ft.	32,670 sq. ft.
1	208.7 lin. ft.	43,560 sq. ft.
1-1/2	255.6 lin. ft.	65,340 sq. ft.
2	295.2 lin. ft.	87,120 sq. ft.
2-1/2	330.0 lin. ft.	108,900 sq. ft.
3	361.5 lin. ft.	130,680 sq. ft.
5	466.7 lin. ft.	217,800 sq. ft.

BUILDING MATERIAL WEIGHTS

ASBESTOS — 110-120 lbs. per cu. ft.
BRICK (Common) — 2-1/2"x4"x8-1/4", 5.4 lbs. each; 2.7 tons — M.
BRICK (Fire) (Standard) — 9"x4-1/2"x2-1/2", 7.0 lbs. each; 3.5 tons — M.
BRICK (Hard) — 2-1/4"x4-1/4"x8-1/2", 6.48 lbs. each; 3.24 tons — M.
BRICK (Paving) — 2-1/4"x4"x8-1/2", 6.75 lbs. each; 3.37 tons — M.
BRICK (Paving Block) — 3-1/4"x4"x8-1/2", 8.75 lbs. each; 4.37 tons — M.
BRICK (Soft) — 2-1/4"x4"x8-1/4", 4.32 lbs. each; 2.6 tons — M.
CEMENT — Bag — 94 lbs. each; bbl. weighs 376 lbs.
CLAY (Dry) — 63-95 lbs. — cu. ft.; 1700-2295 lbs. — cu. yard
CLAY (Fire) — 130 lbs. — cu. ft.; 3510 lbs. — cu. yard.
CLAY (Wet) — 120-140 lbs. — cu. ft.; 2970-3200 lbs. — cu. yard.
CONCRETE — 138 lbs. — cu. ft.; 3726 lbs. cu. yard.
CONCRETE: Cinder concrete — 112 lbs. per cu. ft.
Gravel and limestone concrete — 150 lbs. per cu. ft.
Trap-rock concrete — 155 lbs. per cu. ft.
CRUSHED STONE — 100 lbs. cu. ft.; 2700 lbs. — cu. yard.
GRAVEL — 95 lbs. — cu. ft.; 2565 lbs. — cu. yard.
HYDRATED LIME — Abt. 40 lbs. per cu. ft.
LIME — 75 lbs. — bu.; 320 lbs. — bbl. large; 220 lbs. small.
MORTAR — 103 lbs. per cu. ft.
PLASTER OF PARIS — 98 lbs. per cu. ft.
REINFORCED CONCRETE — 150 lbs. per cu. ft.
SAND (Dry) — 97-117 lbs. — cu. ft.; 2619-3159 lbs. — cu. yard.
SAND (Wet) — 120-140 lbs. — cu. ft.; 3240-3780 lbs. — cu. yard.
SHINGLES — Bundles 24" long, 20" wide, 10" high — weighs 50 lbs. Approx. 250 per bl.
SLAG — 1755-1890 lbs. per cu. yd.; 65-70 lbs. per cu. ft.
SLAG CONCRETE — 135 lbs. per cu. ft.
STONE RIPRAP — 65 lbs. — cu. ft.; 1775 lbs. — cu. yard.

5.2

Planning and Design of Studio Lighting Equipment

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Visually nothing more fully characterizes a television studio than the vast array of lighting equipment suspended overhead. What may seem to be a random array of pipes, fixtures and cables is actually several carefully planned systems which allow a lighting director to accommodate virtually any lighting requirement (Fig. 1). The careful design and planning of the studio lighting systems is the key to achieving the primary function of a studio—a technically ideal environment for television production.

The modern television studio naturally evolved from the combined input of both film and theatre techniques. Like a film stage, television production required an acoustically treated, large open space, with the capability for overhead suspension of both lighting and scenic materials. The theatrical influence on television studios and their lighting systems came as result of early television broadcasting requirements. To meet the continuous production requirements of live television and the high illumination levels required by early cameras, television borrowed methods of theatrical efficiency such as C-clamps, pipe grids, and dimming, which are now standard features of a fully equipped studio.

In designing a television studio, the lighting system is a major consideration as it is so closely inter-related to the physical size and shape of the room. While the television industry has seen tremendous technological improvements throughout its evolution, the basic physics of light, and its ability to describe three dimensional form, cannot change. Consequently, many older studios have upgraded virtually every piece of broadcast equipment while the basic components of an older, well designed lighting system still function extremely well.

Various developments in broadcast equipment have progressively lowered the footcandle requirements for studio lighting. Parallel to these improvements, lighting control systems have also grown more sophisticated and have actually become less expensive. These factors combine to

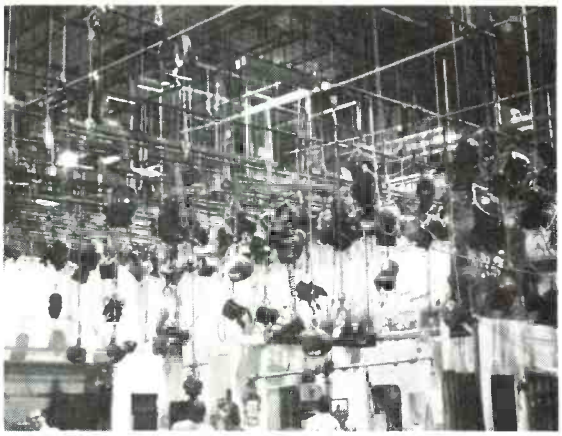

Fig. 1. This large fixed grid studio illustrates the relationship of the various aspects of the studio lighting system. (WPBT)

give the lighting director even greater control of the televised image.

STUDIO SIZE

Ideally, a studio provides the optimum environment for any type of production. In practice however, studios of various sizes tend to function best for particular types of production. The typical broadcast plant requires several sizes of studios to most effectively service its programming schedule. Typically, a small studio (1,200 sq. ft.) is dedicated to news, interviews, and public affairs, while 2,400 sq. ft. is the most common size of a small production studio. For general production, a studio of 5,000 sq. ft. or more will offer fewer limitations. The size of your studio must be carefully determined by your existing and future programming requirements, keeping in mind that the larger the studio, the fewer limitations you will build in. These are important decisions, since the lighting system must be planned relative to the size and specific requirements of the studio.

Studio Height

When determining the height of a new studio, the lighting suspension system must be given careful consideration. The grid height for small and medium size studios is a function of the current fixed TV aspect ratio 4:3 and the normal wide angle zoom lens. Most zoom lenses can cover approximately a ±45° field. By calculating the actual width of this maximum horizontal dimension, the height of the grid can be determined by applying the aspect ratio. You must also keep in mind that the studio lighting fixtures will hang approximately two feet below the suspension structure. In addition, a normal studio pedestal and cameraman will keep the lens approximately five feet in front of a wall or any other obstruction, further reducing the maximum coverage of the lens (Fig. 2). This method describes the theoretical maximum picture possible and the height the fixtures need to be mounted to stay out of the picture. However, television is primarily a close-up medium, and limitations in grid height can be overcome by various camera angles and special fixture mounting systems. Current experiments in high-resolution television may bring about a new, more rectangular aspect ratio in the near future, which will actually reduce the studio height requirement.

SUSPENSION SYSTEMS

The suspension system for the studio lighting is also a critical factor in determining studio height. All types of lighting suspension systems are commonly referred to as the "grid". In its simplest form the system consists of a series of pipes suspended below the studio ceiling, in a regular pattern (Fig. 3). A grid must allow lighting fixtures to be hung anywhere over the entire studio. Actually the grid can consist of several different types or combination suspension system.

Fixed Grid

A fixed grid is the most common and least expensive system of suspending lighting fixtures. The pipes are generally laid out on a perpendicular 4′ × 4′ spacing, which provides adequate flexibility in hanging positions (Fig. 4A). The most common fixed grid height is 14 feet. This height offers a good compromise between easy

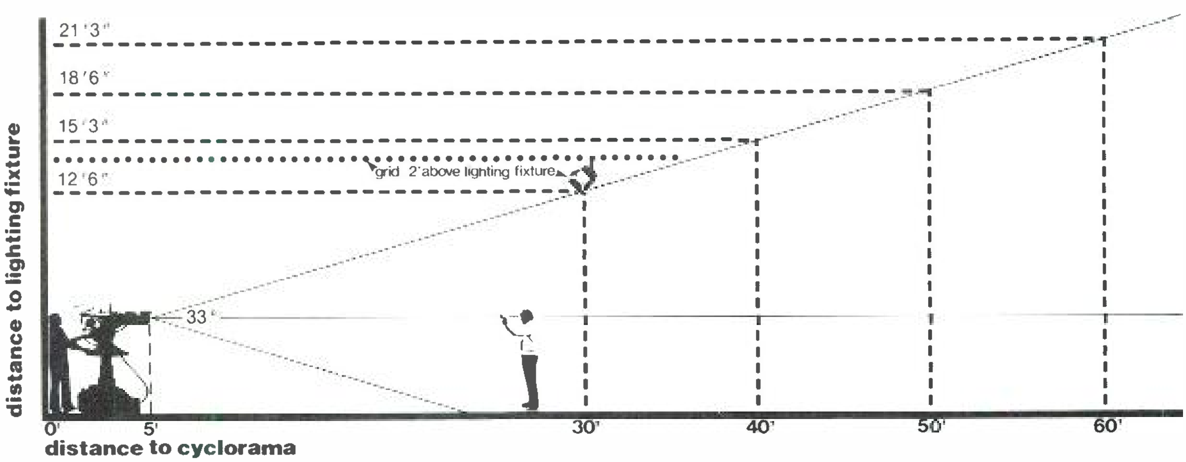

Fig. 2. This drawing illustrates how the vertical aspect of a camera lens at eye level determines the height of a studio grid.

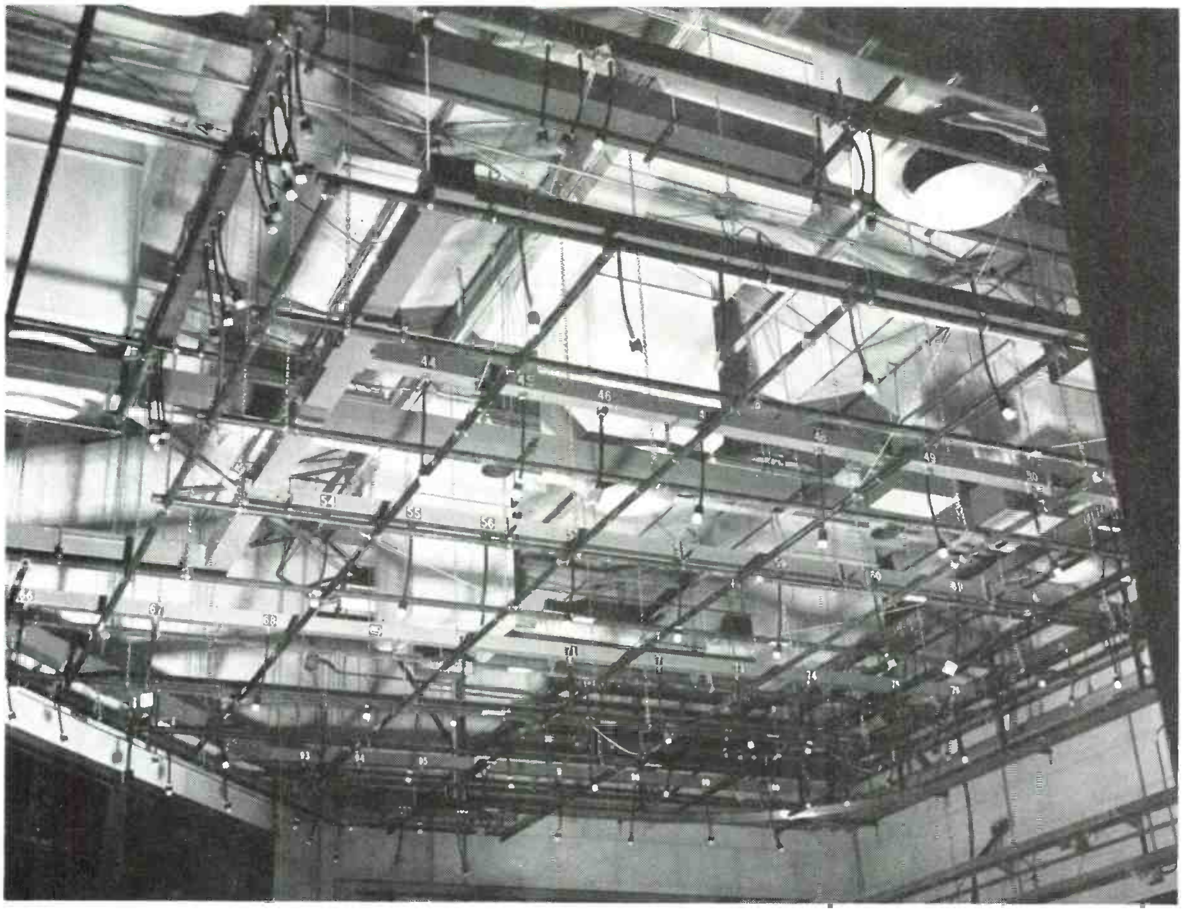

Fig. 3. This fixed grid studio employs two levels to create a higher apparent background, multiple pigtail outlets are positioned for cyc lighting units. (WFAA-Dallas)

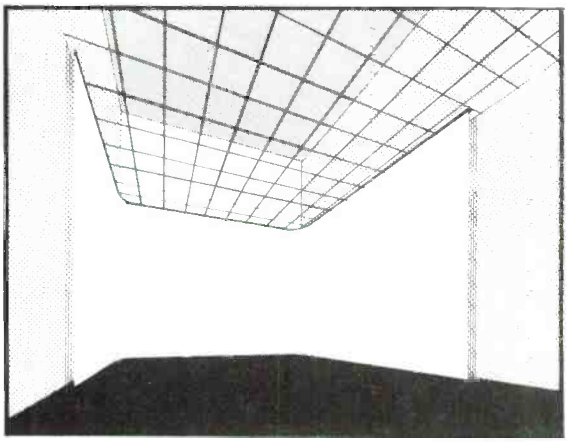

Fig. 4A. Fixed grid plan.

ladder reach and adequate clearance for wide shots. A fixed grid that is much higher than 14 feet greatly increases the amount of labor needed to install and adjust fixtures. For a large multipurpose studio, a fixed grid can be a serious limitation. Bi-level fixed grids have been utilized to create a higher apparent background without additional cost and to keep the major portion of the grid at a reasonable height. To solve this problem more efficiently, several other systems have been developed.

Catwalks

Catwalks are generally also hung at a fixed level. Catwalks provide a measure of increased efficiency over a fixed grid. Although they are somewhat more expensive, catwalks allow studio electricians to work on lighting at the same time carpenters are handling the scenery. This can be an important time savings on a tight production schedule. Catwalks are usually arranged to create a fixed grid utilizing their handrails in conjunction with an extension rod on each fixture (Fig. 4B). Catwalks when hung too low create the same height limitation as a fixed grid, but generally catwalks are higher than a fixed grid, with extension rods lowering the fixtures to a more efficient height. Unfortunately, catwalks do not eliminate actually getting on a ladder to adjust the fine focus of each fixture. Catwalks primari-

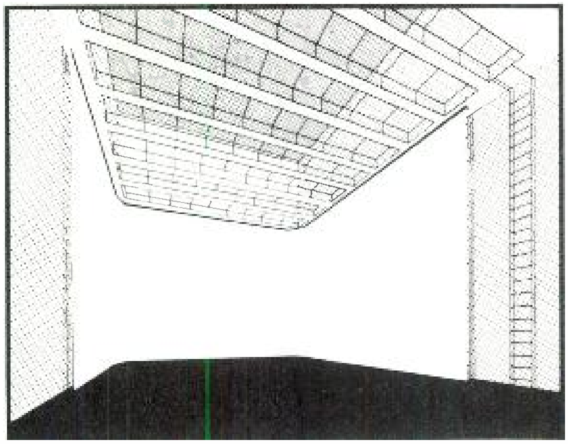

Fig. 4B. Catwalk grid plan.

ly save time in set up and strike and offer the crews a safe work platform above the studio (Fig. 5).

Recently, most fixture manufacturers have developed pole-operated yokes for their units (Fig. 6). Each fixture has a socket, which can be reached by a pole/crank to adjust each of the basic functions, pan, tilt, and spot/flood. There is a reasonable limit to how long a pole can be easily manipulated, but this feature considerably enhances fixed, high level grid systems. It is not possible to focus studio fixtures by this method as quickly or efficiently as by hand, but this system does provide a means of focusing otherwise unreachable units blocked by scenery and is seriously worth considering for any type of grid system.

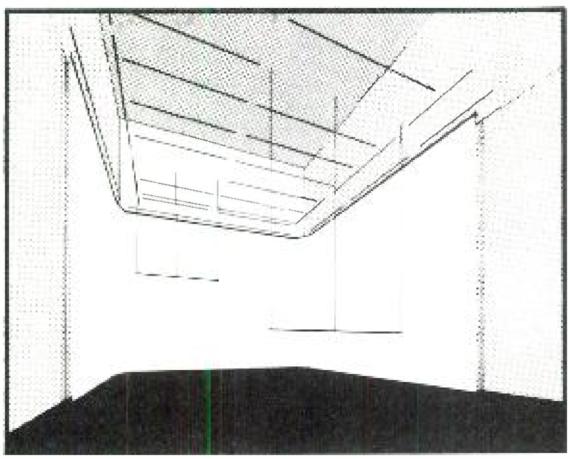

Fig. 4C. Batten plan.

Battens

The larger and therefore higher a studio that is required, the more complex the studio lighting suspension system becomes. In very large studios, a basic theatrical staging factor, in addition to the aspect ratio, determines the ultimate grid height. That is the "flyout clearance". Just as in a theatre stagehouse, flying scenery is a common requirement in larger-scale TV production. Flown scenery commonly requires a minimum grid height of 40 feet. To raise and lower the scenic units and the lighting fixtures, a regular pattern of long battens or pipes run across the width of the studio (Fig. 4C). These battens run up and down on steel cables and are manually balanced by cast iron weights. These counterweighted battens solve the height problems but add additional work in rebalancing the counterweight arbors everytime fixtures are added or removed from the batten. Counterweight rigging mounted along the studio wall encroaches on valuable floor space (Fig. 7).

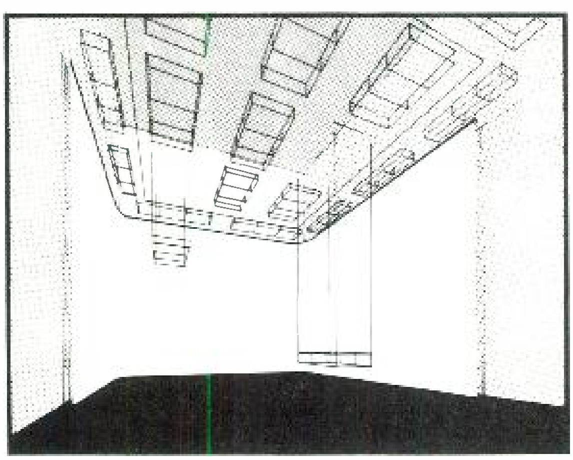

Fig. 4D. Modular grid plan.

Winches

To solve the negative aspects of counterweight systems, many installations use electric winches to raise and lower the battens. These winches can be operated by sophisticated control systems which will allow the batten to be lowered for service or adjustment and then returned to an exact preset trim or level (Fig. 8). The motors will easily lift any variable load within its designed capacity.

Many variations on the motorized batten system have been developed. In order to increase flexibility the battens have become shorter. This increases the number of motors required and the cost.

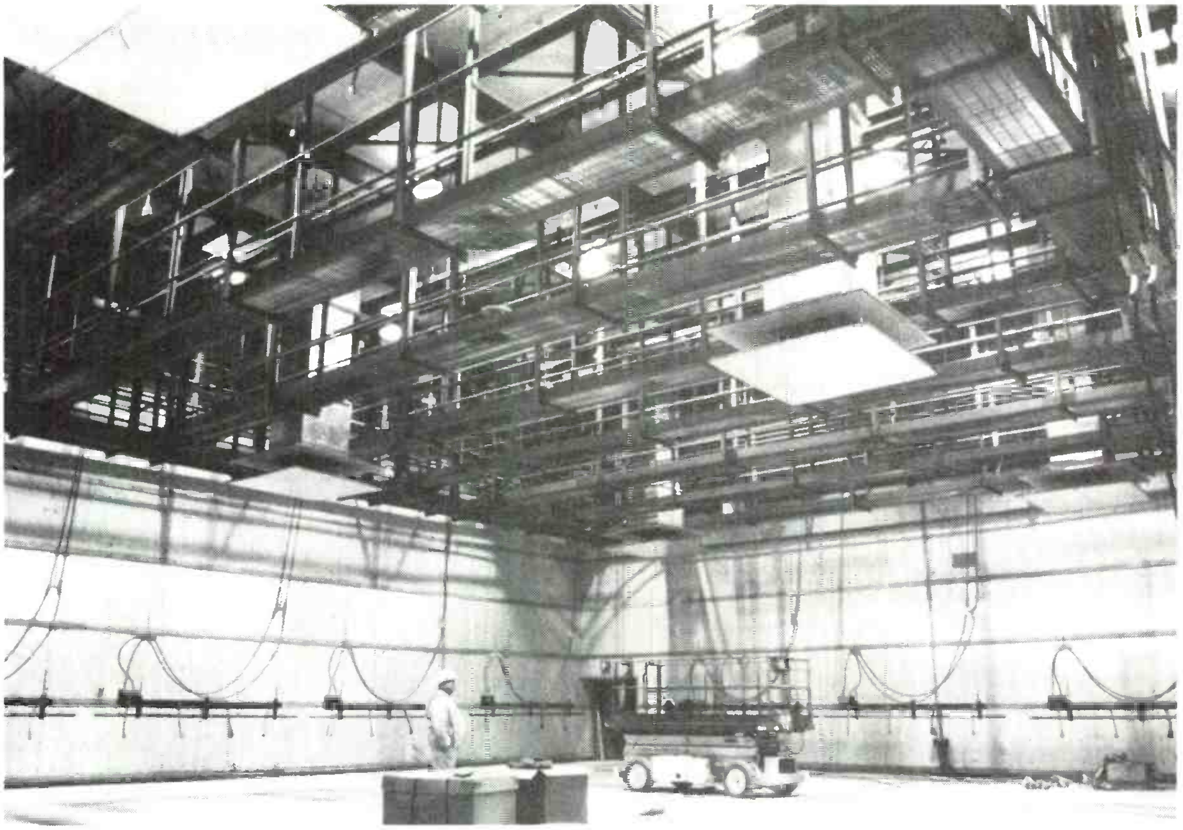

Fig. 5. In a catwalk studio lighting units are positioned in the open bays between catwalks. The motorized battens around the perimeter allow a higher backlight position where ceiling height is limited.

Fig. 6. By inserting a long crank into the adjustment cups, this pole operated fixture can be panned, tilted, or spot/flood focused from the floor. (Courtesy of Mole-Richardson)

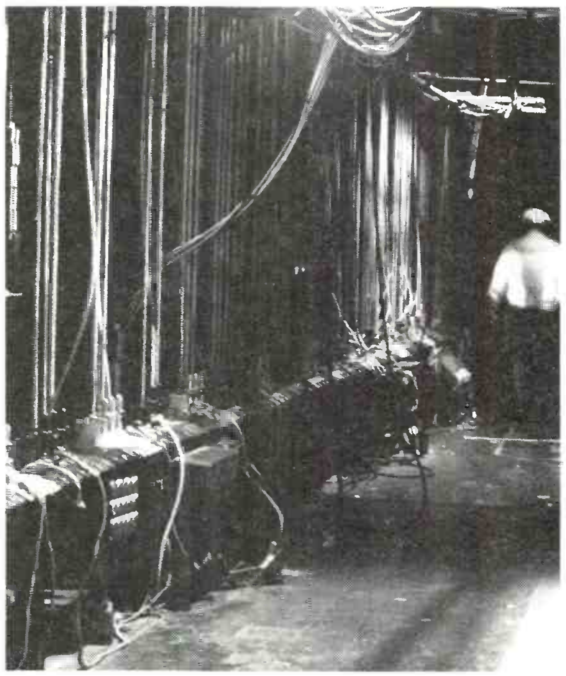

Fig. 7. A typical studio counterweight rigging system lockrail requires about 5 feet of floor space along one studio wall. (NBC-Brooklyn)

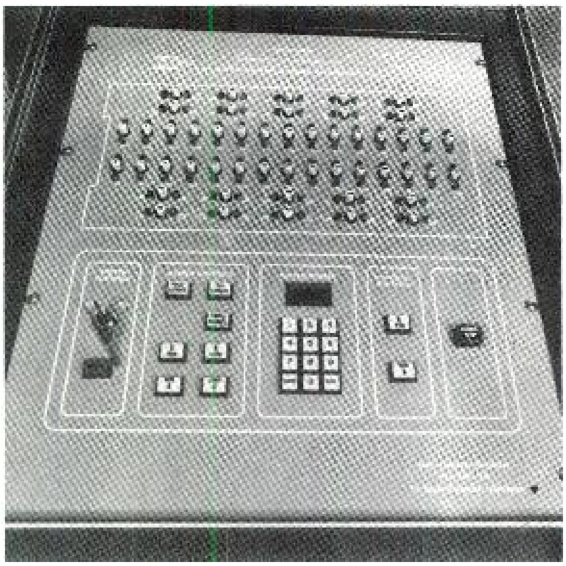

Fig. 8. Computerized control panel for the motorized batten system at WNET. The panel includes an illuminated display which mimics the actual batten layout. (Courtesy of Peter Albrecht Corporation)

Modular Grids—Self Hoisting

To take advantage of the convenience and power of electric winches without excessive cost, a number of modular grid systems have been installed. These modules are essentially small pipe grid sections, which can be raised and lowered. They offer a flexibility similar to short battens, but since they cover a larger area fewer motors are required (Fig. 4D). Because these modules are designed to be self-hoisting, the structural requirements of the studio structure are greatly simplified, and the grid remains clear for motors and steel hoisting cables (Fig. 9).

In studios which have either counterweighted or motorized battens a full walk-over grid should cover the entire studio. This grid-iron usually is made of steel grating or channels as in a theatre stagehouse and provides the support for the adjustable rigging. This walkover grid provides easy access for any overhead suspension task any production may require.

In developing a fully integrated grid system, it is essential to coordinate all the building's structural, electrical and mechanical systems, in relationship to each other and the grid. Whereas, in normal construction many of the mechanical elements in a ceiling are placed where convenient, it cannot be over-emphasized that improperly planned or installed ductwork and conduit runs can be a hindrance to production.

Cyclorama

Designing the cyclorama is integral in planning the studio lighting and grid systems. The grid or suspension system must provide a mounting position for the cyc lights located at the proper relationship to the cyc. Also, the total area of cyclorama will affect the calculation of the studio power service.

Hard Cyc

Small studios often incorporate "hard" cycloramas: smooth plastered surfaces which actually blend flush into the floor. These provide the ideal infinity effect which draperies cannot equally simulate (Fig. 10). Hard cycloramas can be easily painted any color as needed and are especially effective for certain chroma-key techniques that can be spoiled by even the most invisible seams. Hard cycloramas are generally limited to small studios since their hard surface area invariably creates acoustical problems (Fig. 10).

Cyc Pit

In very large studios a cyc pit (Fig. 10) is a compromise solution for creating this infinity effect. The pit contains and conceals the bottom cyclorama lighting, which is essential for a tall cyc and the bottom of the drapery. When it is shot from the proper angle, the pit will simulate a background without a horizon. (Fig. 11).

For most studios, a drapery cyclorama is the most convenient solution. This seamless drapery is hung on carriers which roll along a track and allow it to be positioned anywhere around the perimeter of the studio. Two parallel tracks permit another type of background to be pulled in front of the stretched cyc. Switches on the track system allow the draperies to be easily transferred to the front or rear tracks (Fig. 12). Leno or "filled scrim" is the most common drapery material, although seamless muslin is an inexpensive alternate material. Recently, a translucent plastic rear-projection screen type material has become a popular material because it can be lit from either the front or the rear.

In larger studios where experienced lighting directors are available, true white cycloramas are used to achieve greater color intensity on the cyc. However, in smaller studios, where the talent occasionally must work close to the background or where limited control equipment is available a 60% TV white cyc should be used for better control of contrast.

Drapery cycloramas are generally furnished with jack chain weights. Removable pipe weights bent to match the shape of the track should also be provided to create a wrinkle free background. One of the most common errors in cyclorama design is an insufficiently large radius at the corners of cyc. No matter what material the cyc is made of, the larger the radius the easier it is to light evenly and accomplish the desired effect.

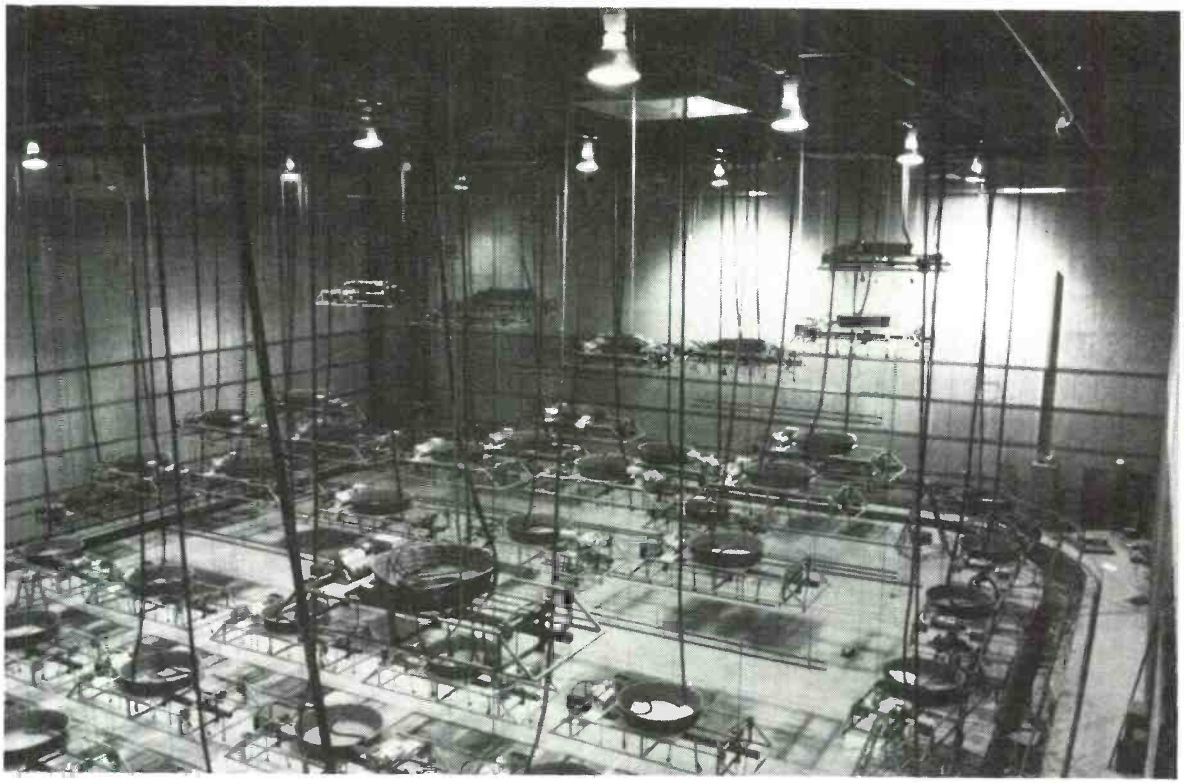

Fig. 9. Modular self-hoisting grids allow variable height suspension of fixtures in a regular pattern through the studio. The round tubs serve as cable collectors for lighting and winch power. (Dallas Communications Complex)

Fig. 10. Notice how well this hard cyc creates an infinity effect even with work lights and the floor partially finished. (American Express-NY)

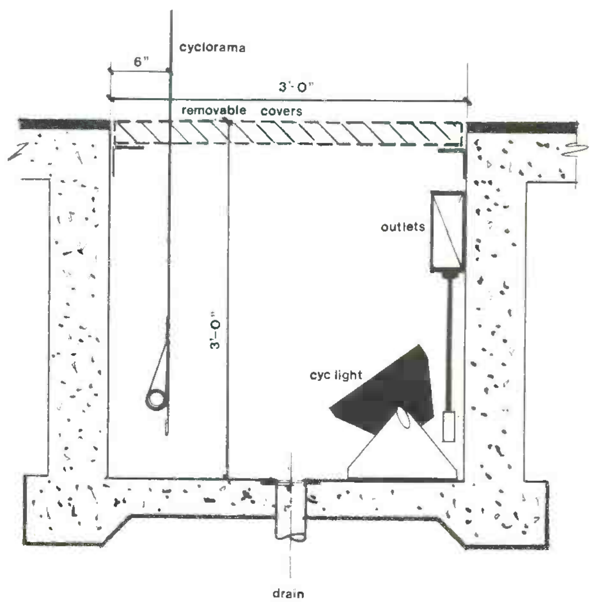

Fig. 11. Cyc Pit: Section view through a typical cyclorama pit.

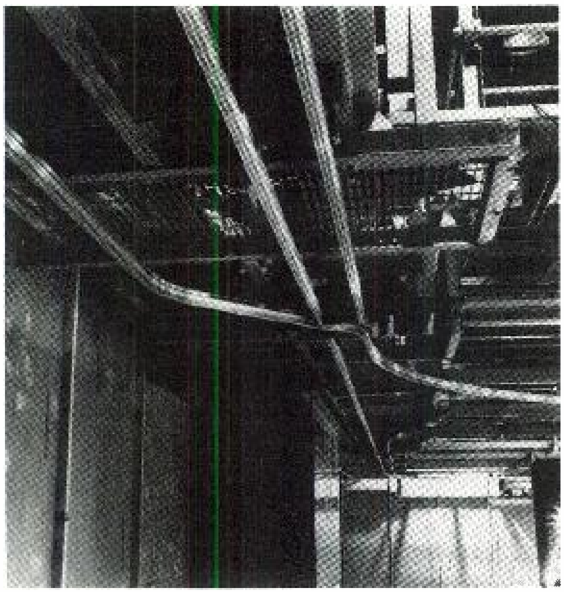

Fig. 12. Cycloramas and draperies can be shifted to various track configurations by utilizing transfer switches. (CBS-NY) (Courtesy of Peter Albrecht Corporation)

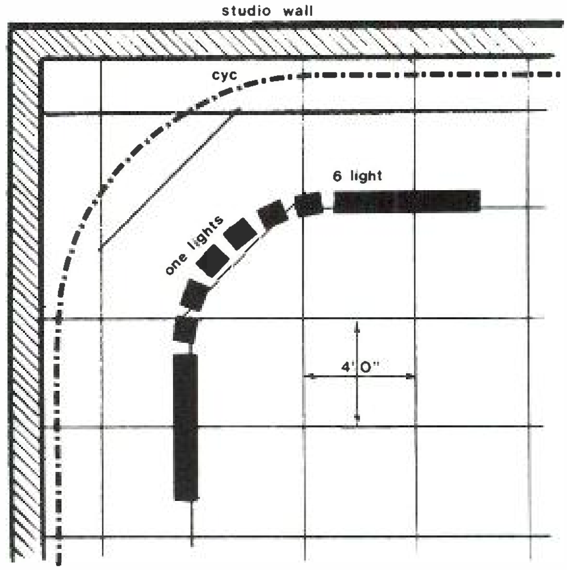

Fig. 13. Cyc Light Layout: Typical arrangement of strip cyc lights relative to the cyc curve.

Generally the arrangement of doors into the studio will define the most functional area for the cyc to be normally positioned. In the basic design of the grid or suspension system the type of cyc lighting system should be pre-determined, and the proper hanging system provided accordingly.

There are two basic types of cyclorama lighting fixtures. They are Striplights and a type of wide-spread fixture commonly known as "Far Cycs". Striplights are continuous rows of quartz halogen or Par lamps, which for an 18 foot high cyclorama should be mounted 5 to 6 feet from the cyc, while the "far cyc" units should be mounted 7 to 8 feet from the cyc (Fig. 13). The entire suspension system should be designed around these dimensions. Far cycs generally will light the cyc as evently, with less wattage than striplights. Because far cycs are mounted a greater distance from the cyc they force the talent further away from the background. Generally, striplights should be limited to three colors otherwise the separation between alternately colored lamps is too great to provide even coverage.

ARCHITECTURAL CRITERIA

Electrical

With the basic studio size determined and the "net production area" (NPA) defined by the cyclorama, it is possible to determine the power requirement for any given studio. The power requirements for studio lighting is a direct function of area and the required level of illumination. This power requirement remains consistent regardless of the grid height. For a lower grid, a greater quantity of smaller wattage fixtures are used, while a higher grid will require fewer fixtures of increased wattage. In either case, the total watts per square foot will remain roughly the same.

An average of 55 watts per square foot of NPA has been proven in production to provide sufficient power for any normal television lighting requirement. This method of calculating the studio load will provide sufficient power for virtually any situation. This is generally more power than is actually required for the average studio with today's state-of-the-art equipment, but it allows for overlighting by novices and higher levels necessary for special situations. It also provides sufficient power for an average cyclorama. Very tall cycloramas which may require double rows of fixtures at the top and/or the bottom cyc lights should be calculated as an additional power requirement based on the wattage per lineal foot of cyclorama according to the lighting system chosen.

This calculated power service describes a real maximum probable load, and the feeders must be able to supply this full amount of power (Fig. 14). Only certain limited productions will ever require this full amount. Also, note that for dimmer-per-circuit systems, the dimming capacity will be far greater than this calculated power service. The full dimming capacity need not be fully serviced, since the larger number of dimmers is a matter of convenience, and will never be fully loaded beyond the maximum probable load.

Airconditioning (HVAC)

Because the maximum lighting loads seldom occurs, it is unnecessary to use that maximum load as the basis for the airconditioning capacity. Production practice has shown that a diversity of 60% can be applied to the maximum load and still provide sufficient capacity for full period shooting. Of course any other heat generating devices and the population of the studio should be included in the airconditioning calculations. A properly designed HVAC system will require very large ducts to meet stringent acoustical requirements. These large ducts can often interfere with a grid system and should be closely coordinated.

Determining Studio Power and HVAC Requirements

1. **Net Production Area (N.P.A.) =**

   N.P.A. = Actual Studio Area in Sq. Ft. minus area behind cyc and other areas unusable for production.

2. **Approximate Number of Studio Lighting Outlets Required:**

   $\frac{\text{N.P.A.}}{\text{15 Sq. Ft.}}$ = Number of outlets required (Patch Panel)

   $\frac{\text{N.P.A.}}{\text{18 Sq. Ft.}}$ = Number of outlets required (Dimmer-per-Circuit)

   Note: *Actual number of outlets determined by specific layout.

   *Breakdown of 20A. circuits and 50A. circuits varies by grid height.

3. **Studio Lighting Load:**

   N.P.A. Sq. Ft. × 55 Watts = Total ______ Watts.

4. **Studio Lighting Power Service:**

   $\frac{\text{Total Watts}}{\text{120V.}}$ = Total ______________ Amps.

   $\frac{\text{Total Amps}}{3}$ = Power Service 3ϕ/4 Wire 120/208V.

   Note: Round off to the next larger standard panel size.

5. **Studio Lighting Heat Load for HVAC:**

   Studio lighting Load (KW) × 60% (Diversity) = Heat Load for HVAC Design (KW)

6. **Dimmer Room Heatload for HVAC:**

   HVAC Heat Load (Item 5 above) × 5% = Dimmer Room Heat Load

Fig. 14. Determining Studio Power Service—Table: the above describes the method of determining studio power.

Electrical Distribution

For a studio to be totally flexible, lighting equipment power must be distributed uniformly throughout the studio. At the grid level, power is commonly distributed through prewired plugging strips (Fig. 15). These strips are mounted directly to the grid, catwalks, or fly in and out on battens or moveable grid sections.

Each circuit terminates in the studio in a pigtail/outlet. There are two types of connectors in common use—stage pin connectors and twist lock. Stage pin connectors are less expensive and more common in rental equipment. If you envision renting additional fixtures on occasion, this may be an important consideration. In addition, the cost savings of stage pin connectors recur with fixture and cable purchase. Twist lock connectors have a positive locking feature. The final choice should be based on your specific requirements.

The number of circuits and their capacities is also related to studio size. The actual number of circuits is based initially on the square footage (approx. one outlet every 18 sq. ft. NPA) and then altered as necessary to conform to the particular grid system and the cyclorama layout. Dedicated circuits for the cyclorama are often overlooked. The cyc lights consume a large number of circuits and once they are hung in place they will seldom, if ever, be moved. Because of the rather wide spacing of the cyc units it is sometimes more efficient to feed these lights from individual grid mounted junction boxes rather

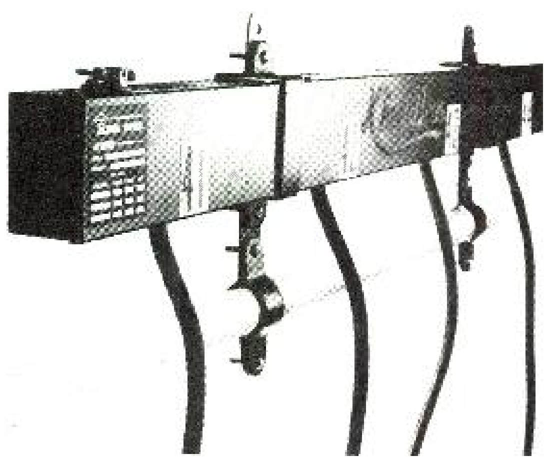

Fig. 15. Grid mounted connector strips provide a convenient means of distributing large quantities of circuits throughout the studio. (Courtesy of Kliegl Bros.)

than a plugging strip. Also, when using "far cyc" lights it is often convenient to double up the outlets to take full advantage of the 20A. capacity of the dimmers.

Additional circuits should be located around the perimeter of the studio at grid level. This is a natural backlight position for a set facing away from the wall. Properly located circuits save considerable time in running jumper cables.

At 30" above floor level, around the perimeter of the studio wall, mounted outlet boxes should be provided (Fig. 16). Generally, the governing factor for their placement is relative to a layout of floor mounted cyc strips. Otherwise these outlets are used for miscellaneous lights on floor stands and practical fixtures on the set.

For most studios the vast majority of circuits will be rated 20 amperes. The larger the studio, the greater the quantity of 50 ampere circuits. In a medium size studio (3,500-5,000 sq. ft.), 50A. circuits are generally located in a regular pattern throughout the center area of the grid and slightly more frequently around the perimeter backlight position. In larger and higher studios the density of 50A. circuits must be increased, although the overall outlet density should not be significantly decreased. On adjustable height grids, a full complement of both 20A. and 50A. outlets should be provided for using the appropriate fixtures for the varying heights. Large studios (10,000+ sq. ft.) with high grids must be furnished with 100A. circuits (Fig. 17).

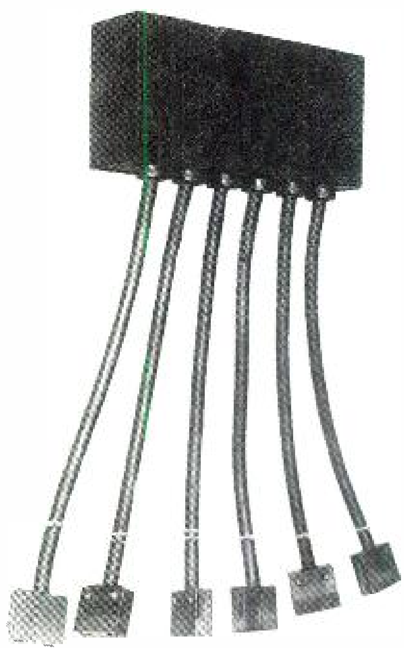

Fig. 16. Wall mounted outlet boxes are generally located around the perimeter of the studio. (Courtesy of Kliegl Bros.)

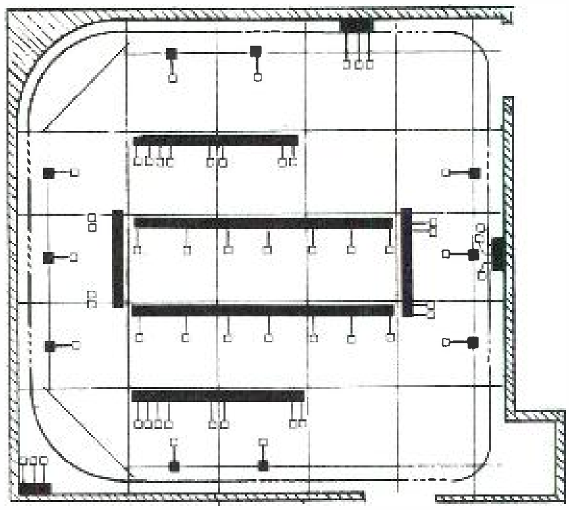

Fig. 17. Typical Studio Distribution Plan for a small industrial studio.

LIGHTING CONTROL

The importance of a dimming and control system cannot be underestimated. Dimmers allow easy control of numerous fixtures, balancing and recording of levels, and the blending of colors. A dimmer system frees the lighting director of the unnecessary burden of calculating and controlling the loads through more labor intensive mechanical methods. Electronic dimming and control allows the execution of complex lighting cues, which are a very effective production element.

Until recently, standard practice would have terminated all the circuits in a patch panel at which point they would be plugged into and powered by a limited number of large capacity dimmers. Because of the decreasing cost of mass produced electronic components, and the increased cost of hand fabricated equipment, patch panels are being phased out in favor of dimmer per circuit systems. The dimmer-per-circuit system offers greater flexibility at a comparable equipment cost and reduced installation labor. In this type of system every circuit terminates in its own dimmer, with the integral circuit breaker protecting both the dimmer and the circuit. The dimmer per circuit system gives the lighting director greater control through specific control of each individual fixture.

Dimmer Bank

The individual dimmer units generally plug into electronic equipment racks to form the dimmer bank (Fig. 18). This modular system also allows quick plug-in substitution of faulty dimmer

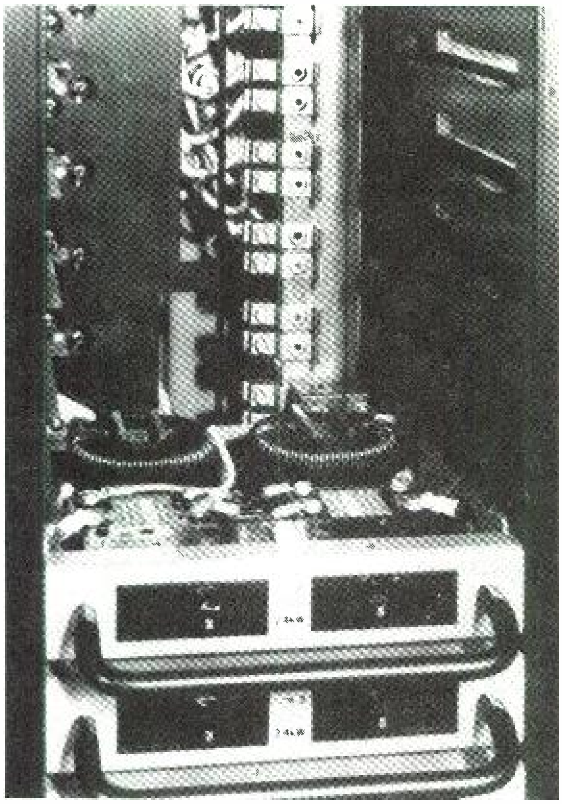

Fig. 18. Tray mounted dual dimmer modules allow high density and quick replacement. (Courtesy of Colortran)

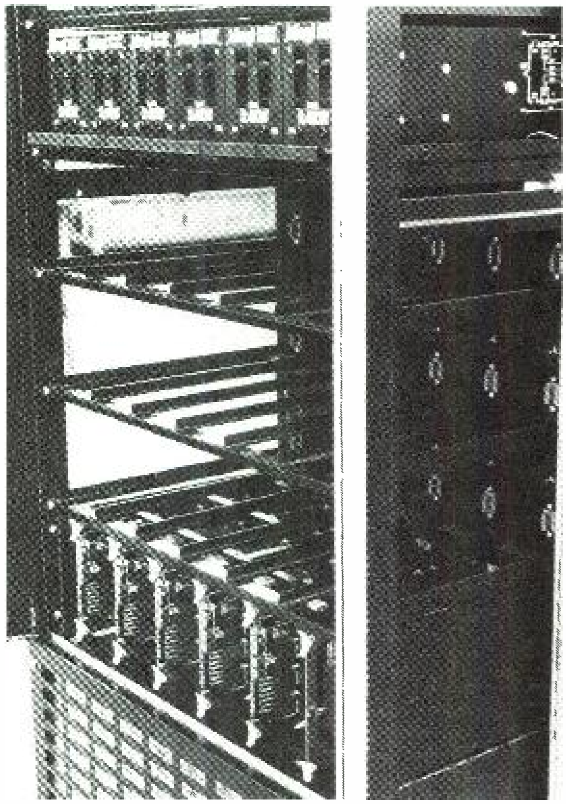

Fig. 19. Electronics common to multiple dimmer modules are located on cards in the bottom of each dimmer rack. (Courtesy of Kliegl Bros.)

modules (Fig. 16). Depending on the manufacturer, in excess of 96 individual 2.4 kW dimmer modules can fit into a single rack. All modern studio quality electronic dimmers utilize SCRs (silicon controlled rectifiers). These units are very reliable and are universally available in 2.4 kW, 6.0 kW, and 12.0 kW (20A., 50A., 100A. @120V) ratings. Only dimmers which have sufficient filtering to prevent unwanted RF interference and excessive filament vibration should be considered for use in television studios.

The location of the dimmer bank is an essential part of the initial studio space planning (Fig. 19). The dimmer room should be centrally located to minimize the length of all the circuit homeruns to avoid voltage drop and excessive installation cost. This room should be sized to allow sufficient space for required conduit radii, access to the feeder lugs, and adequate front clearance as specified by the local code.

Virtually all SCR dimmers are approximately 5% inefficient. Therefore, they could create heat equal to 5% of the energized lighting load. Since the maximum lighting load is an infrequent occurance the dimmer room cooling should more reasonably be based on 5% of the diversified load on which the studio HVAC is based.

Control Consoles

Along with the reduced cost of dimmers, the application of computer technology to lighting control systems have made today's dimmer per circuit system a financial reality.

In manual electronic dimming systems, an individual potentiometer or slider is required to operate each dimmer. To retain a particular dimmer setting, duplicate sets of sliders are necessary. Each preset group of these sliders is called a scene. While two scenes are sufficient for many productions, even a relatively simple production could require many more. The physical size of multi-scene control panels is cumbersome, and requires extensive paper work to record the setting for each preset.

Sophisticated control is possible through patching dimmers/circuits to a reasonable number of control channels. The simplest systems physically resemble the standard manual two scene preset system (Fig. 20). On these systems, each slider represents a channel rather than a hardwired individual dimmer. The specific number of control channels is a matter of convenience. Obviously, more channels afford greater control within a single scene.

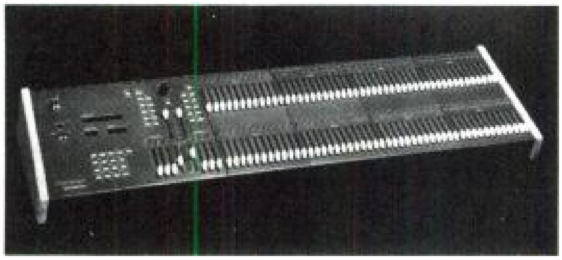

Fig. 20. Manual control panels with proportional patching can be programmed with a full scene in memory on each channel (Courtesy of Colortran)

Fig. 21. Midsize light computer with single display can program complex cues and can drive a convenient hand-held remote unit. Data cassettes provide library storage of all settings. (Courtesy of Kliegl Bros.)

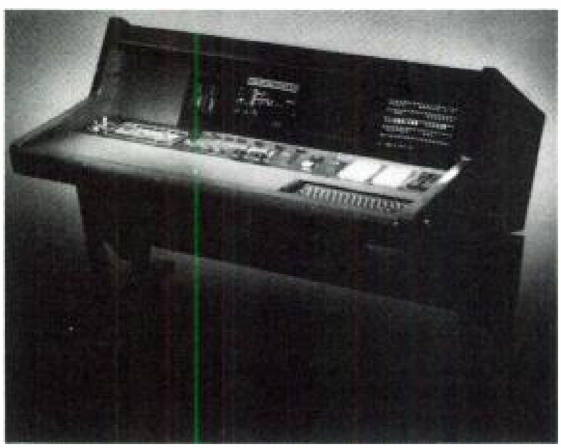

Fig. 22. The largest computer system utilizes two displays and floppy disks for storage. These systems can include complete backup systems. (Courtesy of Strand-Century)

In a dimmer per circuit system it is not unusual for a medium size studio to have in excess of 200 dimmers. To control this large quantity with only 24 channels it is necessary to patch the dimmers to channels utilizing only the dimmers required. The patching function allows any dimmer to be controlled by any channel. For example, the blue cyc lights load may require 12 separate dimmers. By patching them into the same channel they will operate together in perfect unison. All this patching occurs within the console and does not require any cord, plugs, or diode pins.

After patching the dimmers to channels and setting the desired levels, the entire preset can be stored in a memory by assigning a memory number. This preset can then be recalled by keying the appropriate number or operating the slider to which the preset has been assigned. This type of system is very economical and is suitable for most small and medium studio situations.

The greatest limitation with these simple systems is that once the levels are preset, there is no visual record of them unless you write them down or run a printout. In either case individual channels cannot be altered without reprogramming the entire scene.

This limitation is overcome in some systems with add on displays and functions available at extra cost. For studios which encounter more complex production requirements or too many dimmers for a small system, there are a number of systems with much greater capabilities. These control systems resemble a personal computer with specialized keypad and controls.

One type of system which is common to several manufacturers utilizes a single CRT to display various functions (Fig. 21). In operation, the screen displays the channel numbers and below each channel is a two digit number for the intensity. At the bottom of this field of numbers is the "cue sheet" which displays various operational functions such as cue numbers, fade times, etc.

The control panel on this type of computer control generally has a series of submaster sliders, X,Y faders, numerical keypad and a function keypad, and some tactile type of encoder for easily altering fade times and intensities.

The most complex systems employ dual CRT displays to actively display more information (Fig. 22). These systems offer so many esoteric features that most studios would never begin to tax their full potential.

Computerized lighting systems store their memory on either floppy disk or data cassettes for reuse. This permits a complete copy of all the settings in memory. In addition most larger systems are available with varying degrees of internal backup memory systems. The operational software is permanently stored in ROM within the machine.

Most computer lighting systems were originally designed for the legitimate theater where the easy daily repetition of very complex multipart cues is the primary goal. Some manufacturers have modified their programming to be more sympathetic to television's somewhat unpredictable demands. In selecting a system with this caliber of sophistication, one must carefully determine which features are really necessary.

Computerized lighting systems have greatly simplified the installation of the control wiring (Fig. 23). While manual systems require at least one wire for every dimmer, these computrized systems can be run over a single coax or a few twisted pairs for the entire system. An advantage to this simplified cabling is that the console can be easily relocated to any of the plug-in stations. In addition, when the main computer console is located outside the studio, a small focus remote the size of a hand-held calculator can be used in the studio to activate fixtures as necessary for focusing or other simple operations (Fig. 24).

This allows the console to be located in the most convenient position for a particular phase of the production. It also allows several studios to share a more sophisticated system as required, since the largest systems can plug into the same control wiring as the smallest.

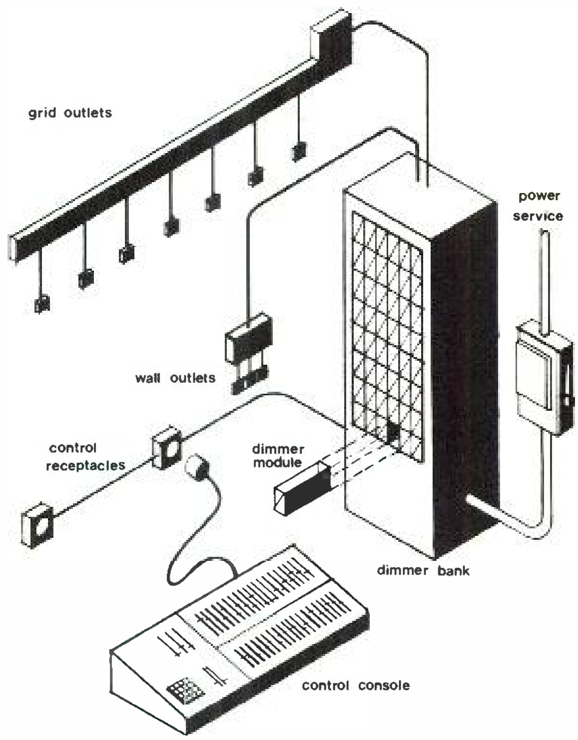

Fig. 23. Dimmer System Riser Diagram: indicating the basic components of the dimmer system.

LIGHTING FIXTURES

A television lighting director has quite a variety of lighting fixtures or luminaries available to accomplish the basic objectives of TV lighting: separation, modeling, accent, illumination, and directing viewer attention. This variety of fixtures and the various qualities of light which they produce have been developed to enhance the efficiency of studio operation. The luxury of utilizing a basic fixture and manipulating the quality of light by external diffusers, cutters, reflectors, etc. for each shot as in film is not normally possible in multi-camera television production. Therefore, a complement of fixtures with given characteristics provide a lighting director with a pallet of choices to design the lighting.

Reduced to its simplest form, television lighting design develops from a three point system including a key light, the principal source of illumination, base or fill light, a uniform diffuse illumination to establish sufficient levels for camera pickup and backlight, illumination from the rear, which highlights a form separating it from the background. There are a number of excellent texts specifically on lighting design techniques listed in the bibliography. These basic techniques are men-

Fig. 24. A complete line of "BABY" size units are available for tight locations and special applications. (Courtesy of Mole-Richardson)

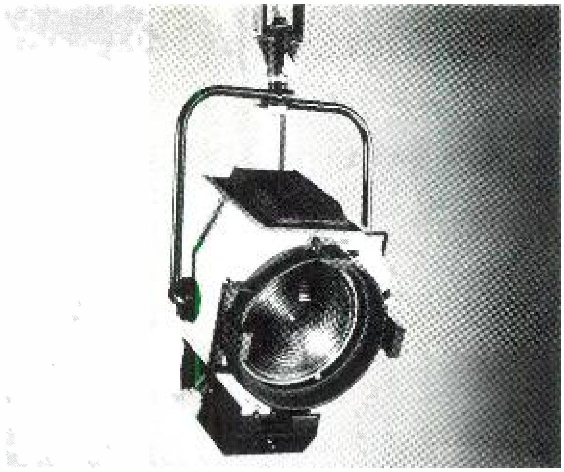

Fig. 25. Studio quality fresnel spotlights are much more durable and have superior optical systems compared to more inexpensive theatrical units. (Courtesy of Colortran)

Fig. 26. Studio fresnel spotlights must offer good quality long leaf barndoors as an essential accessory. (Courtesy of Strand-Century)

Fig. 27. A typical scoop unit. (Courtesy of Mole-Richardson)

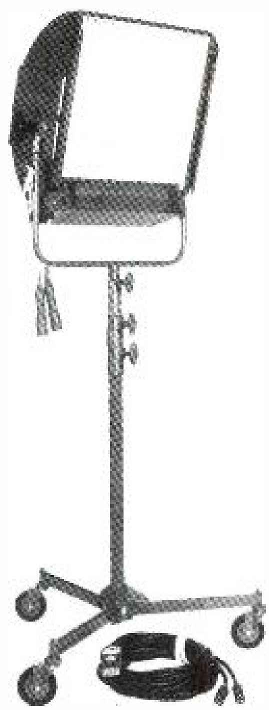

Fig. 28. Softlights produce a very diffuse source and have switches to select as many lamps as necessary. (Courtesy of Mole-Richardson)

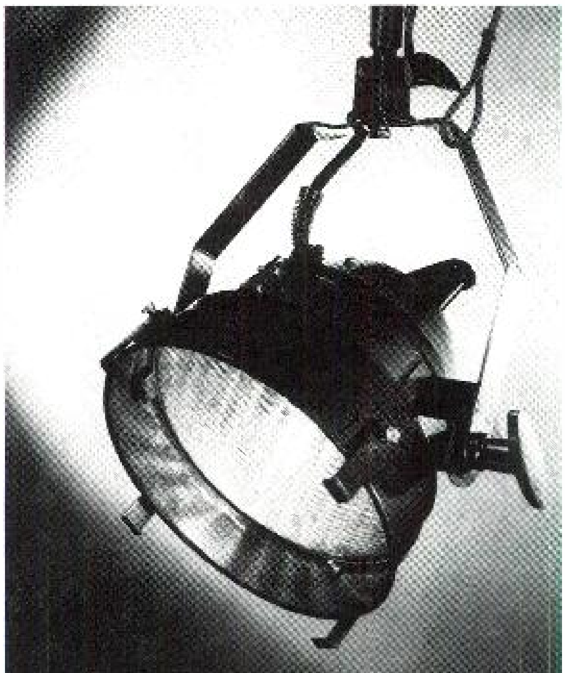

Fig. 29. Par head fixtures produce a tremendous amount of light in a very small package. (Courtesy of Colortran)

tioned only in reference to the following discussion of studio fixtures.

Fresnels

In a studio, the most common and useful fixture is the fresnel lens spotlight. The light from a fresnel is very controllable and has a smooth even field. By a simple adjustment of the position of the lamp, the fixture will produce either a narrow spot or wide flood beam of light. Equipped with barndoors this beam can be further shaped to virtually any pattern. Slots in front of the lens allow color, diffusion media or screens for intensity control to be slipped in place. A complement of single and double thickness, and half and full frame screens should be provided for all fresnels even when dimmers are available.

For most 14′ high grid studios the 6″, 8″ and 10″ in wattages from 500 to 2,000 are the most common units (Fig. 24, 25, 26). In smaller studios or tight applications "baby size" fixtures are often employed. These units enclose the same wattage lamps in small housings. A number of people prefer these smaller size units for their easier handling and somewhat different optical characteristics. As noted previously, the higher the grid, the higher the wattage fixtures are required. Therefore, in higher studios, the standard complement of fixtures might be 5,000 and 2,000 watt units or even up to 10,000 watts. Conversely, fresnel lens units are also available in sizes down to a 3″ lens with a 150 watt lamp.

In selecting a line of fresnels for your studio you must carefully weigh both the optical, and mechanical features. Poorly made fixtures will not

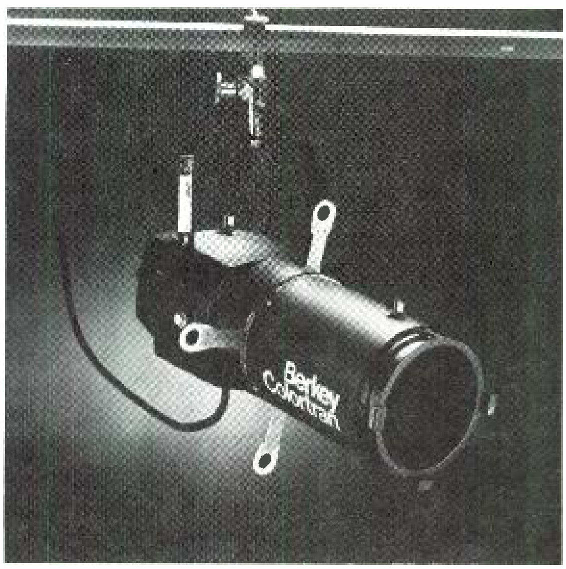

Fig. 30. Shutters on ellipsoidal spotlights allow the beam of light to be shaped into hard edge shapes or can project patterns. (Courtesy of Colortran)

focus properly once they are hot. Fixtures should also be well balanced, even with barndoors to remain in focus. The manufacturer should also provide a complete line of suitable accessories including barndoors, which are long enough, screen sets, diffusion frames and stand mounting hardware.

Scoops

Scoop floodlights are used for a variety of soft diffuse requirements most commonly in groups to provide base light to a setting. These are very simple fixtures available in several sizes, where bigger is generally better (Fig. 27). Scoops are used invariably with diffusion media, so it is essential that a diffusion frame is ordered. This is the only fixture which should not use a Tungsten Halogen lamp. The large old fashioned PS shape incandescent lamp produces a much softer quality of light, because of its larger filament area. Some manufacturers make focusing scoops, which is generally a pointless additional cost, when used with diffusion material. While the majority of lighting can be done with fresnels and scoops, there are a vast variety of fixtures which each produce a different quality of light. For a harder rectangular field of fill light a broad can be used. It is very similar to a scoop in concept, except its housing/reflector creates a somewhat less diffuse light and also is always used with diffusion media.

The softlights are another type of diffuse base light fixture, in which the light sources are totally concealed and the light is reflected in an indirect manner. These units are not particularly efficient, but are unequalled in providing a shadow free light. They are available in a variety of sizes (Fig. 28). Large 4,000-8,000 watt units are pretty unwieldly for many applications, but the smaller units in the 1,000 to 2,000 watt range are becoming very popular as fill lights, because of the reduced levels required.

Probably, the most efficient incandescent fixture used in television lighting is the Par unit (Fig. 29). The lamp itself contains both the reflector and the lens, which is available in 5 different beam spreads. While limited to very large studios and remotes, the Par is a very inexpensive and powerful tool for the lighting director.

The ellipsoidal reflector spotlight is a unique effect light in a studio (Fig. 30). The optical system of this unit allows the beam of light to be hard or soft edged, or it can be cut-off by internal shutters. It is the most commonly used to project patterns on the cyclorama. Numerous photo-etched metal patterns are available, which drop into the pattern slot. Ellipsoidal reflector spotlights are available in a variety of optical systems, but the wide angle/short throw sizes are most useful in small and medium studios.

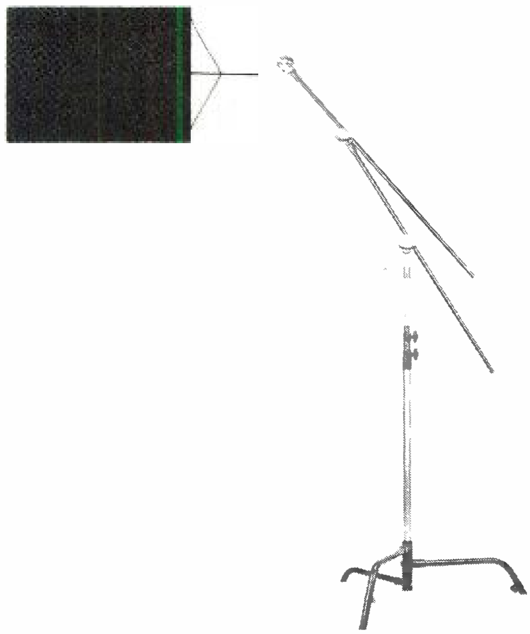

Fig. 31. The three level legs allow the stand to fold flat for storage. The head can grasp numerous flags, cutter, cookies, etc.

There are a number of accessories which make the lighting system work. Despite the expense and care expended on planning the grid system, it cannot satisfy every fixture mounting requirement; every studio must have some fixtures on rolling floor stands. In addition, many types of clamps and general grip equipment are available for special mounting situations.

One of the key accessories in the lighting system are extension rods. This device allow fixtures to be hung at a lower level than on the fixed grid. For example, a key light at a 30° angle to the talent could be much closer to the talent, and therefore brighter, than if it were hung lower. Counter-balanced devices such as pantographs and spring load telescoping hangers cannot be locked in place. Another drawback of these devices is that they must be precisely adjusted to the weight of a particular fixture and cannot be relied upon to stay in that position. Extension rods, which are nothing more than a pipe, which slides up and down are the cheapest and best choice.

The century stand (Fig. 31) has endless application around the studio. It can grasp all types of material such as flags, cutters, cookies, and support reflectors. Century stand use is limited only by your imagination. No matter how many are available, they will all be put to good use.

Ultimately, the proper quantity of fixtures, accessories, and other components of a lighting system for a studio of a given size can vary widely according to the requirements of a specific situation. For example, a studio with a fixed grid and a tight production schedule will function more efficiently with a heavy saturation of fixtures. With a large quantity of fixtures, major relocation of all the units is somewhat minimized. When sophisticated motorized grid systems are available, the ease of relocating fixtures will reduce the total quantity of fixtures for the same size studio. The choice of either of these two approaches is also dramatically affected by labor costs within a particular facility. Too often the expense of a few extra fixtures is not compared to the additional time and labor required to fully utilize a minimal complement of equipment or the cost of elaborate rigging. Final discussions on equipment purchase require a careful analysis of your specific production requirements in coordination with all the financial ramifications of various approaches.

The numerous technological improvements in reducing required lighting levels and improved control have allowed the television lighting director to become more of an artist and less of an illumination engineer. In planning a new studio, it is essential that the systems provided do not hinder this creative process.

All systems must be logical, and easily understood by the production personnel. The incorporation of specialized features tends to limit the flexibility of a studio. It should be possible to light a set equally well regardless of where it is placed within a studio.

Most television production people have the opportunity to be involved in planning a new studio maybe once or twice in their entire careers. Even years of experience in studio production are not necessarily the best preparation for coordinating studio requirements into a construction project. Often a new studio provides an opportunity to acquire a complement of equipment, which is more sophisticated than the existing staff's level of experience. In this situation, an experienced lighting director should be consulted in preparation of the equipment requirements and evaluating the quantities required. While manufacturers are sometimes helpful in this area, they are still primarily interested in selling you their product, and no single manufacturer offers a full line of suitable equipment in every area. By working with a consultant, you can share their experience in the planning and design of the lighting system for your studio.

5.3

The Broadcast Television Camera System

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OVERVIEW

The purpose of this chapter is to provide basic, practical information on broadcast camera systems, circuits and components. Major topics covered here are as follows:

- The Basic Camera System
- Camera Types
- The Optical System
- The Camera Tube
- Solid State Imagers
- Measurement and Performance of Imaging Devices
- Camera Performance
- Choosing a Camera

INTRODUCTION

The broadcast television camera system is a combination of many technologies. The zoom lens and beamsplitter are products of advanced optical design; the camera tube combines solid state and vacuum physics; solid state imagers reflect the state of the art in high density integrated circuit design; and the electronic control and processing circuitry is the result of sophisticated analog, digital and computer engineering.

It is not the purpose of this chapter to give a detailed insight into these technologies; such specialized information is better found elsewhere, but rather to provide the broadcast engineer with a practical understanding as to how these technologies work together and the various factors, which are important in the design and operation of a television camera system.

In the past all broadcast cameras were "tube-type", which means that they utilized camera tubes as imaging devices. However, recent advances in solid state imagers have resulted in improved resolution from these devices and they are being effectively used in portable cameras. Undoubtedly, as the imager technology further advances, these devices will begin to appear in the higher performance cameras.

THE BASIC CAMERA SYSTEM

All live television cameras are similar in principle, but differ in size and configuration according to their application. The basic elements of a live camera system are shown in Fig. 1, and described as follows:

The Optical System

Consists of a lens, filters, beamsplitter and, in the case of automatic cameras, some kind of pattern projector.

The Lens

Usually the lens is a "zoom" type, which means that the focal length can be varied and,

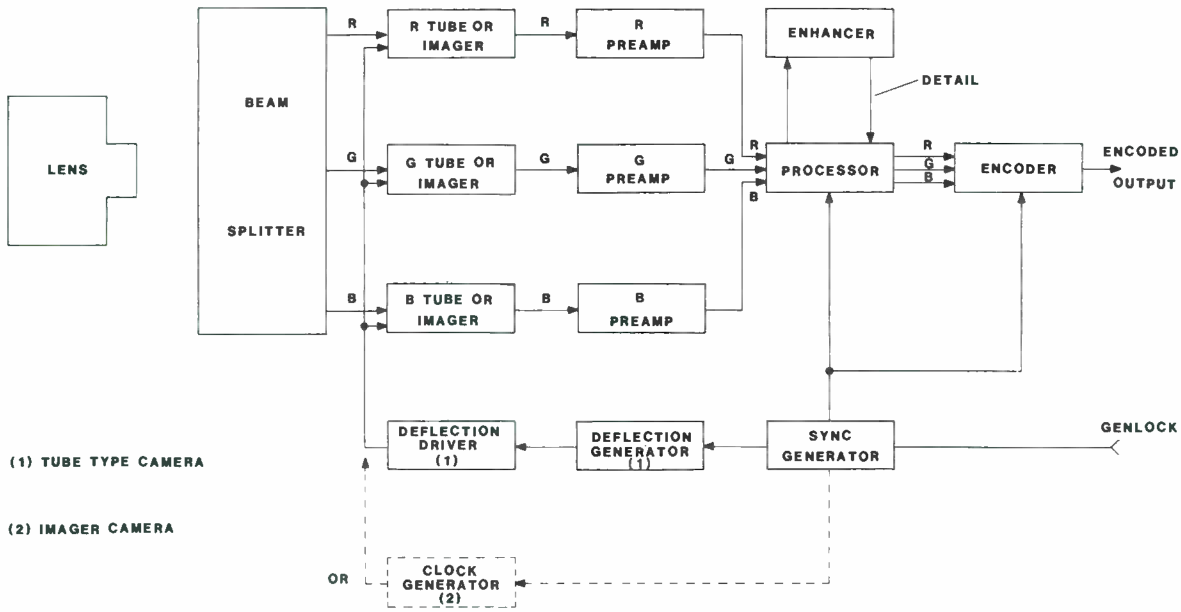


Fig. 1. Basic camera system.

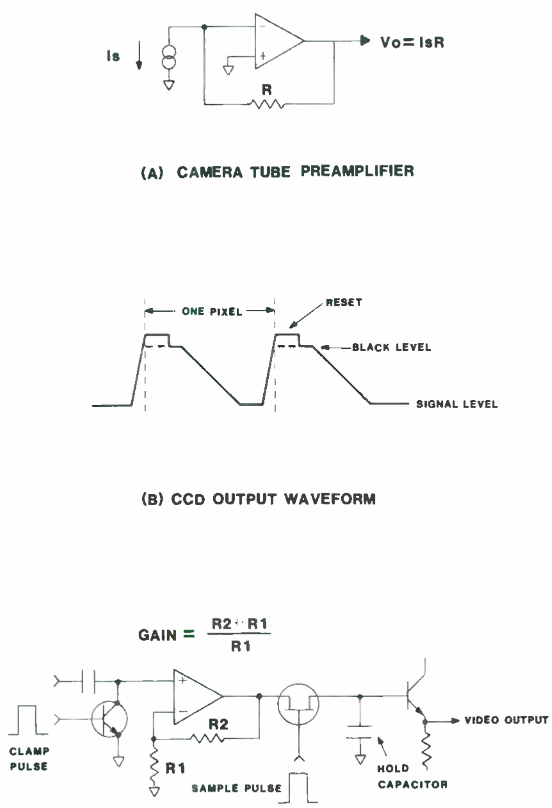


Fig. 2.

therefore, is the magnification. The purpose of the lens is to focus the object of view into the camera head, through the beamsplitter and onto the light-sensitive surface of the imaging devices, which may be solid state imagers, or camera tubes.

The Beamsplitter

Beamsplitters are either of the "relay", or "prismatic" type, however broadcast cameras invariably use the latter. The purpose of the beamsplitter is to split the incoming light into its various color components, usually red, green and blue, and to provide the spectral response, or "taking characteristic" necessary for good colorimetry.

The Imaging Device

Usually a broadcast camera has three color channels; red, green and blue; and, therefore, utilizes three imaging devices, which may be camera tubes of the lead oxide, or Saticon (trademark Hitachi Ltd) type, or solid state imagers. At the present time the camera tube offers higher resolution and is used more often in broadcast cameras, particularly in the more critical applications, such as studio, or field production. However, the resolution of the solid state imager is continually increasing and we are certain to see more of these devices as technology progresses.

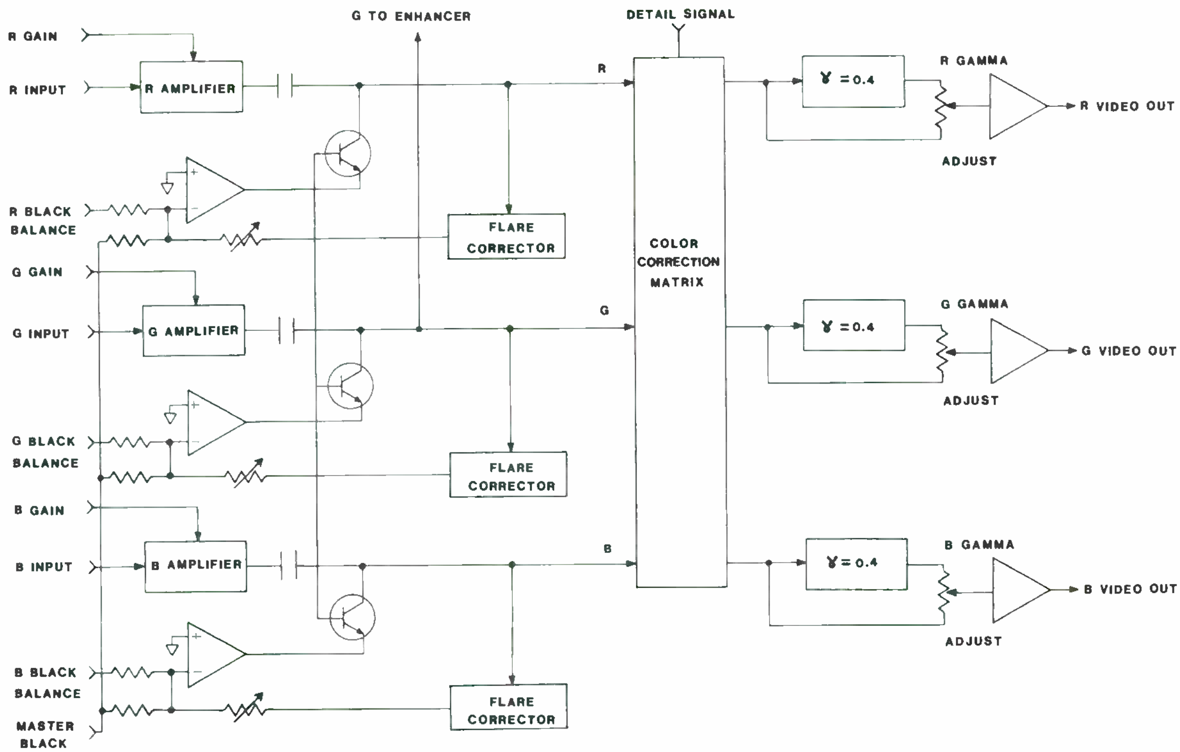


Fig. 3. Video procesor.

The Camera Tube Preamplifier

The purpose of the preamplifier is to amplify the relatively small signal current from the camera tube, which is usually in the range of 20 to 400 nanoamperes. The most common amplifier is the "constant transconductance" type, shown in Fig. 2A. Such an amplifier has a large loop gain, with current feedback through resistor, *R,* so that the output signal is given by:

$$V_{out} = I_{signal} \times R$$

The signal current is thereby predictably and accurately converted to a signal voltage.

The Image Preamplifier

The solid state imager usually includes circuits to integrate and amplify the transferred charge, so that the resulting output signal is in the form of a small modulation voltage riding on the tips of a rather large clock pulse, as shown in Fig. 2B. Therefore, the imager preamplifier must perform the dual function of further amplifying the imager output voltage and separating the signal from the clock pulse.

The block diagram of a typical imager preamplifier is shown in Fg. 2C, in which case the signal from the imager is first clamped and amplified. The signal is then removed from the clock pulses by means of a synchronous sample and hold technique. A low pass filter, designed to pass the reclaimed video signal, but to reject unwanted the sample pulse, should follow.

The Video Processor

A typical video processing system is shown in Fig. 3, with corresponding waveforms in Fig. 4. The function of the video processing system is described as follows:

Black Level Clamp This first clamp may be one of many, but is the most important, because it must establish the black level reference and identify the "capped" black level, which is the signal

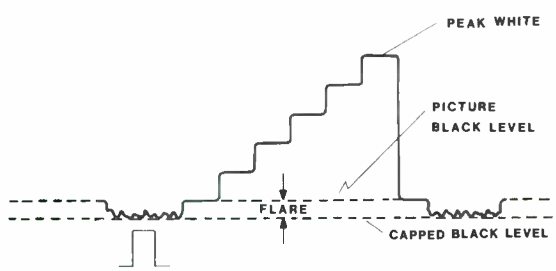


Fig. 4. Video waveform.

level present when there is no illumination entering the camera. It is indicated by the dotted line in Fig. 4. In the example shown, a switching type clamp is used, with the clamp pulse timed to coincide with either the horizontal blanking interval, in the case of a camera tube, or the overclocked horizontal period for an imager.

Black Level Control is necessary to provide master black level adjustment and to match the red and blue black levels to that of the green channel, by using the "black balance" controls.

Gain Equalisation is required to compensate for the varying spectral sensitivity of the imaging devices, and the gain is adjusted to obtain equal signal levels from the three color channels. Usually, the green imaging device is operated at a predetermined signal level and the video processor is designed to provide 100% output under this condition.

Adjustable gain is then provided in the red and blue channels so that these signals may be matched to that of the green channel, to achieve "white balance".

Flare Correction This phenomenon is caused by scattered light in the optical system, which gives rise to an elevated "picture" black level, as shown in Fig. 4. If flare was non-existent, a perfect black in the scene would produce a corresponding signal level equal to the "capped" black level. Because some degree of flare is always present, this scattered light will distribute over the entire picture area and raise the general level of the video signal, producing a black level error.

The spectral distribution of the scattered light depends on the reflectances in the optical system and, in particular, the reflectance characteristic of the imaging device photo-sensitive layer. For example, the brownish color of the lead oxide photosurface indicates that much of the red light is being reflected, a fair amount of green, but little blue light. Therefore, the amount of flare varies in each of the color channel and, in the case of the lead oxide camera tube, will be least in the blue channel.

To compensate for flare, the camera designer assumes that the total amount of scattered light is proportional to the total amount of light entering the optical system. For an imaging device, which has a linear transfer characteristic, the total illumination is proportional to the Average Picture Level (APL). The camera designer can determine the APL by integrating the video signal and producing a dc voltage which is proportional to the total incoming light. An adjustable percentage of this dc voltage is then applied as a flare correction to the black level control circuits in Fig. 3.

Gamma Correction is required to compensate for the non-linear characteristic of the monitor CRT. The typical CRT has the characteristic shown in Fig. 5C, which shows that the control grid voltage has progressively more effect on output brightness for high signals (white) than for low signals (black).

If the camera system, which includes the monitor, is to function as a linear transducer, the camera itself must have the complimentary, non-linear characteristic of Fig. 5B. These non-linear characteristics are known as the "gamma" characteristics of the device, which in the case of the CRT, is approximately 2.7. The general form of the CRT characteristic equation is therefore:

$$B = A(V_c - V_o)^{2.7}$$

Where:

B = the screen brightness.
V_c = the control grid voltage.
V_o = the control grid cut-off voltage.
A = a constant.

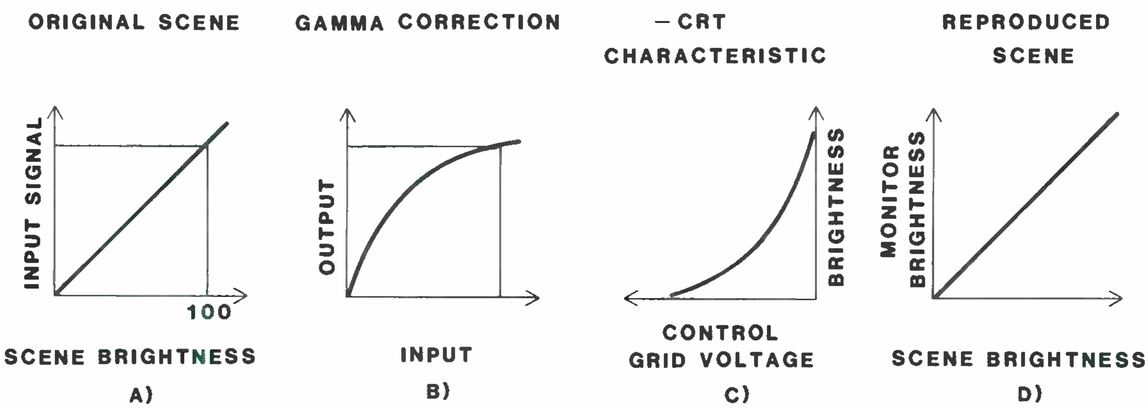


Fig. 5. Gamma correction.

The camera transfer characteristic must then be of the form:

$$V_c = KE^{\gamma}$$

Where:
K = a constant.
E = the scene illumination.
γ = the gamma of the camera.

For B to be proportional to E,

$$2.7 \times \gamma = 1$$

therefore, $\gamma = 0.37$

In practice, this degree of gamma correction would require a greatly amplified black signal, which would emphasize black noise. Also, it turns out that a system gamma slightly greater than unity gives a more dramatic quality to the picture, and is preferred by many broadcasters. Therefore, typical camera gamma correction usually falls in the range of 0.4 to 0.45, resulting in an overall system gamma between 1.08 and 1.22.

Aperture Correction The modulation depth of the camera tube as shown in Fig. 6, is limited by the diameter of the scanning beam spot size in conjunction with the lateral leakage of the photo conductor. In early types of camera tubes the spot size was the most significant factor and, because this dimension was controlled by the diameter of the electron gun limiting aperture, the resulting resolution characteristic became known as the "aperture response".

As explained in the section on solid state imagers, the modulation depth of the imager is limited by the size of its smallest picture element, the "pixel". For an imager with 403 horizontal pixels, the theoretical response is very similar to that of the camera tube, as shown in Fig. 6.

Good camera resolution requires that the MTF be corrected. The correction circuit must be of linear phase, such as the one shown in Fig. 7a, consisting of a pair of delay lines, with characteristic frequency, $f = 1/t$, Where t is the delay. The resulting correction characteristic is shown in Fig. 7b. "out of band" correction refers a characteristic frequency above subcarrier, and "in band" when the frequency is below subcarrier. The corrected response is shown in Fig. 7c.

Color Correction the color television camera and its monitor comprises a "tri-stimulus" system, which means that it employs an additive mixture of three primary colors to produce any other color within its range. The spectrum of colors that can be displayed on a television screen are limited by the coordinates of its phosphors. In 1953 the US National Television Standards Committee (NTSC) recommended the coordinates shown in the CIE diagram of Fig. 8, which were adopted by the Federal Communications Commission (FCC). The NTSC coordinates defines the range of colors which can be displayed by the television system.

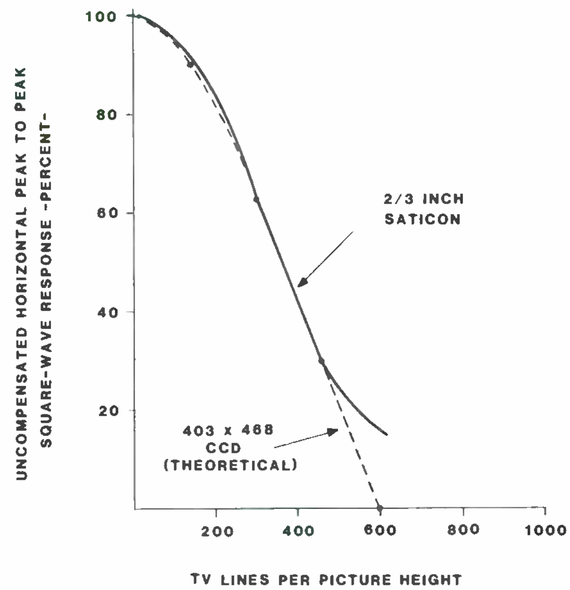


Fig. 6. Modulation depth.

In 1931, in an effort to standardize the interpretation of color, a set of "standard observer" tristimulus values were recommended by the Internation Commission on Illumination (CIE) as shown in Fig. 9. If we analyze the color camera channel responses required to obtain perfect colorimetry, in conjunction with an NTSC coordinate system, we obtain the characteristics shown in Fig. 10. Note the negative responses, or negative sensitivities required to obtain perfect colorimetry.

Fortunately in television, such negative sensitivities can be achieved to a close approximation by matrixing the three color signals, as shown in Fig. 11.

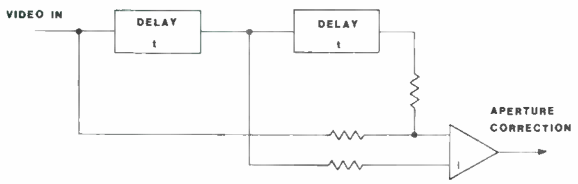
After matrixing, the corrected color signals have the form:

$$R' = a_{11}R + a_{12}G + a_{13}B$$
$$G' = a_{21}R + a_{22}G + a_{23}B$$
$$B' = a_{31}R + a_{32}G + a_{33}B$$

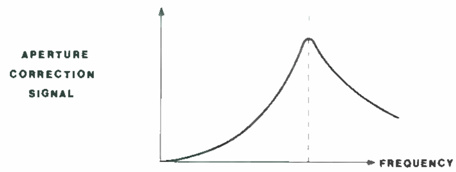
The sum of the coefficients:

$$a_{11} + a_{12} + a_{13} = a_{21} + a_{22} + a_{23} = a_{31} + a_{32} + a_{33} = 1$$

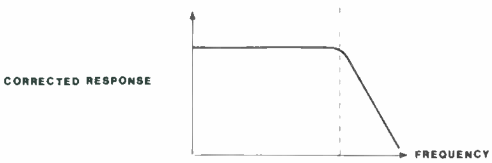
The values of the coefficients must be optimized for the corresponding taking characteristics of the beamsplitter.



(a) PHASE-LINEAR APERTURE CORRECTOR



(b) CORRECTION SIGNAL



(c) CORRECTED RESPONSE

Fig. 7.

However, certain camera manufacturers elect to utilize narrowed R,G,B taking characteristics to obtain increased color saturation, which achieves a result similar to that of the combination of broader responses and a subtractive matrix. While the latter method provides the most accurate color reproduction, narrowed response curves can provide good colorimetry and avoid the slight increase in noise arising from a subtractive matrix.

The Image Enhancer

Is also known as "contour correction", which, as the name implies, is a technique for emphasizing the horizontal and vertical edges, and greatly improves the subjective picture sharpness.

One form of image enhancer is shown in the simplified block diagram of Fig. 12. Usually the input signal is first passed through a "threshold" circuit, which clips out the black level signal and prevents it from entering the enhancer. There is no point in enhancing black level noise, which will later be further emphasized by gamma correction.

The principle of contour correction is illustrated in Fig. 13, which shows the process of enhancing the signal from a vertical white bar, Fig. 13a, which is passed through the two horizontal delay circuits of Fig. 12, resulting in undelayed (L), once delayed (M) and twice delayed (R) signals. The horizontal contour, or "detail" signal, D_h, is given by:

$$D_h = 2M - L - R$$

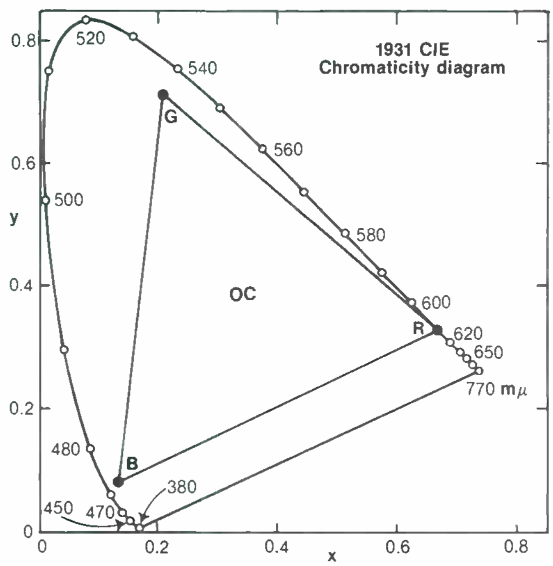


Fig. 8. CIE chromaticity diagram showing the NTSC color primaries.
(Courtesy of Wiley Publishing).

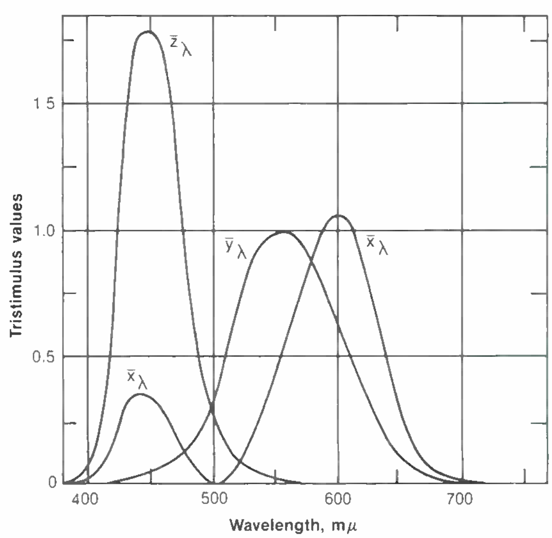


Fig. 9. The 1931 CIE standard observer curves.
(Courtesy of Wiley Publishing).

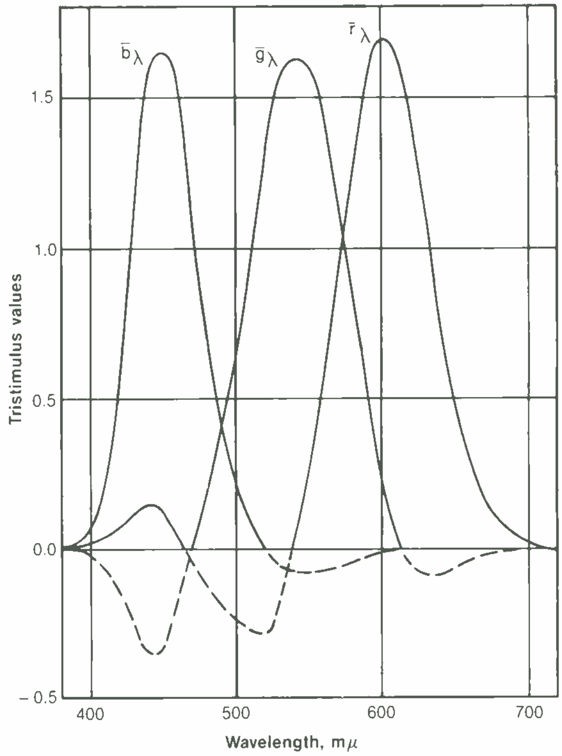


Fig. 10. Ideal taking characteristics.
(Courtesy of Wiley Publishing).

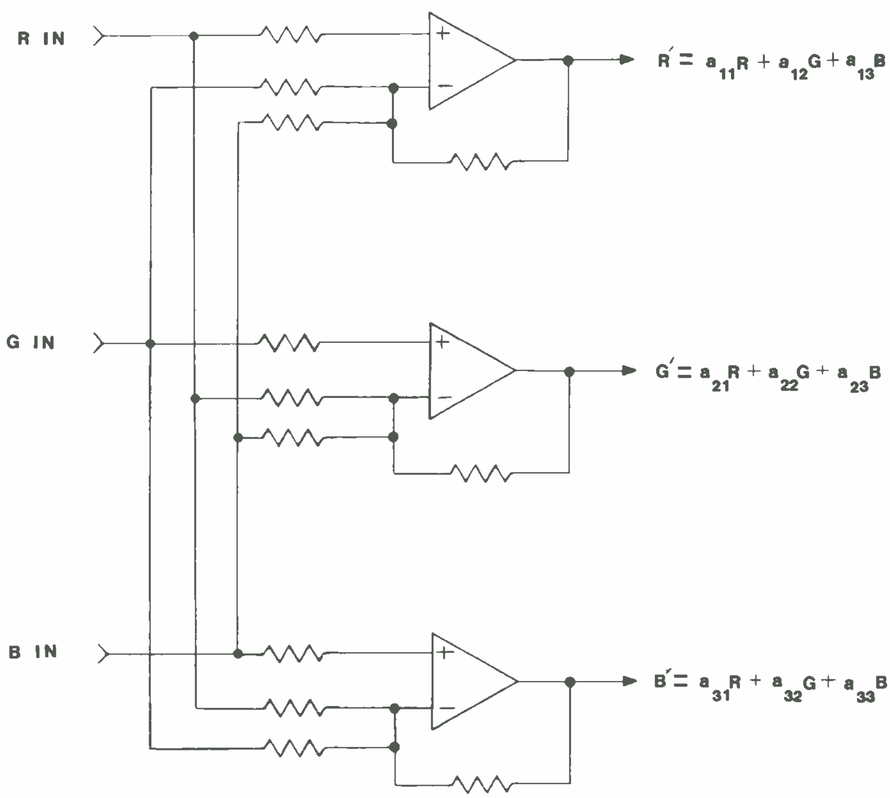


Fig. 11. Color correction.

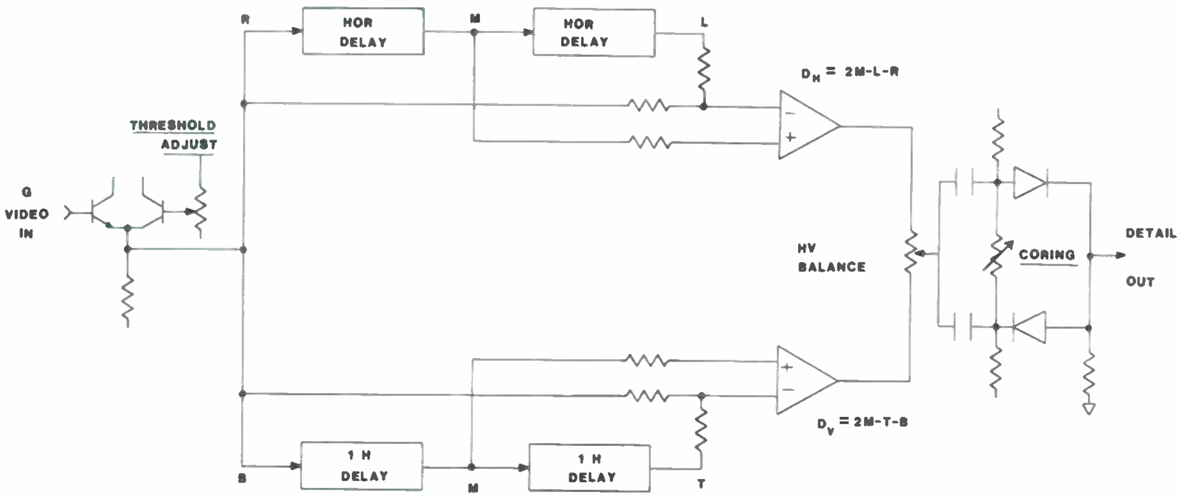


Fig. 12. Image enhancer.

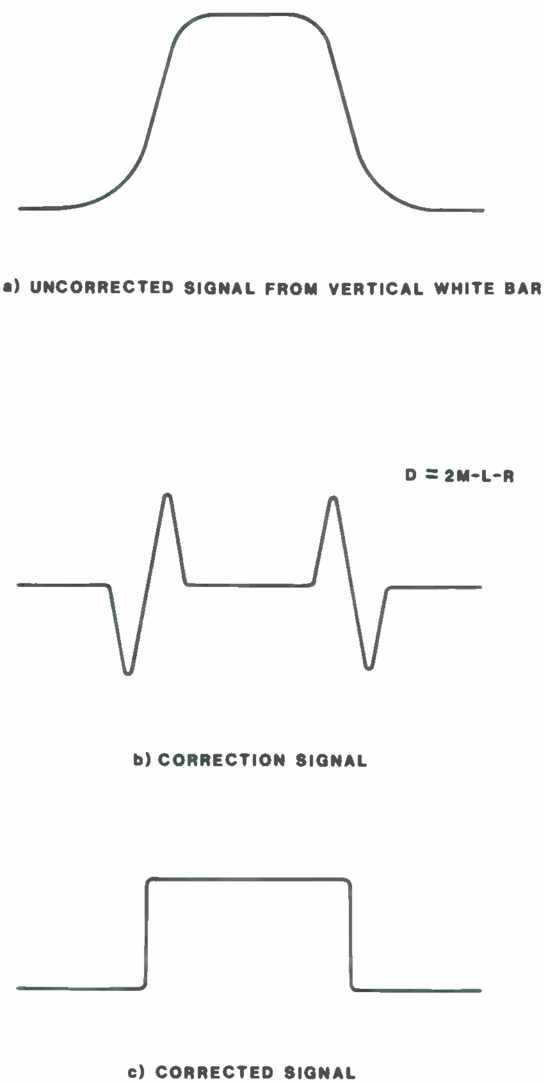


Fig. 13. Aperture correction.

and is shown in Fig. 13b. Adding this correction signal in the correct proportion to the uncorrected signal in Fig. 13a produces the corrected waveform of Fig. 13c. Increasing the amplitude of the detail signal causes the contour to become more exaggerated. Too much correction can over-exaggerate the edges with a rather unpleasant result.

The vertical enhancement signal is generated in the same way, using delay lines of one line duration, either of quartz, or CCD (Charge Coupled Device) type. In this case the undelayed (*T*), once line delayed (*M*) and twice delayed (*B*) signals produce the enhancement signal:

$$D_v = 2M - T - B$$

The Encoder

The purpose of the encoder is to combine the red, green and blue signals from the video processing system into a composite color signal, according to the color standard employed. In the USA the signal is encoded according to the NTSC standard, with the frequency characteristic shown in Fig. 14.

A simplified encoder block diagram is shown in Fig. 15. First, the red, green and blue input signals are combined in a matrix circuit to obtain the luminance signal (*Y*), and the in-phase (*I*) and quadrature (*Q*) components of the color signal. The matrix equations for these signals are:

$$Y = 0.30R + 0.59G + 0.11B$$
$$I = 0.60R - 0.28G - 0.32B$$
$$Q = 0.21R - 0.52G + 0.31B$$

The low pass filters limit the bandwidths of the *I* and *Q* signals to 1.3 MHz and 0.5 MHz respectively. Such bandpass circuits cause an inherent

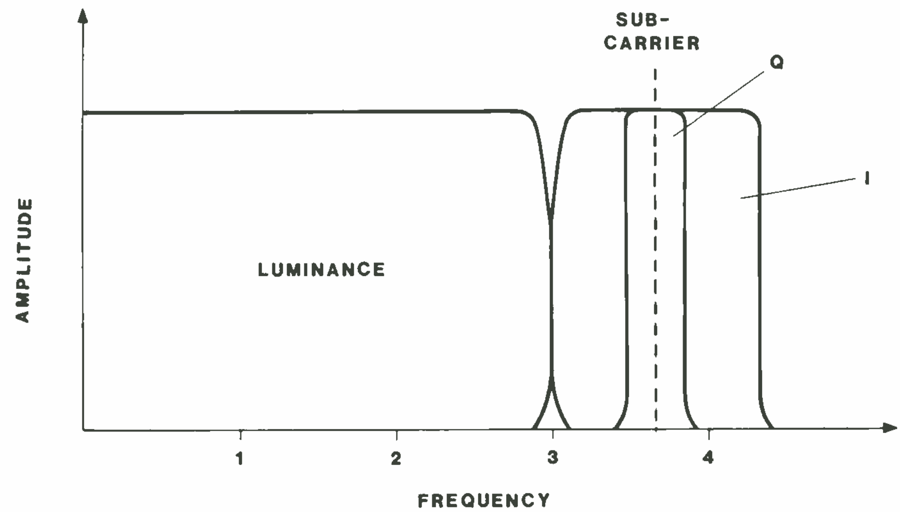


Fig. 14. NTSC frequency spectrum.

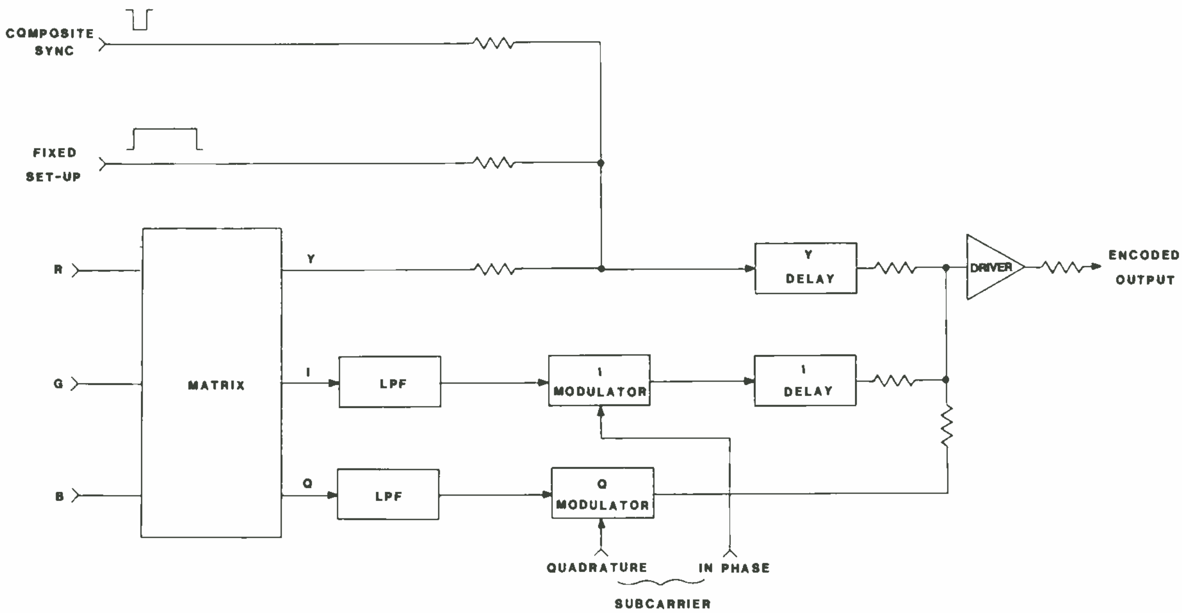


Fig. 15. Simplified encoder block diagram.

delay, the longest being the Q signal, with a delay of approximately 750 nanoseconds, depending on the actual network used. In order that the three signals remain in phase, it is necessary to provide a 750 nanosecond delay in the Y channel and approximately 400 nanoseconds delay in the I channel.

The two modulators receive in-phase and quadrature (90 degrees phase shifted) subcarrier components, as shown in Fig. 16. The in-phase component is modulated by the I signal, and the quadrature component by the Q signal. During horizontal blanking, correctly proportioned burst pulses are applied to both modulators, resulting in the subcarrier burst vector oriented at 180 degrees.

A fix set-up of 7.5 IRE units is added to the delayed luminance signal, together with the output of the subcarrier modulators, and a correctly shaped synchronizing pulse. The result is the composite waveform of Fig. 17, shown, in this case, with a color bar test signal.

Usually, the encoder includes a color bar generator, which provides a standard RGB signal to test the encoder set-up, and to provide an important aid for setting up the color monitor.

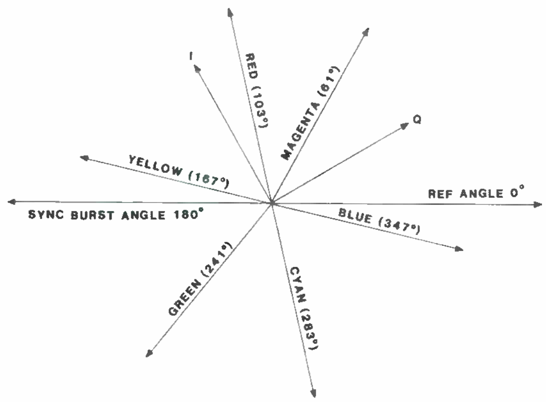


Fig. 16. Subcarrier phase relationship in the NTSC encoder.

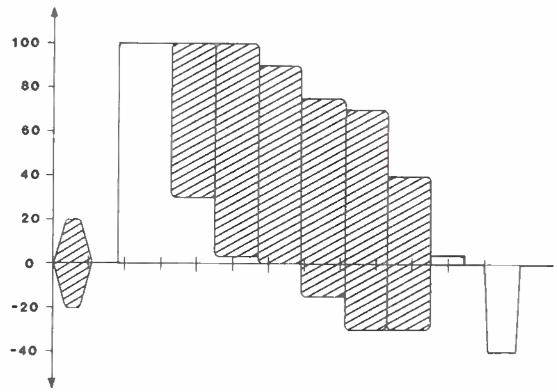


Fig. 17. Encoder output waveform.

The Synchronizing Generator

Is the master timing system, which provides all the pulses required by the camera, in accordance with the appropriate color standard? In the USA the output signal waveform must conform to EIA standard RS-170A (see Appendix A).

In event that the camera must be synchronized with another camera, or system, the synchronizing generator is provided with a "genlock" ability, whereby it can be phase-locked to an external composite color signal.

The Deflection System

The deflection system is employed in a tube type camera to provide the necessary scanning waveforms to the yoke scanning coils. Simplified deflection system is shown in Fig. 18.

In this case the vertical and horizontal master sawtooth generators produce the main scanning waveforms, dc voltage-controlled sawtooth generators provide the registration functions of size, skew and rotation and parabolic generators provide linearity adjustment.

Usually, the drivers function as current sources with sufficient feedback to ensure the scanning current through the deflection coil is a linear function of the input voltage waveform. In Fig. 18, the deflection current is given by:

$$I_{hor} = \frac{V_{master} + V_{size} + V_{linearity} + V_{skew}}{R}$$

$$I_{vert} = \frac{V_{master} + V_{size} + V_{linearity} + V_{rotation}}{R}$$

The Clock Driver

Usually, the imager contains three registers, as shown in Fig. 19, which represents a frame store CCD, the image, or "A" register, in which the image charge pattern is formed; the storage, or "B" register, into which the image is transferred from the "A" Register; and the horizontal shift register, which provides a line by line readout of the "B" register.

As illustrated in Fig. 19, the clock driver must generate the various pulses necessary to address the imager. The driver must produce a set of $V_{(A)}$ pulses during the vertical blanking interval to transfer the charge pattern from the "A" register to the "B" register. The $V_{(B)}$ pulses are required to subsequently move the data from the "B" register, line by line, to the horizontal shift register.

Once each line of charge has been transferred to the horizontal shift register, it is clocked out by the "H" pulses, pixel by pixel. Usually, the charge is shifted into a floating diffusion zone, which functions as a resettable capacitance, converting the charge pattern to a signal voltage. An additional horizontal reset pulse (*HR*) is required to reset the diffusion zone after each pixel is clocked out.

Power Supply

The power supply must produce the various dc voltages required by the entire camera system, including the relatively high voltages required by the camera tubes.

In the case of a portable camera, the power supply will be a dc to dc converter, which generates all the voltages required by the camera system from a portable rechargeable battery pack. Usually, the studio camera power supply operates from an ac input.

Automatic Set-up

The automatic set-up facility is only advantageous in tube type cameras, which require

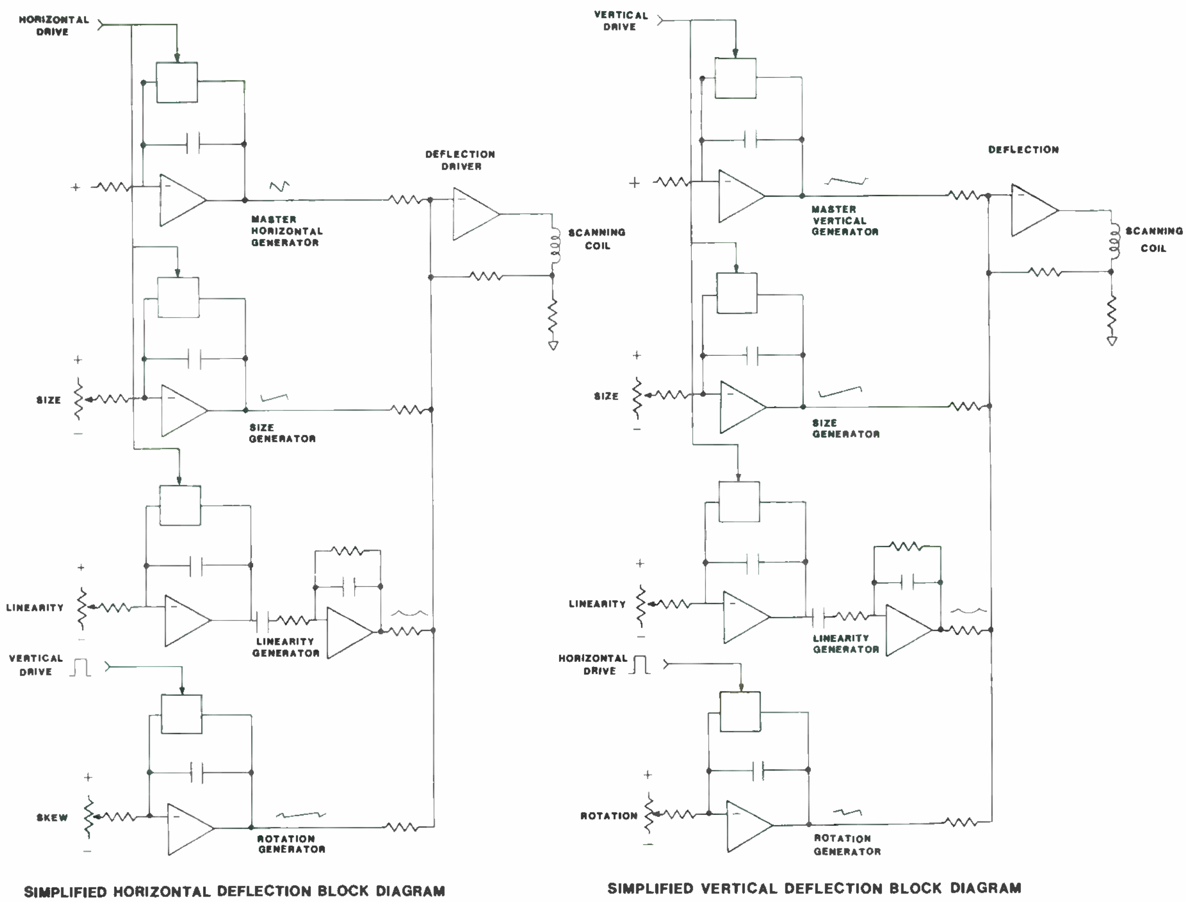


Fig. 18.

periodic adjustment, due to the inherent instability of the camera tube target and its thermionic electron gun.

The function of the automatic set-up system is to analyze the various video signals and registration errors and to generate either control voltages, or correction signals to accurately adjust the camera. Cameras with this feature require a pattern projector in the optical system (usually in the lens), which provides a test pattern compatible with type of automatic system employed.

CAMERA TYPES

The Studio Camera

This is a misnomer that previously referred to the relatively large cameras, which are most often used in the studio. Such cameras are neither restricted to studio use, nor the only cameras used in the studio. For example, many producers have discovered the advantage of having at least one portable camera in the studio; and many studios have found the smaller ⅔-inch (18mm) cameras to be a cost effective option.

The studio camera system usually consists of a camera head and CCU (Camera Control Unit). The camera head can be separated from the CCU by up to 1,500 feet of multiconductor cable, or as much as 6,000 of triaxial cable.

Studio cameras generally employ a 1¼-inch (30mm), 1-inch (25mm), or ⅔-inch (18mm) camera tubes. Due to its weight, the studio camera must be supported on a sturdy tripod, or pedestal, which also enables the use of a larger, higher performance lens. For this latter reason, primarily, the studio camera performance tends to be better than its "portable" counterpart.

Many cameramen favor the larger studio camera, even in OB (Outside Broadcast) applications, because they believe that a larger mass is necessary for a smooth, controlled pan. To a degree this is so, but a tripod with a fluid head can provide much of the same "feel," in fact, the movie industry has depended for many years on the smooth movement provided by the fluid head.

The Portable Camera

This camera must be small and light weight, so that it may be comfortably supported on the cameraman's shoulder for long periods of time. The portable camera can operate as a self-contained unit, or may be capable of modular adaptation to work with an RCU (Remote Control Unit). The RCU may be separated from the camera by up to 1,000 feet of multiconductor cable, or up to 6,000 of triaxial cable.

Due to the necessity for minimum size and weight, portable cameras utilize ⅔-inch (18mm), or ½-inch (12.5mm) camera tubes, or solid state imagers. Also, in the interest of size, the lens must be small and, therefore, its performance range is limited, compared to the studio camera lenses.

If the portable camera is used in an application where it can be supported by a tripod, a larger, so called, EFP (electronic field production) lens can be used. Such a lens greatly improves the portable camera's performance, which consequently can approach that of a studio camera.

The ENG Camera

The ENG (Electronic News Gathering) camera is a particular version of the portable camera, which design is optimized for televising the news.

The requirements of ENG puts severe constraints on the camera design, beyond the need to be small and light weight. The ENG camera must be extremely rugged, reliable and stable; the very nature of its use means some degree of physical abuse is unavoidable and there is no time to go through a lengthy camera set-up just as a news story is breaking. Because the ENG camera is used under a wide range of uncontrolled lighting conditions, it must also be very sensitive and able to handle spurious highlights without severe picture degradation from veiling glare, or other problems.

The EFP Camera

This is a special type of portable camera, which, as the name implies, is used for portable production. Fundamentally, the needs of portable

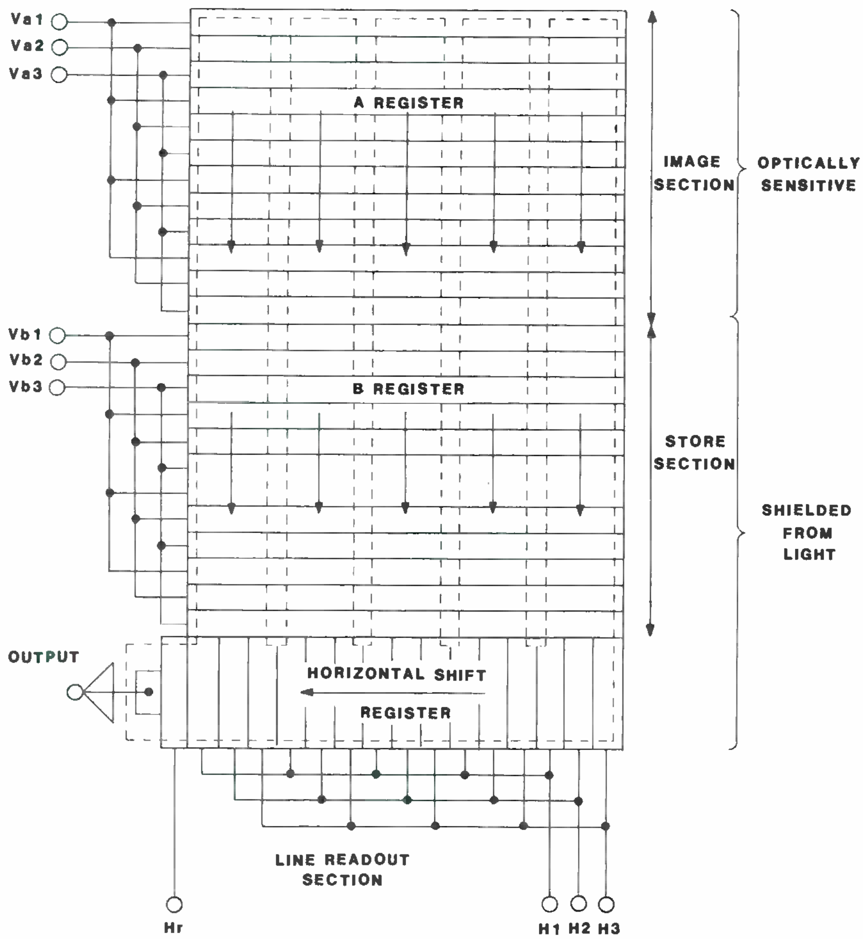


Fig. 19. CCD frame transfer array schematic.

production are similar to those of the studio, except that the camera system must also be easily transported and capable of both handheld, or tripod operation. The EFP camera, therefore, should have the performance of a studio camera, combined with the virtues of a good ENG camera.

Ideally, the EFP camera should be portable, but with the ability to adapt to large "EFP" lenses and to be remote controlled from considerable distances up to 6,000 feet. Such cameras provide great flexibility in both field production and sports applications.

The Sports Camera

This term that can be applied to any camera that is used in sports coverage. As discussed above, "studio" cameras are frequently used in this application, which is also ideal for the EFP camera.

Several factors are of prime importance in a camera used for sports coverage: The camera must accept a lens with a large zoom range if it is to cover a ball game effectively from high in the stadium, follow the action around a race track, or at a golf tournament. The sports camera must also be remote controllable from the great-distances encountered in these applications.

THE OPTICAL SYSTEM

Lens Parameters

Focal Length determines the angle of view of the lens, which relationship depends on the camera image format, as shown in Fig. 20. For example, a 9.0mm focal length in ⅔-inch format gives the same angle of view as 13.0mm in 1-inch format, or 18.0mm in 1¼-inch format.

Angle of View is related to focal length as shown in Fig. 20. Because of the 4:3 aspect ratio of our television standard, the angle of view is greater in the horizontal direction than in the vertical direction.

Field of view is related to the angle of view as follows:

$$\text{Field of View} = 2D\tan\,(\text{angle of view}/2)$$

Where D is the distance from the camera to the object of view. For example, if the object of view is 1,000 feet away and the lens has an angle of view of 2 degrees;

$$\begin{aligned}\text{Field of View} &= 2 \times 1{,}000 \times \tan\,(2°/2)\\ &= 2{,}000 \times 0.0175\\ &= 35 \text{ feet}\end{aligned}$$

This situation could represent a camera located 1,000 feet from the starting gate at a race track.

Relative aperture (F/Number) describes the speed, or light gathering power of the lens. The relative aperture is defined as:

$$\text{F/Number} = \frac{\text{Focal Length}}{\text{Effective Diameter}}$$

Because the light gathering power of a circular aperture is proportional to the square of its diameter, the light gathering power, or speed of a lens is inversely proportional to its relative aperture, or F/Number.

For example, the typical sensitivity of an ENG Camera is 200 foot candles at F/4.0. If the camera lens has an F/1.6 maximum relative aperture, the following illuminations are required

F/Number	Illumination
4.0	200 ft candles
2.8	100
2.0	50
1.6	32

Switching in 12 dB added gain (×4) would give such a camera a maximum sensitivity of 32/4 = 8 ft. candles.

Ramping is a term applied to the increased F/Number (reduced speed) of some lenses as the focal length is increased. In the example of Fig. 21, the lens has a relative aperture of F/1.6 through a zoom range of 12mm to 120mm, but may "Ramp" down to F/2.8 at its maximum focal length, 180mm. For an ENG camera with the sensitivity indicated in the above example, 32 ft. candles would be sufficient illumination at a focal length of 12mm to 120mm, but 100ft candles would be necessary at 180mm.

Ramping is an inevitable result of a small objective element at the front of the zoom lens. In ENG Camera lenses the objective element is deliberately small to minimize the overall weight and some ramping is not uncommon. Ramping is only a problem if there is insufficient illumination for the lens to be used at its maximum focal length.

Modulation Transfer Function (MTF) defines the resolving capability of the lens and is measured by looking at a 100% amplitude sine wave bar pattern and determining the depth of modulation obtained in the image. With few exceptions, the MTF is better in the center of the picture and decreases towards the corners. The difference between the center and corner modulation is less when the lens is stopped down than when the iris is wider open.

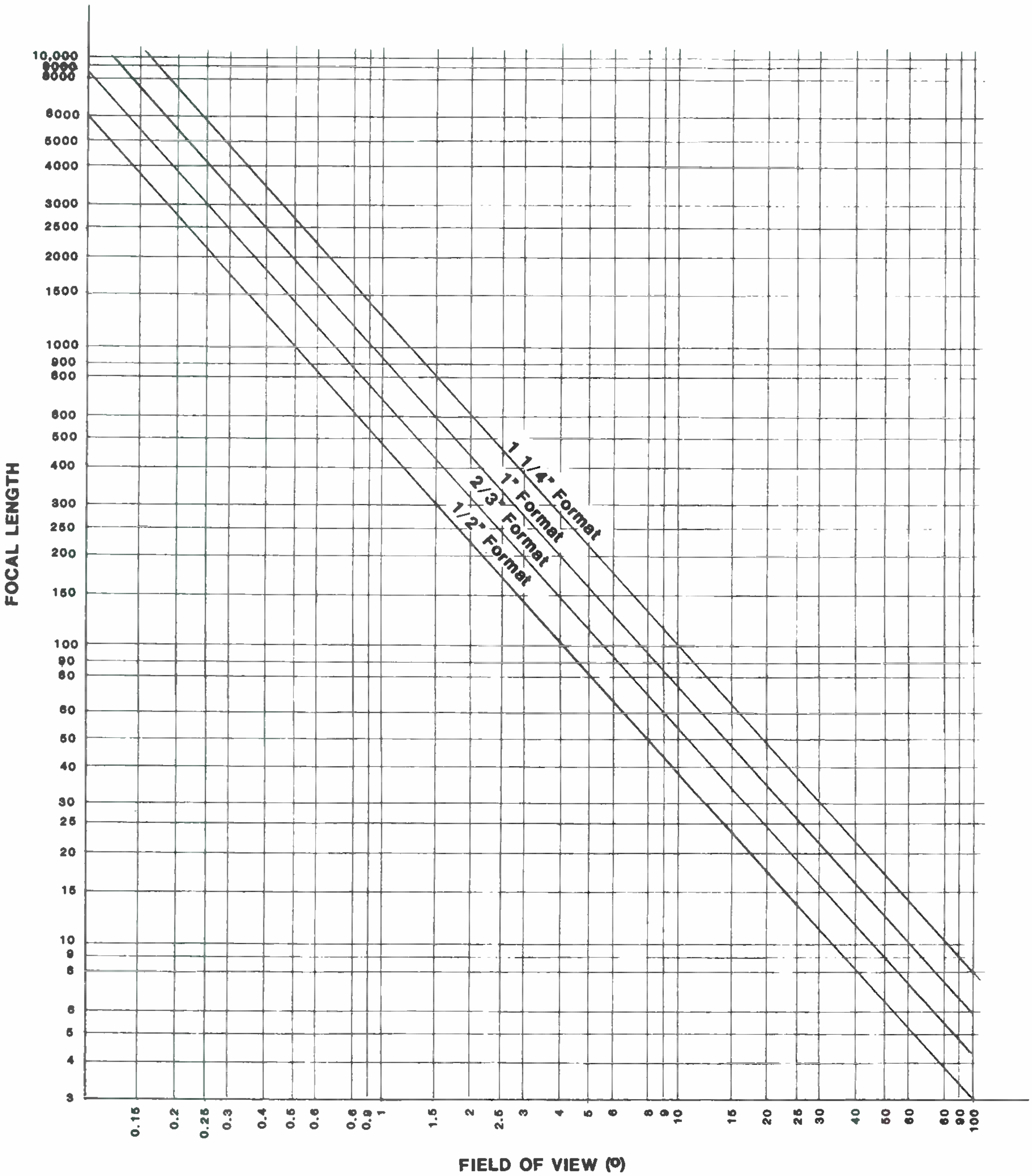


Fig. 20. Focal length vs. horizonatal field of view.

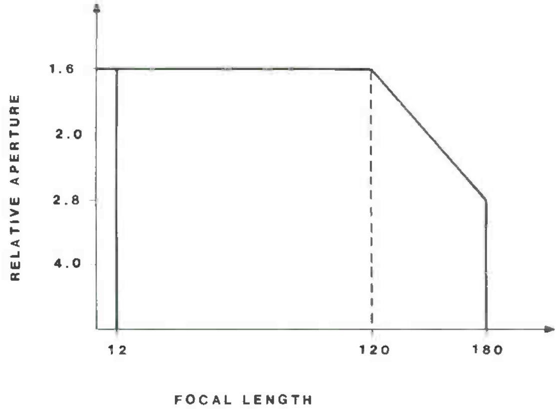


Fig. 21. Lens ramping.

Most lenses exhibit their best MTF when operated in the middle of their aperture range. For example, a lens with aperture range of F/1.6 to F/16 will probably give its best modulation at about F/4.0. If the lens is further stopped down towards F/16 some small reduction in modulation may result due to the light being defracted as it passes through such a small aperture. However, as the lens is opened towards its maximum aperture of F/1.6 a more severe loss of modulation depth is likely to occur as the spherical aberrations become more significant.

The MTF characteristic of a typical zoom lens is shown in Fig. 22. Note that the MTF becomes worse as iris is opened and that this difference is more exaggerated in the corners, which illustrates the importance of evaluating a lens at its maximum aperture.

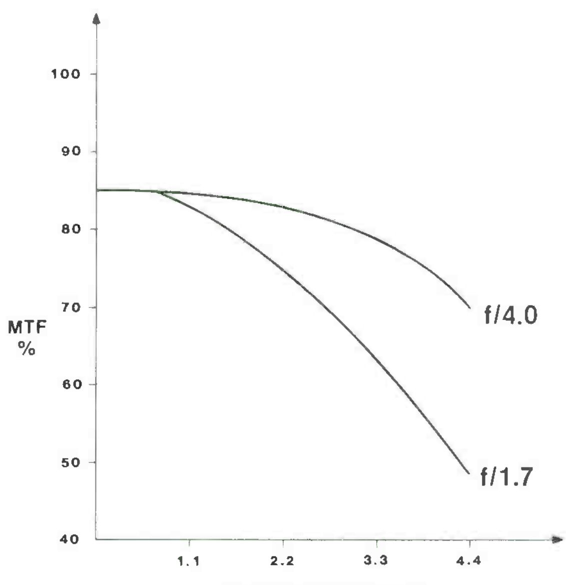


Fig. 22. Typical MTF characteristic for ⅔ inch lens. (average for all focal lengths)

Uniformity of field (vignetting) refers to the response to a white field over the full picture area. The typical uniformity of field characteristic of a zoom lens is shown in Fig. 23, which illustrates that the response is usually greatest in the center and least in the corners. The non-uniformity in the corners becomes more pronounced at wide aperture, and varies with focal length. Again, it is important to evaluate a lens at maximum aperture, when such characteristics are at their worst.

Longitudinal chromatic aberrations. Although the lens designer makes a great effort to achieve consistency, the focal length of a zoom lens tends to vary slightly with color throughout its zoom range. This means that there will be some small differences between the color channels, in both focus and image size, as the zoom is operated.

The spectral variation in focal length is called the longitudinal chromatic error and is, to some extent, inevitible with today's state of the art. In general, the more severe the constraints that are placed on the lens designer, such as the need for a smaller, faster lens for a portable camera, the more difficult it is to keep these aberrations small.

Usually, the lens designer can decide where the chromatic aberrations will be distributed. Often he will decide that most of the error will be in the blue, because this channel contributes less to the overall picture quality.

The longitudinal chromatic aberration for a typical zoom lens is shown in Fig. 24. Note that the red and blue channels tend to have a fixed focal length error with respect to green, plus some error that varies with focal length. The fixed component of error can be corrected using the focus tracking adjustments of the red and blue yokes, however the variable error will inevitably result in some degree of defocussing at certain focal lengths.

Similarly, the average value of the magnification error can be corrected in a tube type camera by adjusting the scanned raster sizes, however the variable error component that changes with zoom will result in some slight misregistration at certain focal lengths. It is best to set up the camera registration at some point in the middle of the focal length range, so that the variable errors are evenly distributed throughout the zoom range.

The CCD camera presents a special problem, because the image size is fixed. Lenses designed for CCD cameras must have minimum chromatic aberrations.

Geometric Distortion. All lenses exhibit some degree of geometric distortion, which is a symmetrical error, as shown in Fig. 25a, effecting the red, green and blue images equally. In the case

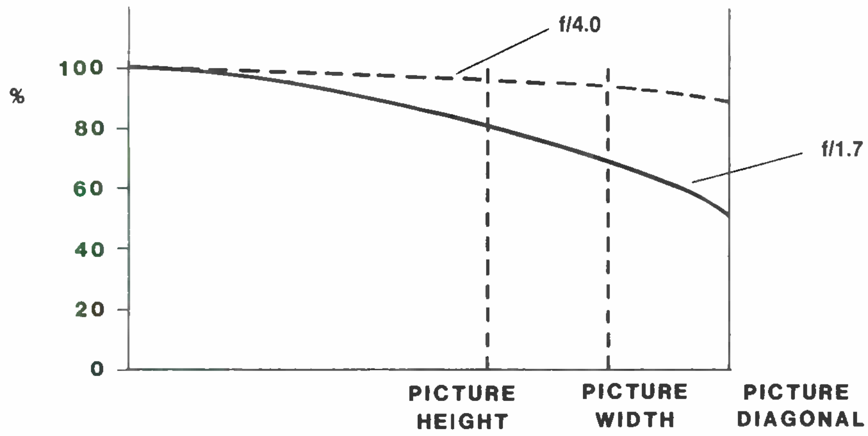


Fig. 23. Uniformity of field.

of the zoom lens, the distortion varies with focal length (Fig. 25b), such that the negative (barrel) distortion occurs at short focal lengths, and positive (pincushion) distortion occurs at long focal lengths. Again, it is advisable to set up the camera at some point in the middle of its zoom range, where the average value of distortion is present.

Minimum Object Distance (MOD) is the closest focussing distance of a lens. Wide angle lenses generally have the smallest MOD and telephoto lenses the longest. This is a most important factor in choosing a lens to ensure that the MOD is adequate for the particular application.

The Relay Type Beamsplitter

The relay type beamsplitter is shown in Fig. 26, in which the taking lens focusses the scene onto the field lens, which is an intermediate focal plane. The re-imaging lenses, in turn, focus the field lens image onto the imaging devices via the dichroic mirrors, which provide the beamsplitting function.

The first mirror is usually a red reflecting dichroic surface, with the transmission characteristic shown in Fig. 27a, which separates out the red component of the image by reflecting it back to the first surface mirror, from which it is focussed onto the imaging device, via the re-imaging lens and red trim filter. The transmission characteristic of the trim filter combines with that of the red dichroic mirror to produce the overall red response curve required in Fig. 27c.

The blue-green light passes through red dichroic filter and the blue component is reflected by the blue dichroic mirror, with the transmission characteristic shown in Fig. 27b. The blue image

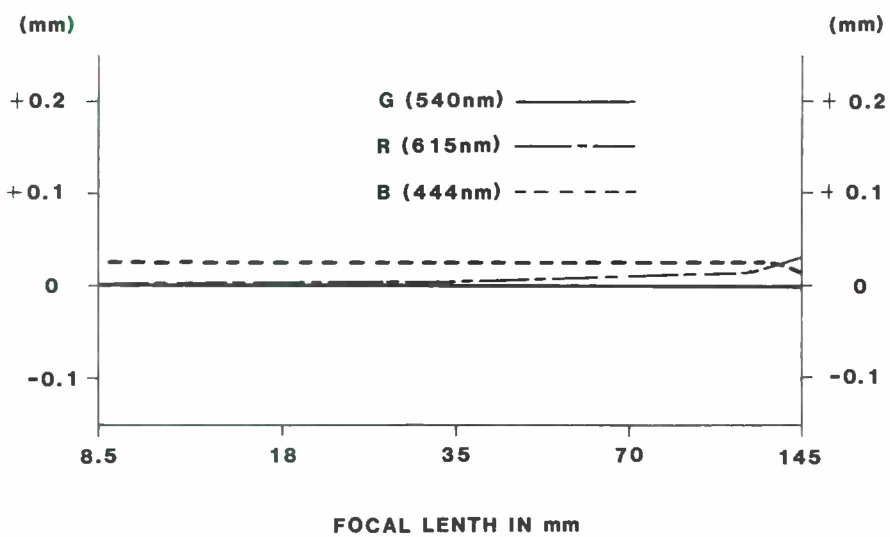


Fig. 24. Longitudinal chromatic aberration.

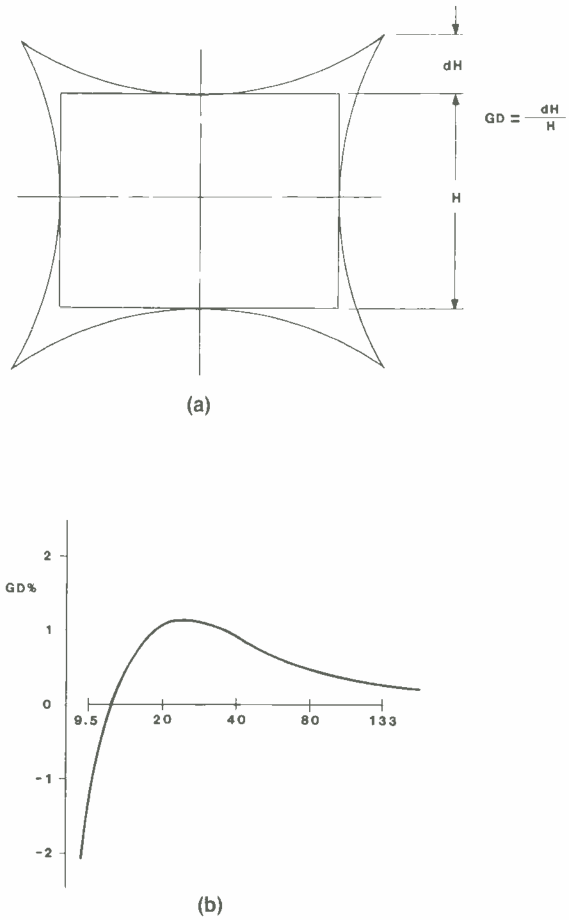


Fig. 25. Geometric distortion.

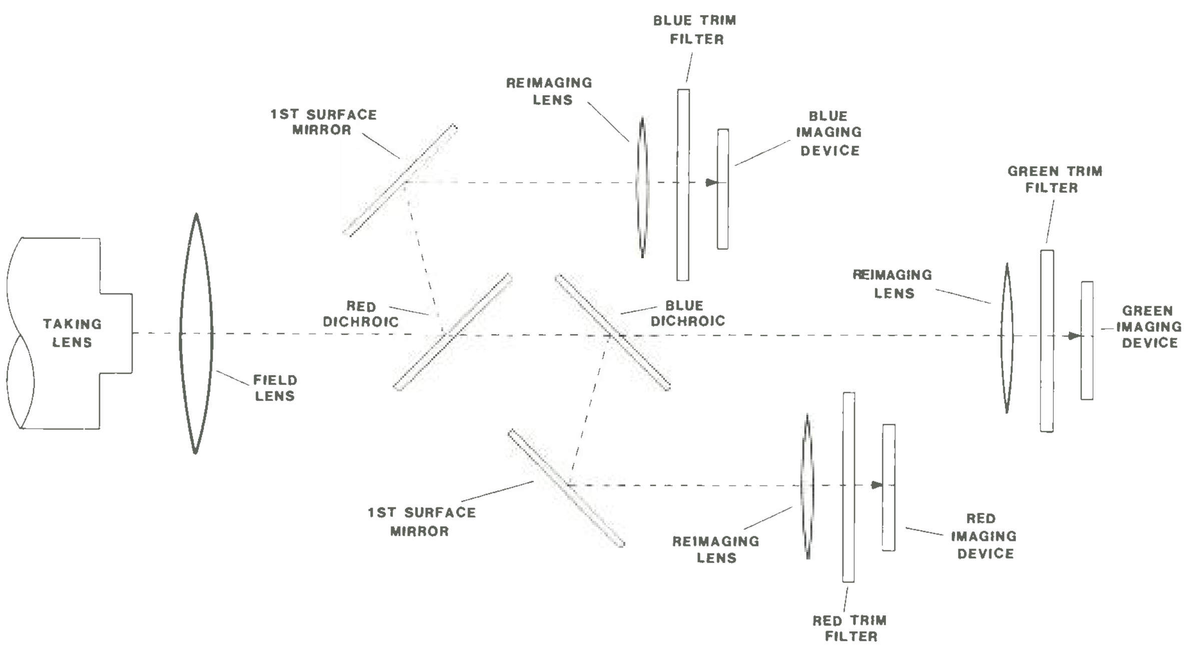
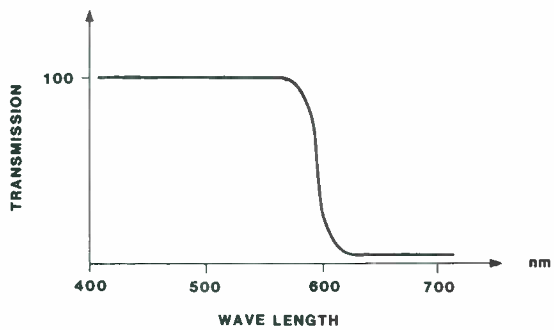


Fig. 26. Relay type beamsplitter.

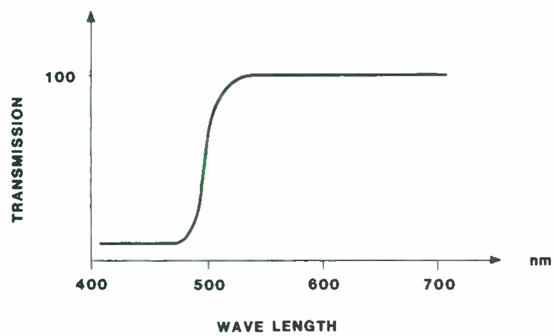
is then focussed onto the blue channel imaging device, via the remaining first surface mirror and blue trim filter. The remaining green light passing through the blue dichroic mirror is focused onto the green channel Imaging Device via the green trim filter. The overall response is shown in Fig. 27c.

The purpose of the field lens is to ensure that the maximum amount of light from the taking lens is directed into the re-imaging lenses, otherwise extremely poor sensitivity and vignetting would result.

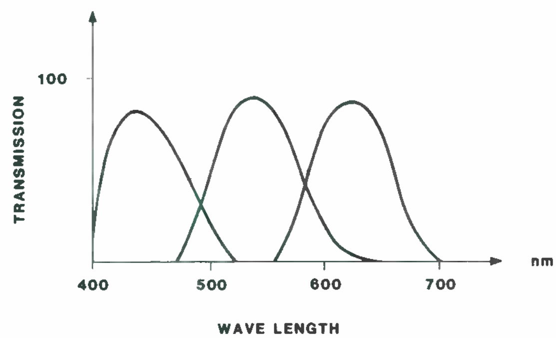
There are several disadvantages in the relay type beamsplitter for broadcast camera applications; it is relatively large and, unless fast and expensive re-imaging lenses are used, it has lower sensitivity, resolution and more vignetting than a prismatic system. It also requires more surfaces than its prismatic counter part, which results in increased transmission loss and flare.



a) RED REFLECTING DICHROIC FILTER



b) BLUE REFLECTING DICHROIC



(c) TAKING CHARACTERISTIC

Fig. 27.

The Prismatic Beamsplitter

As shown in Fig. 28, the prismatic beamsplitter is usually constructed of three precision blocks of glass. The taking lens must have a long back focal distance, in order to focus the image through the prism onto the imaging devices.

Light from the taking lens enters prism block (a), the front surface of which is often coated with an infrared blocking dichroic filter to prevent this light reaching the imaging devices, which would otherwise impair the color fidelity. The red light is reflected by the dichroic mirror on the first surface of prism block (b), which has the characteristic shown in Fig. 27a, and, because of the acute angle it makes with inside of the first surface of block (a), it is totally reflected through the red trim filter onto the red imaging device.

The green-blue light passing through the red dichroic filter is similarly reflected by the blue reflecting dichroic mirror on the first surface of block (c) and totally reflected into the blue imaging device. The green light continues onto the green imaging device.

As with the relay system, the trim filters tailor the overall spectral response so to that required in Fig. 27c.

The principle advantage of the prismatic beamsplitter is that it is small and has good mechanical stability. Also the prism can be designed to pass a large cone of light, which means that it can accept a fast lens. Typical lens apertures for existing prismatic beamsplitters are as follows:

Format	Typical Aperture
1¼″	F/2.1
1″	F/1.5
⅔″	F/1.2
½″	F/1.2

Filters

Color filters are usually housed in a wheel between the taking lens and beamsplitter. The following filters are commonly used;

CLEAR	—Clear glass
WRATTEN 85B	—Converts daylight to studio color temperature, 3,200 degrees Kelvin.
0.6 ND	—Neutral density filter reduces camera sensitivity by 4 times.

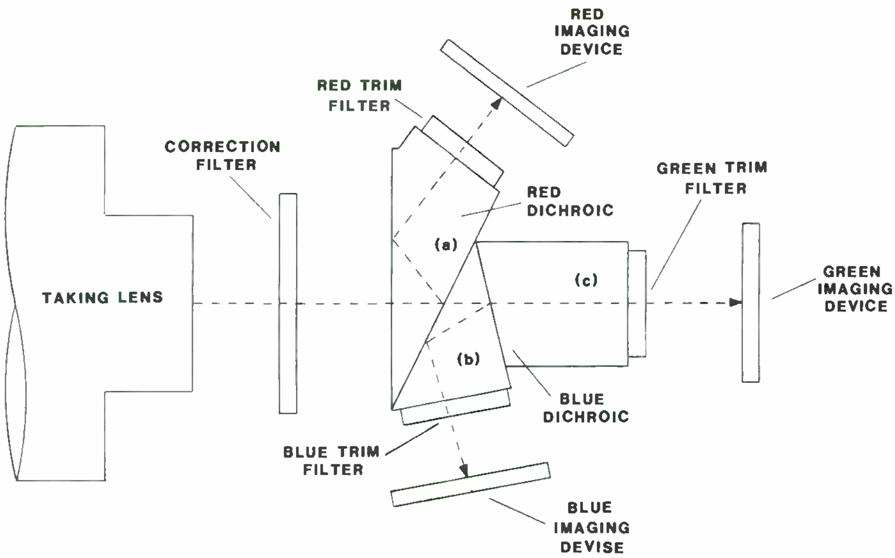


Fig. 28. The prismatic beamsplitter.

1.0 ND	—Neutral density filter reduces camera sensitivity by 10 times.
1.0 ND + 85B	—Reduces sensitivity by 10 times and converts daylight to 3,200 degrees Kelvin.

CAMERA TUBES

Two types of camera tubes are currently used in broadcast television cameras; those with a Lead-oxide photosurfaces, and the "Saticon" (trademark Hitachi Ltd), which uses a photosurface consisting of a selenium, arsenic and tellurium compound.

Lead-Oxide Camera Tubes

The Lead-oxide camera tubes are marketed under various trade names; the Plumbicon (trademark Philips ND), and the Leddicon (trademark English Electric Valve Co. Ltd). The lead-oxide camera tubes are fundamentally similar, consisting of a complex layer of lead monoxide formed by evaporation techniques. Because part of the layer is formed by evaporation in a partial pressure of inert gas, the lead-oxide surface has a fluffy texture, which results in a lower capacitance for reduced lag.

The standard lead-oxide layer has a spectral response, which peaks in the green-blue part of the spectrum and does not exhibit good red sensitivity. For the red channel an "extended red" version of the tube is used, which is formed by further doping the layer to enhance the red sensitivity.

The Saticon

The Saticon has a glass-like layer which is amorphous in structure and gives a more uniform spectral response than the standard lead-oxide surface, so that a special red channel tube is not necessary and flare is reduced.

Comparison Between Lead-Oxide and Saticon Tubes

Due to competitive pressures, these two camera tubes have advanced steadily and their performances are very similar. Because it is normal for there to be a significant tolerance in the various performance parameters of both tube types, it may not be possible to decide which tube type is best from a single comparison.

Camera Tube Set-Up

Before attempting to measure any camera tube characteristics, it is necessary that the device be set up in accordance with the manufacturers recommendations. In general, the following set-up adjustments are necessary:

X and Y Alignment. Incorrect beam alignment can significantly impair the camera tube performance. Usually a camera is equipped with a focus "rock" facility to aid in accurate alignment. If focus rock is not available, manually rock the focus control a little each side of true focus and adjust the X and Y alignment control until the focus rocking causes the center of the picture to rotate without horizontal, or vertical movement.

Signal Current. The correct signal current is necessary for best performance. If the signal current is too large, beam pulling can result and, together with the excessive beam current required, reduced resolution can result. If the signal current is too small, the signal to noise ratio will be impaired. The following signal currents are usually recommended;

Format	Signal Current
30 mm	300-400 nA
25 mm	220-300 nA
18 mm	180-200 nA
13 mm	150-200 nA

Beam Current misadjustment can also significantly impair performance. If the beam current is too small, obviously the target will not be adequately discharged, resulting in greyscale distortion and blooming.

The effect of too much beam, however, can be more subtle. As the beam current is increased the average energy of the electrons in the beam changes, in particular in the radial direction, and the cross-section of the focussed beam becomes larger. Excessive beam, therefore, results in reduced resolution, increased build-up lag and, in severe cases, can produce a distortion of the raster, which impairs registration.

The correct setting of the beam depends on whether the camera employs ABO (Automatic Beam Optimization), or not. A standard logarithmic greyscale (Fig. 29) is a convenient test pattern for this purpose.

Adjusting the Beam Without ABO—make sure that adequate beam is available and adjust the lens iris to obtain 100 IRE units of signal, then open one more f/stop. Reduce the beam and note that the when insufficient beam is available the signal level reduces and the white chip of the grey scale is not fully discharged, so that its edges bloom and become ill-defined. Increase the beam current until the white chip is visibly discharged, the blooming effect disappears and the edges "snap" into place.

Adjusting the Beam Current With ABO—Because the various ABO systems require different set up, refer to the manufacturer's instructions.

SOLID STATE IMAGERS

Imager Types

The solid state imagers that are currently used in broadcast cameras are of the CCD (Charge Coupled Device) type. These imagers are organized into horizontal rows and vertical columns of image elements, known as "Pixels". The greater the number of pixels, the higher the resolution of the device.

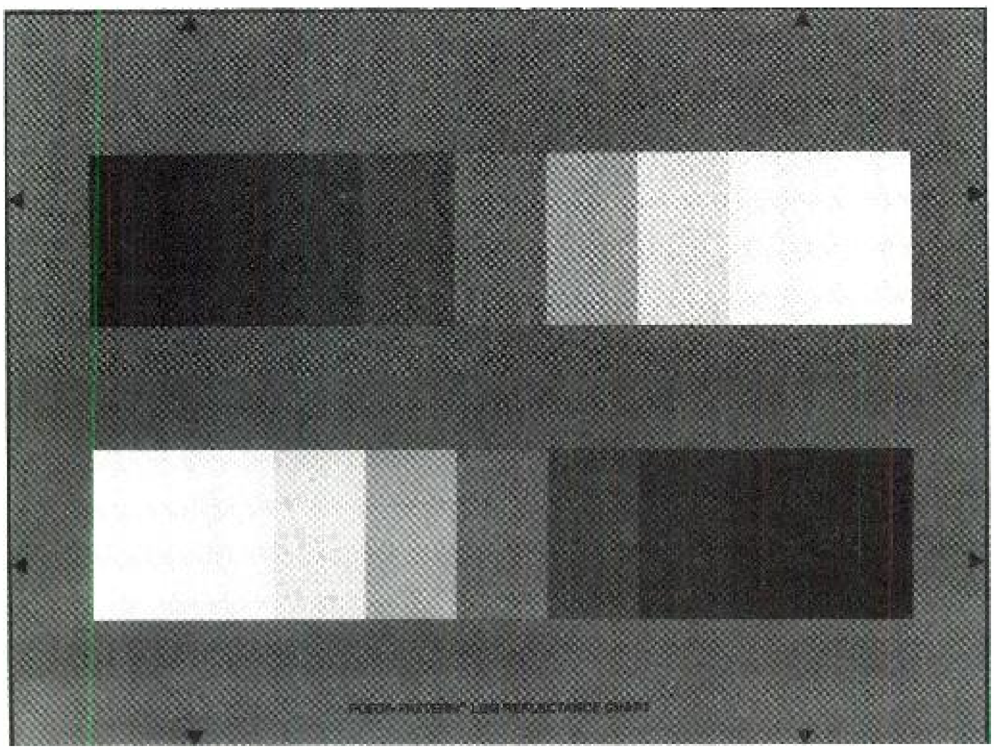


Fig. 29. 9-step log grey scale chart.

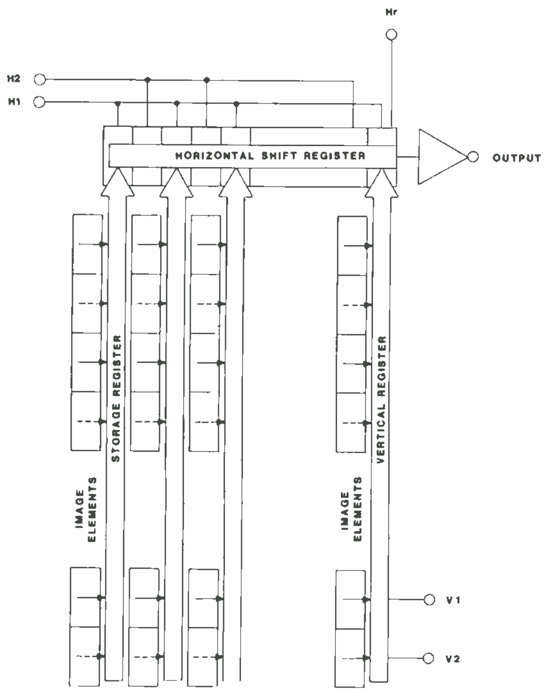


Fig. 30. Interline transfer CCD.

The charge pattern formed in the CCD pixels is allowed to integrate during the unblanked frame period, and is then transferred to a separate storage register during the vertical blanking interval. The stored image is then transferred row by row into a horizontal register, from which it is clocked out during the unblanked horizontal period. This transfer process passes the charge pattern from pixel to pixel, hence the term "Charge Coupled Device".

There are two basic kinds of CCDs; the "interline transfer" type uses a storage register, which is integrated with the image register, as shown in Fig. 30, the "frame transfer" type uses a separate storage register, as shown in Fig. 19. The interline transfer type of CCD has the disadvantage that almost half of the image area is used for the storage register, which reduces sensitivity and increases aliasing. However the completely separate storage register of the frame transfer device requires almost twice as many transfers, which means that the transfer efficiency must be very high to avoid resolution loss.

CCDs may be of the "front-lighted" type, in which case the illumination must pass through the address structure of the device, consisting of evaporated metal, or poly-silicon, resulting in a significant light loss, particularly in the blue.

In the case of the "back-lighted" CCD, an extremely thin silicon wafer is be mounted on a glass substrate and the light enters the device from the glass side, thereby avoiding losses through the address structure.

Imager Set-Up

Set-up requirements for the various imagers differ significantly and the user should refer to the manufacturers instructions for details. Basically, however, all imagers require certain drive pulse levels and voltages for efficient operation. Failure to make the correct adjustment can result in poor performance, such as low resolution, blooming, smear and poor signal to noise ratio.

Imager Resolution

The imager is a spatial sampling device with a characteristic Sin x/x response, Where $x = L_cF_s$. F_s is the spatial frequency and L_c is the pixel dimension. For a CCD with 403 horizontal elements, the spatial frequency of the pixels is 7.6 MHz and, according to Nyquist's Theorum, the maximum spatial frequency that can be resolved is 3.8 MHz (304 TV lines).

However, the theoretical response at the Nyquist limit is 64 percent and there is significant response above this limit due to "aliasing" or "moire" effects. It was earlier supposed that the frequencies above the Nyquist limit should be suppressed by the use of an optical low pass filter to prevent beat patterns. However, more recent studies indicate that there is some merit in allowing some aliasing to occur, which effect yields a better subjective sharpness than would be expected from consideration of the Nyquist limit alone.

Comparison Between Solid State Imagers and Camera Tubes.

Resolution

The response of the solid state imager is limited by the number of horizontal pixels, whereas the camera tube resolution is limited by its scanning beam spot size and the photosurface lateral leakage. The MTFs of a 2/3 inch format Saticon and a 403 horizontal pixel CCD are compared in Fig. 6. Using some aliasing can produce a desirable effect, the subjective sharpness of portable cameras using these devices can be similar.

Sensitivity

The back-lighted, frame store CCD, which avoids the light loss in the address structure, can provide approximately the same sensitivity as the camera tube. Front lighted imagers are about ½ to 1 f/stop less sensitive.

Signal To Noise Ratio

Primarily due to the very small capacitance of the floating diffusion zone in the CCD (approximately 0.07 pf) the imager is capable of an extremely high signal to noise ratio, 64 dB, or more. By comparison, the camera tube S/N ratio is limited by the total capacitance shunting the preamplifier, to approximately 57 dB.

Stability

The mechanical and thermal instability of the camera tube's electron gun prohibits long term registration stability, and the photoconductor characteristics slowly change, so that the color balance must be occassionally adjusted. The imager, however, has stable sensitivity and no mechanical parts to move.

Lag and After Image

As discussed later, the mobility of the charge carriers in the photoconductor, and the inability of the electron beam to discharge the target in a single field, give rise to significant lag and after image effects in camera tubes. On the other hand, these phenomenon are completely lacking in the solid state imager.

THE MEASUREMENT AND PERFORMANCE OF IMAGING DEVICES

Limiting Resolution

This is a somewhat arbitrary and not so accurate measurement of an imaging device's resolution, but is conveniently performed with the resolution chart shown in Fig. 31. To be meaningful, however, this measurement must be performed using a high resolution monitor and a video system with adequate bandwidth. A lens with known resolution should be used and the illumination adjusted so that the lens iris can be set at approximately three f/stops closed from its maximum aperture. For example, an f/1.6 lens should be set at about f/4.0. Cameras tubes should be aligned before making this measurement.

To measure the limiting resolution, the camera is adjusted to frame the resolution chart, such that the tips of the alignment triangles coincide with the raster edges. The camera is focussed so that best center resolution is obtained. The limiting resolution is determined from the resolu-

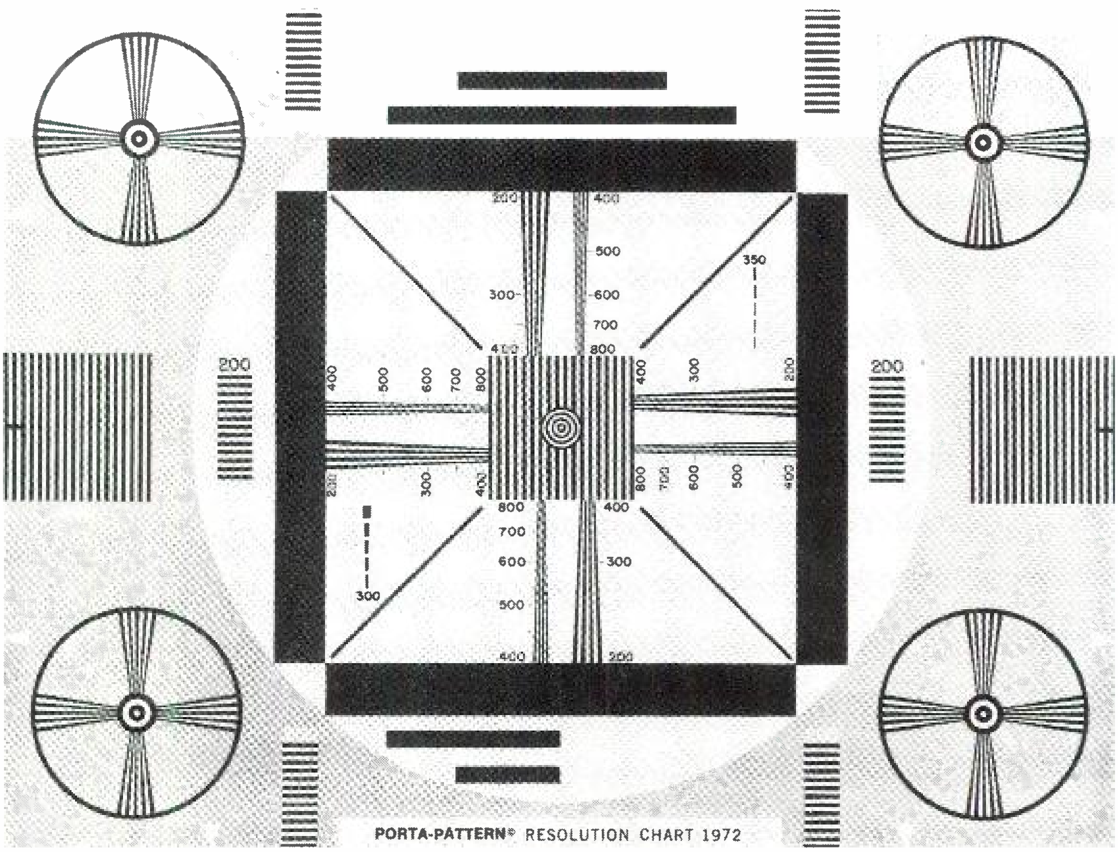


Fig. 31. Resolution chart.

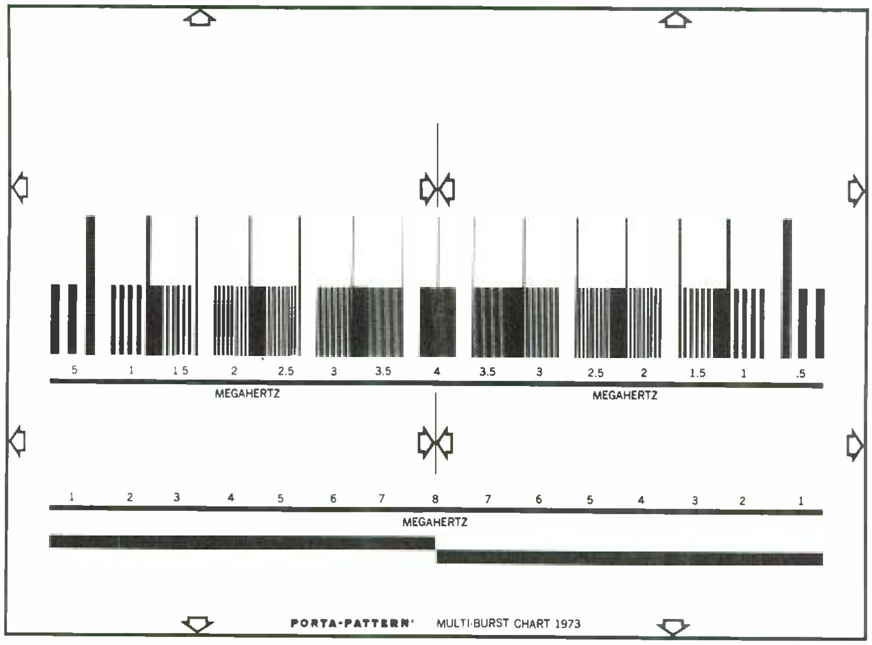


Fig. 32. Multi-burst chart.

tion wedges and defined as the point beyond which the black and white bars can no longer be resolved. Center resolution is determined from the center wedge, corner resolution from the corner wedges.

Amplitude Response

This is a more precise measurement of an imaging device's resolution using a standard burst pattern (Fig. 32) and a waveform monitor with line selection capability. The same lens and video system requirements apply as in the measurement of limiting resolution. The camera aperture corrector and gamma correction should be turned off.

Adjust the camera to accurately frame the burst pattern so that the tips of the alignment triangles coincide with the raster edges. Position the line selector so that the line passing through the center of the bursts is displayed and adjust the lens iris and master black level control so that the signal from the 0.5 MHz burst ranges from 0 to 100 IRE units.

The Amplitude response is determined from the peak to peak signal corresponding to each burst relative to the 0.5 MHz amplitude. Usually, the amplitude response of a studio camera is measured at 5.0 MHz, which corresponds to 400 TV lines, and a portable camera is measured at 4.0 MHz, which corresponds to 320 TV lines.

The burst pattern waveform also provides a simple method of checking the video amplifier response. If the amplifier has a flat response the waveform from the 0.5 MHz burst will have the symmetrical shape shown in Fig. 33a. If, however, the amplifier is over peaked, the waveform will exhibit an overshoot similar to that shown in Fig. 33b. An amplifier with poor high frequency response will give the waveform of Fig. 33c.

An accurate amplitude response measurement can only be made if the amplifier has a flat response. An over-peaked amplifier will not only give an unrealistically high response measurement, but will also result in ugly picture edges. An under-peaked amplifier will give poor response measurements and a "soft" picture.

Sensitivity

The imaging device manufacturer usually expresses sensitivity in microamps/lumen, but this measurement does little to inform the user what sensitivity he can expect from his camera.

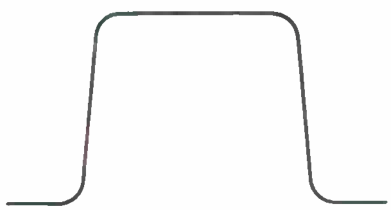
The camera manufacturer's specification, however, is more useful to the broadcaster, because it tells what lens aperture is required to achieve full output (100 IRE units) from the green channel at a given illumination. For example, an ENG Camera's sensitivity is usually specified as 200 fc at f/4.0, for a 60% reflectance white, with 3,200 degrees Kelvin illumination.

Several factors affect camera sensitivity:

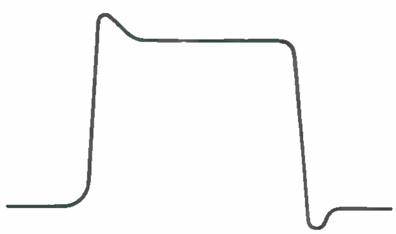
- *The taking characteristic, or spectral response of the beamsplitter.* As this parameter is fairly well defined for good colorimetry, most cameras are similar in this regard.

- *The choice of imaging device.* Tube type cameras, using lead oxide and Saticon type tubes, have similar sensitivity. The sensitivity of cameras with solid state imagers, however, depends greatly on the type of imager used, as explained in (6.1). In general, cameras, which use back-lighted frame-transfer type CCDs (Charge-coupled Devices), can provide a sensitivity similar to tube type cameras. Cameras using other types of front-lighted imagers tend to be a half of one f/stop less sensitive than tube type cameras.
- *The imaging device operating level*, which is the recommended signal current in the case of a camera tube, or a percentage of the saturation level in the case of a solid state imager.

a) CORRECT RESPONSE



b) EXCESSIVE HIGH PEAKING



c) POOR HIGH FREQUENCY RESPONSE

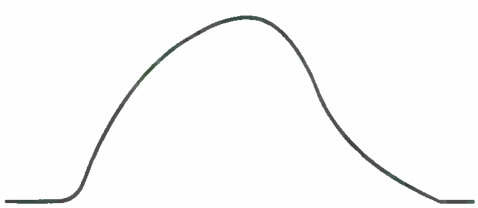


Fig. 33. Effect of frequency response on the reproduction of a white bar.

For example, a 25mm tube type camera operating at a green channel signal current of 300 nanoamps will require 36% more illumination than the same camer tube at 220 nanoamps.

- *The choice of lens.* The imaging device illumination for a given lens aperture is given by:

$$\text{Illumination} = \frac{TB}{4f^2(1+M)^2} \quad \text{footcandles}$$

Where:
T = lens transmission
B = scene brightness in footlamberts
f = relative aperture
M = magnification from scene to device

M is usually small enough to be neglected. Usually T has a value of approximately 0.8, which means that there is a 20% light loss through the lens. Larger and more complex lenses employ a larger number of optical elements, which means that the transmission losses are likely to be greater than those of a more simple lens.

Camera sensitivity is conveniently measured by using the logarithmic greyscale shown in Fig. 29 and a 3200 degree Kelvin source of illumination, such as standard studio quartz-halogen lighting.

If a reflectance chart is used, make sure the lighting is uniform, hold an accurate light meter against the white chip and set the illumination to the required level, say 200 footcandles. Adjust the lens iris to obtain 100 IRE units from the green channel and record the f/number as the measurement of sensitivity. Usually a new test pattern with a clean white chip has a 60% reflectance, so this measurement can be directly compared with the camera manufacturer's specification.

If back-lighted test pattern and light box is used the white chip brightness should be adjusted using a spot brightness meter. However, in order to compare with the cameras manufacturers reflectance chart specification, the brightness of the white chip should be set to 60% of the reflectance chart illumination. For example, a 120 footlambert transmission chart white chip provides the same brightness as a 60% reflectance chart with 200 footcandles illumination.

Lag

This term is the short term persistence characteristic of tube type cameras. "Build-up" lag refers to the delayed signal increase that occurs after illumination is suddenly applied. "Decay" lag is caused by the inability of the scanning beam to completely discharge the target each field. The camera tube manufacturer uses relatively sophisticated instrumentation to measure lag, which is usually specified as the percentage of signal present three fields after the illumination has been changed. Usually, the broadcaster must use more simple methods to judge lag.

Lag is more pronounced at low signal currents and, because the blue channel signal current is relatively small, such cameras exhibit predominantly blue lag. This can be illustrated by slowly panning the camera backwards and forwards across a greyscale chart (Fig. 29). Note that, as the camera is panned to the left, the white chip exhibits a yellow build up at its leading edge and the black chip shows a magenta decay at its trailing edge.

The presence of build-up and decay lag limits the amplitude response in the case of moving objects, thereby giving rise to a "dynamic resolution", which is lower than that obtained in a static scene.

Because the pixels of the solid state imagers are completely emptied each field, lag is not a problem in these devices.

After Image

This is a long term persistence characteristic in camera tubes, but is usually not a problem unless the tube is defective, perhaps through old age. After image is usually expressed as the time taken for the residual signal from a prolonged 100% highlight to decay to 1%.

Highlight Sticking

This refers to a phenomenon in camera tubes, whereby a long term persistence follows an excess highlight, such as a specular. Usually, highlight sticking is not a problem with lead oxide tubes, but in the case of the Saticon, excess illumination can temporarily collapse the field across the target, so that the charge carrier mobility is reduced and the image decays more slowly, with the effect that a moving highlight leaves a trail across the picture. The occurrance of this problem is greatly reduced in cameras, which employ ABO.

Highlight Overload in Imagers

In conditions of excess illumination the pixel wells become filled and the excess charge can "spill" into adjacent pixels in the same column. If the imager design does not incorporate "drains" to carry the overflow to the substrate, excess highlights are characterised by vertical white lines passing through the highlight and usually extending the full picture height.

Smear in Imagers

A characteristic vertical smear effect occurs in frame transfer CCDs, if the device is not shuttered during the period when the image is being transferred from the "A" register to the "B" register. This is particularly noticable in highlight areas, and is caused by the illumination creating a charge pattern trail while the image is being moved between registers.

CAMERA PERFORMANCE

System Parameters

The overall performance of the camera system depends not only on the performance of its lens, optical system, or imaging device, but also on the effectiveness of its electrical and mechanical design. For example, the factors affecting picture quality depend as much on the performance of the camera electronics, as it does on the lens, beamsplitter and imaging device.

The performance contributions of the optical system and imaging device have already been considered and this section will deal with those independent camera parameters, which have an important influence on the overall system performance.

Basically, the important camera characteristics concern set-up accuracy and stability, overall picture quality, ruggedness and reliability.

Camera Set-Up and Evaluation

Because each camera differs in some way it should be set up precisely according to the camera manufacturers instructions before attempting to evaluate its performance. The following evaluation sequence is recommended:

1. *Color channel performance.* If possible, turn off the image enhancement, frame the resolution chart of Fig. 31, make sure the camera is focus tracked and observe the picture from each of the color channels in turn, both on the picture and waveform monitors.

Visually, the individual pictures should demonstrate good resolution, both in the center and corners. Providing a good lens and black and white monitor are used, the following limiting resolutions should be obtained in the center:

	RED	GREEN	BLUE
1¼ and 1 inch tubes	500	600	700
⅔ and ½ inch tubes	400	500	500
solid state imagers *	300	300	300

* For an imager with 400 horizontal pixels.

Observe the waveform from the burst chart of Fig. 32 and verify that good amplifier response is indicated by a symmetrical waveshape from the 0.5 MHz bar, as shown in Fig. 33a. Note the depth of modulation obtained from each channel. Results may vary considerably for different cameras and imaging devices, so the following is only a guide to level of response expected:

	RED	GREEN	BLUE	BURST FREQUENCY
1¼ and 1 inch tubes	50%	55%	60%	5 MHz
⅔ and ½ inch tubes	40%	45%	50%	5 MHz
solid state imagers	45%	45%	45%	3.8 MHz

Basically, the premise is; if the camera cannot produce good pictures from the individual color channels, then it is unlikely to make a good composite color picture.

2. *Registration*. Frame the registration chart of Fig. 34 and observe the registration accuracy. Registration is usually specified in three zones:

Zone 1—A circle equal to 80% of picture height.
Zone 2—A circle equal to picture width.
Zone 3—The area outside Zone 3.

One relatively simple way of judging the camera's registration performance is to remember that the width of the lines in the registration chart is 0.2%. Therefore, for example, in the case of a 0.2% error between two channels, the corresponding displayed lines would spread and touch. For a 0.4% error, the lines would be separated by a gap equal to the width of a line.

It is also important to check a camera's registration stability, by remeasuring after it has been operating for some time. It is also a good idea to see how the registration holds up to shock and vibration, for example, after carrying the camera in the trunk of a car, or wheeling the tripod across a bumpy parking lot, particularly if it is a portable, or sports camera.

3. *Grey-scale tracking*. Using the pattern in Fig. 29, check to see if the camera is color balanced on all steps of the grey-scale. This may be conveniently done by observing the color monitor and switching the chroma on and off. If the grey-scale is perfectly balanced, there will be no change in color. Open the iris so that pattern is overexposed and the at least half the grey-scale goes into the white clipper. Observe that the black chips are still color balanced, which is a good test as to whether the flare correction circuits are correctly adjusted and operating well.

4. *Image enhancer performance*. Frame the grey-scale chart and adjust the iris to obtain a 100 IRE output signal from the white chip. Observe the color monitor and note whether the edges of each chip are equally enhanced, which confirms that the horizontal and vertical enhancement is balanced.

Note that as the chips become darker, the magnitude of the enhancement decreases in proportion, but remains balanced horizontally and vertically. At some level the threshold circuit will inhibit enhancement of the darker chips. It is important that this occur as close as possible to the black level, without permitting the enhancement of any noise that might be present in the blacks.

Next view a subject, which contains fine, subtle detail, such as hair, or a tennis ball. If the fine detail is clearly defined and the higher contrast edges are not overemphasized, the coring circuit is correctly adjusted and working well. If the depth of coring is excessive, the subtle detail will be lost and the picture will exhibit an imbalanced detail. Excessive coring should not be necessary, unless the camera tube has poor resolution, or for some other reason the camera is excessively noisy.

CHOOSING A CAMERA

Choosing a Studio Camera

In the studio, performance is of principle importance and small size and portability are rarely required. The broadcaster should, therefore, give his first consideration to the larger, tripod mounted cameras using 1¼, 1 inch, or ⅔ inch camera tubes.

Frequently, the studio is small and a wide angle zoom lens with a minimum object distance of three feet, or less, is advisable. However, a word of caution; if a studio audience is involved, it may well be necessary to position a camera 50 to 90 feet from the talent. Consider the case where the camera must be 50 feet away and a head and shoulders shot is required. This means a field of view of 2 feet, so the angle of view is:

$$\theta = 2 \text{ Tan}^{-1} 1/50 = 2.3 \text{ degrees}$$

which means, for example, that a 428 mm focal length is required in 1¼ inch format. If a 17mm lens is required for wide angle purposes (53.45 degrees field of view), then a 25.1 zoom lens is necessary.

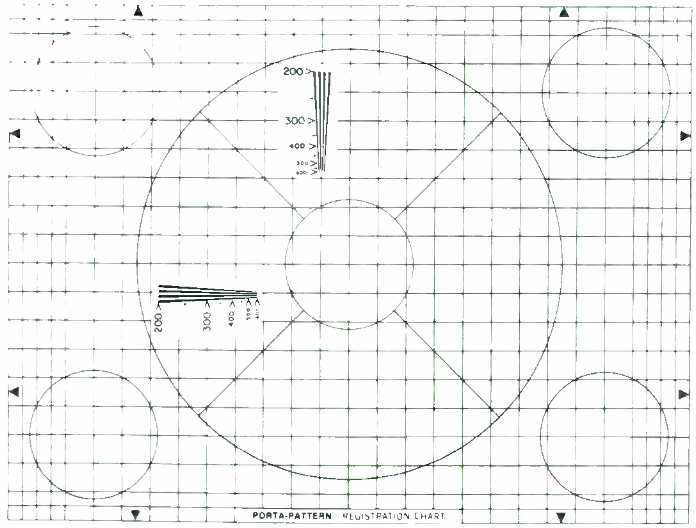


Fig. 34. Registration chart.

Choosing an ENG Camera

The ENG camera must be small, light weight and extremely rugged. The lens must have a reasonably wide angle, for example, 9 mm in ⅔ inch format, with a minimum object distance of 36 inches, or less, to permit use in confined spaces. Because lighting is often uncontrolled and frequently inadequate, high sensitivity is important. Probably, the broadcaster should look at a 14:1, F/1.7 zoom lens, with a minimum focal length of 9 mm in ⅔ inch format, which will give an MOD of about 32 inches. The usable sensitivity should be 5 foot candles, or less.

Choosing an EFP Camera

As previously discussed, the ideal EFP camera must combine the performance of a studio camera with the convenience of a portable camera. Therefore, the camera should be basically portable, but adaptable to the use of large lenses. The camera should be very stable and, if possible, have an automatic set-up feature. This is particularly desirable if the camera is to be used for field production, involving expensive talent and many extras. The producer will not be so happy if he has to wait 20 minutes for camera set-up before he can continue with take number 27!

The configuration will depend on the particular application. If the camera is to be frequently used in a portable application, a high quality ENG type lens is required. If true field production is the application and the camera can be tripod mounted, then a larger, higher quality EFP lens can be used. If the application is sports, then a lens that can zoom to a long focal length is a must, but, because the camera may also be used in close proximity to the commentator, a wider angle and close focussing distance may also be required. This is the ideal application for the new breed of lenses, which combine a wide angle with a large zoom range, for example 44:1.

The EFP camera may be used as a self-contained unit, or may be controlled from a Remote Control Unit (RCU). In certain cases, it may be necessary to control the camera from a great distance, such as sports, which favors the use of a triaxial interconnet.

Because base-band video is sent through the multi-conductor camera cable, the losses can become excessive at distances over 2,000 feet, with the result that there is a severe picture quality degradation after equalization. When long distances are involved, the single, triaxial cable is recommended, in which case the signals and controls are multiplexed onto a single conductor using an RF carrier. Both AM and FM systems are used. Performance is significantly improved and is acceptable to distances of 6,000 feet, or more.

Choosing A Sports Camera

As in the case of the ENG camera, the sports camera must be very rugged and stable and the camera crew will probably be happier if it is not so heavy. All things considered, this is an ideal application for the EFP camera.

However, the lens performance is a key factor. The coverage of many sports requires the use of a long telephoto lens, for example; golf, football and horse racing. Therefore, the sports camera should accept the larger zoom lenses with a zoom range 30:1, or more.

Usually, the camera is controlled from an OB vehicle, which in certain cases may be a long distance from the camera. For this reason, the use of a triaxial camera cable is popular, because it enables distances to 6,000 feet.

APPENDIX A

Please see Fig. 2. in Chapter 5.4.

APPENDIX B

Depth of Field and Focus Tracking

Depth of Field

When a lens is focussed on an object, there is a range of distance, which is also in focus, both in front and behind the object. This is known as the "depth of field" and depends on the angle of view and the relative aperture of the lens.

A wide angle lens (shorter focal length) has a greater depth of field than a telephoto lens. For a given lens, the depth of field is greater at f/4.0 than at f/1.6. However, this comparison does not necessarily hold for lenses with different formats. For example, f/4.0 in 1¼ " format is equivalent to approximately f/2.8 in 1 " format, f/2.0 in ⅔ " format, and f/1.5 in ½ " format. If these lenses are all adjusted to have the same angle of view, their depths of field will be equal.

For this reason, the smaller format cameras have no disadvantage compared to those with larger format, in either angle of view, or depth of field."

Focus Tracking

When the photosurfaces of the imaging devices are correctly located at the exit focal planes of the beamsplitter, the corresponding images will remain in focus as the lens focal length is varied. The optical system is then said to be "focus tracked". The procedure for focus tracking is as follows:

1. Adjust the illumination so that the lens aperture may be fully opened without the video signal clipping. A wide aperture is important, because the depth of field becomes small and the focus adjustment is, therefore, more sensitive.
2. Adjust the zoom to maximum focal length and focus the lens onto a detailed object or test pattern at least 20 feet away.
3. Adjust the lens to minimum focal length and adjust the position of the green channel imaging device to obtain sharp focus of the test pattern. Repeat 2) and 3) to ensure accurate green channel tracking.
4. Return the lens to minimum focal length and adjust the position of the red and blue channel imaging devices to obtain sharp focus of their corresponding images. The system is now focus tracked.

5.4

Video Signal Switching, Timing, and Distribution

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INTRODUCTION

The video signal contains a large amount of information. In addition to basic picture parameters such as brightness, hue, and saturation, the video signal includes horizontal, vertical, and color timing information. And recently, more and more ancillary information has been added, primarily in the vertical blanking interval: vertical interval test signals (VITS), vertical interval reference signals (VIRS), teletext, closed captions, time of day and time code information, source identification data, etc.

Technological advances in the last thirty years have made it possible to pack all of this information into substantially the same bandwidth originally allocated for the monochrome television picture alone. But along with the increased information density has come a need to maintain a very high level of performance along the entire signal path. Therefore, careful attention must be paid to the design of any modern video system if optimum results are to be achieved.

Whenever video signals are distributed, levels, frequency and phase response, and other transmission parameters must be held to extremely tight tolerances. Whenever video signals are switched, the switching system must be designed so that the integrity of the timing as well as the transmission parameters is observed. And whenever video signals are combined, whether by mixing, keying, or special effects generation, the relative timing relationships between the signals must be carefully maintained.

THE VIDEO SIGNAL

Fig. 1 is an oscillogram of a typical video signal. The peak to peak amplitude of the signal is 140 IRE units, or 1 volt. This is the standard level for signal distribution in all professional television facilities. The signal is comprised of +100 IRE units (714 mV) of picture and -40 IRE units (286 mV) of sync.

Even though video is an ac signal, proper operation of most equipment depends upon a fixed dc component. By convention, blanking (0 IRE) is assumed to be at ground (0 VDC). This reference is lost, of course, in portions of the system which are ac coupled, but when the signal is clamped or dc restored, standard practice dictates re-establishing blanking at ground.

The bandwidth of a "studio quality" NTSC video system is determined primarily by the state of the art and economic factors. And while it can be argued that the transmission system cuts off any information above 4 MHz, it is good engineering practice to strive for response which is flat to 8 to 10 MHz in the studio.

Timing integrity is ensured through the use of a master sync pulse generator. In nearly all installations, timing for all signals in the plant is derived from this single reference. In most cases,

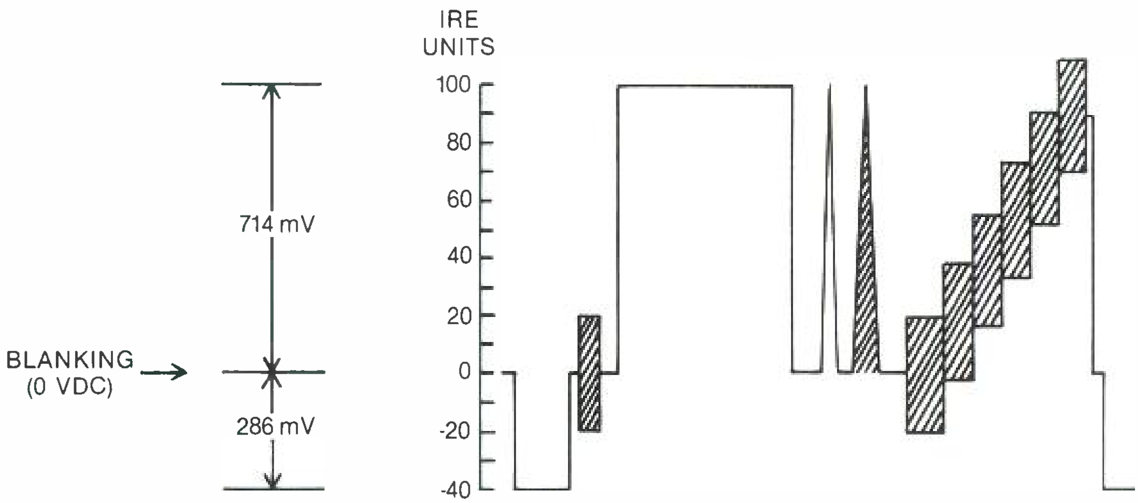

Fig. 1. Oscillogram of a typical video signal.

the master sync generator is driven from a high stability crystal oscillator. Occasionally, a precision rubidium or cesium "atomic" reference oscillator is used. Fig. 2 illustrates the relationships between the horizontal, vertical, and color timing signals.

INTERCONNECTION CONVENTIONS

All studio equipment is designed to be interconnected with coaxial cable having a nominal characteristic impedance of 75 ohms. In the simplest case, a point-to-point connection between two pieces of equipment, a continuous length of cable is driven from a 75 ohm source and terminated in a 75 ohm load. See Fig. 3.

When it is necessary to distribute the signal from a single source to more than one destination, two possible approaches exist. The first is entirely passive in nature: one end of the cable is driven from the 75 ohm source and, instead of terminating the far end in a 75 ohm load, a "loop-through" connection is made to the first piece of equipment. The loop-through is carried on to the next piece of equipment, and so on. A 75 ohm termination is placed on the loop-through connection on the last piece of equipment. This approach, shown in Fig. 4, will work well provided the loop-through inputs are properly designed and the cable lengths are kept short. The number of loop-throughs should also be kept as small as possible. If this is not done, frequency response errors and, in severe cases, signal reflections are likely.

Whenever possible, video signals should be distributed using an "active" device such as a distribution amplifier or routing switcher. This effectively results in the equipment being interconnected in the point-to-point manner described above.

Nearly all modern equipment uses BNC connectors for video input and output connections. Some older equipment still in service may use "UHF" (PL-259/SO-239) connectors. Occasionally, space limitations dictate the use of a subminiature connector such as a "BSM" series. However, the BNC is by far the most common connector in use.

A number of cable types are available which are suitable for high quality video interconnect. The most popular is Belden 8281, a double-shielded 75 ohm precision coaxial cable designed specifically for video use. 8281 provides a good balance between loss and physical size. Its double shielding helps reduce the likelihood of stray signal pickup. Where space limitations, increased flexibility, or other factors suggest the use of a smaller diameter cable than 8281, an "RG/59-type" cable is the usual choice.

POTENTIAL SYSTEM PROBLEMS

Most problems in television system design are a consequence of the fact that individual pieces of equipment must be interconnected by coaxial cable. All cable exhibits high-frequency losses which may result in the impairment of picture detail, especially on long runs. Stray signals, such as 60 Hz hum or noise may be coupled into cables. And since signals take a finite amount of time to travel through cables, there are time delays to be dealt with.

Fortunately, most of the problems are not new, and solutions have been developed over the years. For example, equalizers which compensate for

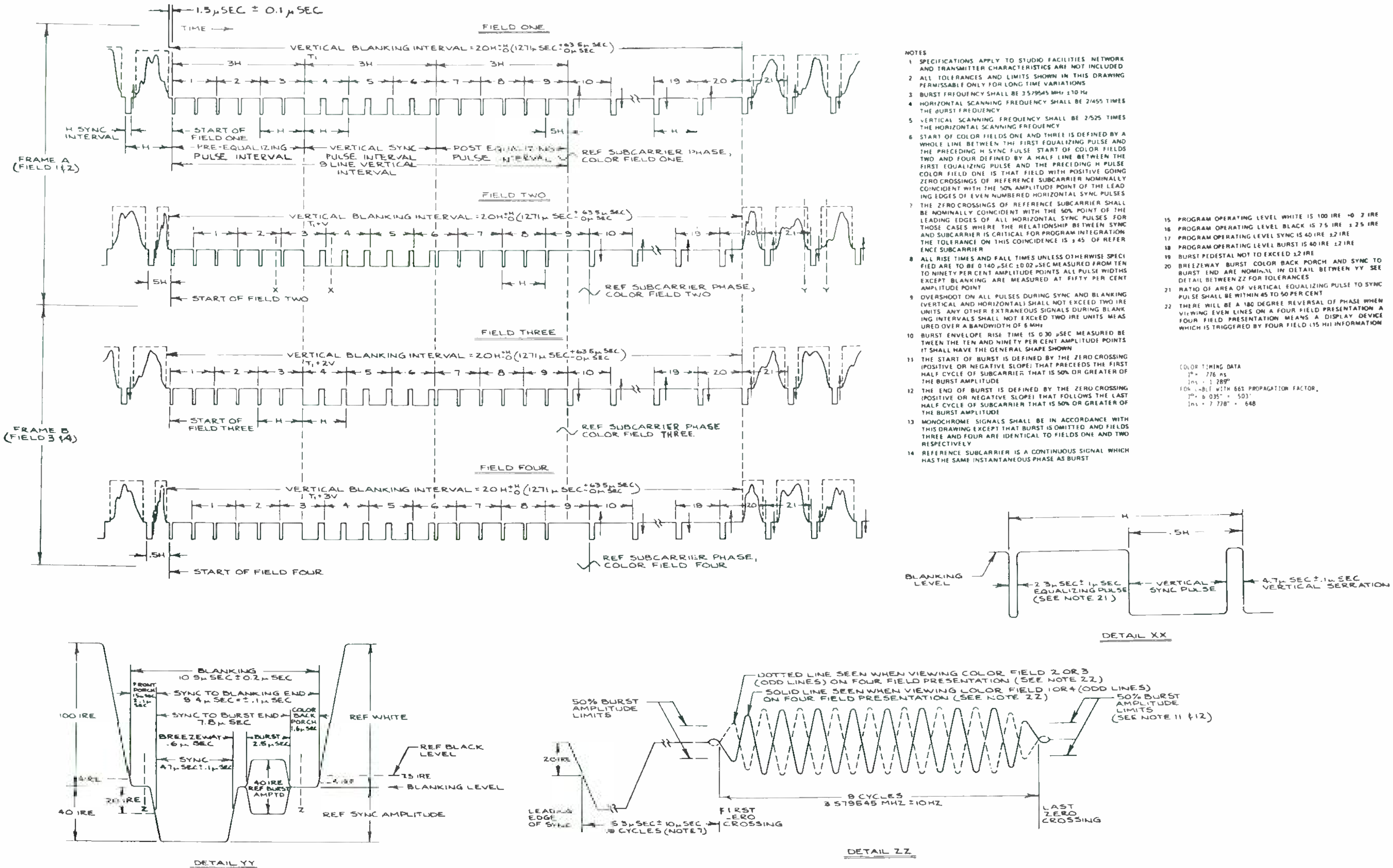

Fig. 2. NTSC timing parameters.
(Courtesy of The Grass Valley Group)

An updated version of the EIA Tentative standard RS 170 A.

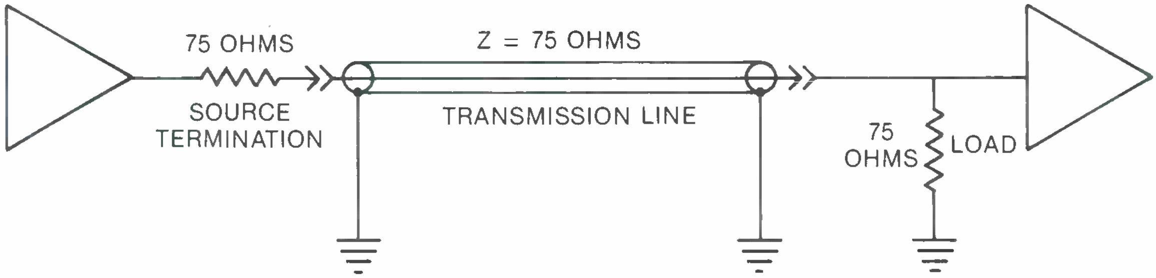

Fig. 3. Point-to-point video connection.

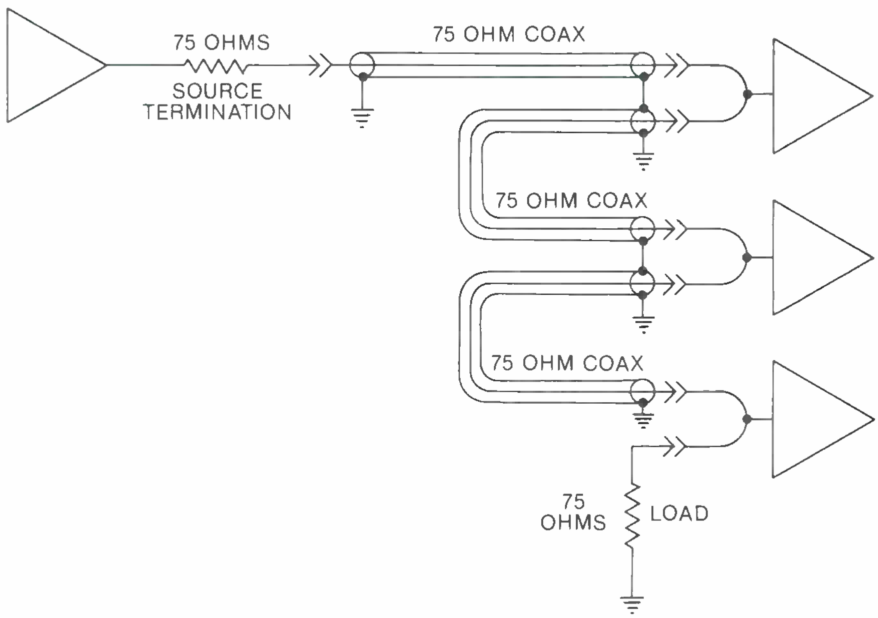

Fig. 4. Loop-through video connection.

high frequency cable losses can be found in a variety of modern video equipment. Quite often, a differential amplifier is used in the input stage to cancel the effects of common mode hum and noise; in some equipment, a clamp stage may be provided for hum suppression. Modern timing concepts have led to the development of a wealth of useful tools: pulse and subcarrier regenerators, slave sync generators, remote phasing sync generators, isophasing amplifiers, and frame synchronizers.

The tools exist to solve nearly any problem in video system design. The key is to anticipate the problems in new designs and to diagnose the problems as they develop in existing systems.

VIDEO SYSTEM DESIGN

Video system designs can be classified as falling into one of two broad categories: "hardwired" or "configurable." A hardwired system is one which is dedicated to a single function—a small ENG edit bay, for example. A configurable system is one which is designed to perform multiple functions, perhaps simultaneously—for example, a centralized VTR room which is used for on-air tape playback, commercial production, and dubbing.

Most systems were hardwired prior to the advent of video tape recorders because program production and presentation was done "live," in real time. VTR's made off-line production and program assembly possible. But the high cost of the early recorders made it necessary to design the system to ensure optimum equipment utilization. This was accomplished by making the system configurable.

The hardwired approach is fairly self-explanatory. Signals are routed point-to-point, using distribution amplifiers where a single source must feed more than one destination. Quite often, patch panels with "normalling" (through-connection) jacks are interspersed throughout the system, thus providing a certain degree of configurability.

A truly configurable system is generally built around a routing switcher. In the limit, all signal sources in the plant appear as inputs to the switcher and all signal destinations in the plant are driven from a routing switcher output bus. Such a high degree of flexibility is rarely required, however. As a result, most video systems are partially hardwired and partially switched.

Source timing is an important consideration in any video system. As a general rule, the more flexible the system is, the more difficult timing becomes. In a hardwired system, for example, timing signals can be hardwired along with the video signals. But in a routing switcher based system, configurations may be possible which create timing problems. The most common of these is the problem of studio switcher delay.

It is standard practice to designate a point in the system as a "zero timing point." The timing of all sources in the plant is adjusted so that they are in time at this point, which is usually a patch panel or routing switcher output. Timing will be correct so long as the path from the source to that point is fixed. But when the system is flexible enough to allow the source to go through one or more studio switchers enroute to the zero timing point, the source timing will be upset by the cumulative delay through the studio(s). If one attempts to treat that source as fully timed (by mixing or wiping between it and another source, for example), the problem will be immediately apparent.

One way around the studio delay problem is to backtime all sources feeding the studio by an amount equal to the studio delay. This can be done quite easily if each source is equipped with its own sync generator. It is important to note, however, that if the source must feed other points in the system in addition to the studio, it may not be properly timed for those points.

The best way around the studio delay problem is to put frame synchronizers on the outputs of all studios. In that way, timing is guaranteed, regardless of how complex the path becomes. Where such an approach is not economically feasible, it is essential to analyze and fully understand the limitations of the system.

SYSTEM DESIGN APPROACH

More often than not, video system design is approached in a haphazard rather than systematic way. This is often due to the pressures of attempting to provide the latest technology to the end user while keeping a station on the air. Even though it is a rare occasion that one is able to start from scratch, any system design project stands to benefit from up front planning.

The task begins with a survey of the facilities. What signal sources are there? What destinations must they feed? What is the physical layout of the plant? What is its size and where will the equipment be located? What allowances have to be made for future growth?

The next step is to analyze user needs. Each operating area should be examined in light of what signals are needed, how frequently they will be used, and what level of quality is necessary. It is important to be realistic when asking these questions. For example, it might be nice to have a seldom used test signal available "on demand" in a particular operating area; but is it really necessary?

Frequency of use will help to determine whether a signal should be hardwired, switched, or patchable. Frequently used signals should be hardwired in order to eliminate possible operational errors. Signals which are used occasionally can be switched, if necessary. Seldom used signals can be handled with manual patching.

The desired signal quality is an important factor. Does the signal have to be timed? What about equalization, clamping, or other forms of signal processing? Unnecessary expense can be avoided if the answers to these questions are carefully considered.

When the research is complete, the planning can begin. During the planning phase, the details of the design will begin to take shape. The physical realities of the system will come to light: equipment layout, interconnect, electrical power requirements, etc. Notes gathered during the research phase will give way to sketches, which will eventually give way to final drawings.

The importance of accurate system documentation cannot be overemphasized. Good system-level drawings will prove to be invaluable when troubleshooting or when making changes. And the documentation will be much easier to follow if wire and rack numbers are used.

After the final "as built" drawings are completed, it is a good idea to produce a simplified drawing of the signal flow through each operational area. These can be posted, perhaps along with information on how to deal with "emergency" situations.

VIDEO DISTRIBUTION AMPLIFIERS

The distribution amplifier, or "DA," is the most elementary building block in a video system. Its primary function is to allow a single 75 ohm video source to feed multiple 75 ohm loads with no apparent loss. Quite often, other functions such as cable equalization are included in the DA's circuitry.

Even though DA functions and designs vary, it is quite easy to construct a generalized model. As shown in Fig. 5, the typical DA consists of an input amplifier followed by a "conditioning" stage (gain, equalization, delay, etc.), followed by an output amplifier.

The input amplifier is generally designed with a rather high input impedance, on the order of several tens of kilohms. This allows the driving signal to be looped through the input without being loaded down. Quite often, a complex network is used at the input connection to cancel out any reactive components which might affect the loop-through. The input stage may be either single ended or differential, the latter being more common on high quality DA's.

A differential input stage can be very useful in combating one of the most serious problems encountered in unbalanced signal transmission: common mode hum and noise. In larger facilities, common mode hum due to ground loops may run as high as several volts. Even in small plants with excellent ground systems it is possible to have several tens of millivolts of difference in ground potential between different pieces of equipment. A differential amplifier will cancel any common mode signals while amplifying the (desired) differential mode signal. Common mode rejection ratios (CMRR) of 50 to 60 dB are typical of most modern differential input DA's.

The conditioning stage may be a wideband gain stage, an active equalizer stage, or a delay amplifier, depending upon what additional function the DA performs. If distribution is the only function, the conditioning stage will be omitted.

The output stage is an amplifier with very low output impedance. It is followed by source termination resistors which ensure that the cable "sees" 75 ohms at the sending end as well as the load end. Output to output isolation is guaranteed by the low impedance at the amplifier output.

Cable equalization is often required in video systems. This is due to the high frequency losses in coaxial cable. Belden 8281 has approximately 0.8 dB of loss per 100 feet at 10 MHz. Since runs of 100 feet or more are not uncommon in most facilities, achieving "flat" system response makes equalization of these losses necessary. The approach taken by most manufacturers is to construct an equalizer network whose response very closely approximates the inverse of the cable loss curve. When this equalization is applied to the cable/DA system, the result is flat overall response.

Cable equalizers must be designed for a specific type and length of cable. When a variable equalization control is provided, it usually serves to adjust the amount of equalization applied. Because cable loss is a complex function, the equalization curve is only "right" for one cable length. Slight response errors will be noted at other length settings.

Cable losses should always be equalized at the load end of the cable. The reason for this is simple: If a "flat" signal is applied to the input of the amplifier, the high frequency boost of the equalizer stage will be added to the signal and the output stage will have to track these larger than normal amplitude high frequencies. In cases where the amount of equalization is more than a few dB, the output stage will be unable to deliver the necessary current into its low impedance load. If, however, the DA is located at the load end of the cable, the cable losses have attenuated the high frequencies at the input of the amplifier. The high frequency boost of the equalizer stage restores the flat response characteristic to the signal, and the output amplifier does not have to deal with larger than normal swings at high frequencies.

Another function which is often required in video systems is clamping. Clamping is used primarily to reduce low frequency hum and tilt but may also be used to restore the dc component of a video signal which has passed through one

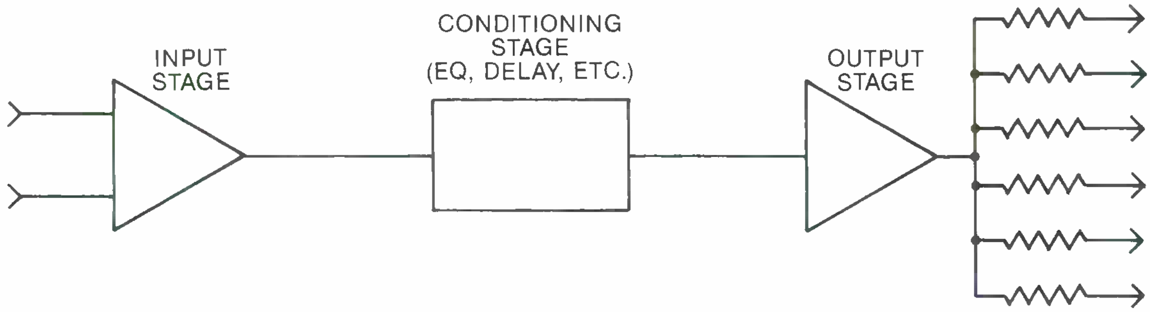

Fig. 5. Simplified block diagram of a video DA.

or more ac coupled devices. Clamps operate on a line-by-line basis and, as such, are fast acting. They can be triggered by noise impulses and should therefore not be used on noisy signals.

Dc restoration may be used in lieu of clamping to reduce the dc signal bounce due to multiple ac couplings. Because of the slow time constants employed in dc restorers, they are not affected by noise; but by the same token, they are not useful in eliminating hum.

Delay DA's are often used in place of coaxial cable or other passive delay elements. Delays of up to a microsecond or more are possible on modules no larger than a conventional DA. Many delay DA's include circuits to compensate for delay line response errors; they produce better results than a passive delay line and DA in combination. Some delay DA's also include other functions such as cable equalization.

PULSE AND SUBCARRIER DISTRIBUTION AMPLIFIERS

Despite the increasing use of color black as a locking signal, separate pulses and subcarrier are used in many systems as timing references. Pulses (sync, blanking, H drive, V drive) are generally distributed as 2 to 4 volt negative-going signals. Subcarrier is generally distributed as a 1 to 2 volt peak-to-peak signal.

Most video DA's will not accomodate signals larger than 2 volts peak-to-peak. So, while most video DA's would be suitable for subcarrier distribution, the typical video DA will not handle pulses.

DA's designed specifically for pulse distribution are optimized for large negative signal swings. They will not function as video amplifiers.

Pulse DA's fall into one of two general categories: regenerative and linear. Regenerative DA's include circuitry for pulse regeneration; typically edge detectors driving one-shots. Timing of the one-shots is usually adjustable, allowing the relative delay of the amplifier to be varied. Delays of up to several microseconds are common. Due to the inherent delay of the pulse regeneration circuitry, these amplifiers always exhibit some value of minimum delay.

Regenerative subcarrier DA's are also available. These amplifiers generally feature adjustable circuits to shift the phase of the subcarrier. The adjustment range is usually slightly more than 360 degrees.

When pulses and subcarrier are distributed through regenerative DA's, there is a good chance that SC/H phase will not be maintained throughout the system. This is because it is possible to adjust sync and subcarrier timing independently. For this reason, linear (non-regenerative) pulse and subcarrier DA's should be used in SC/H phased installations. Specially designed linear pulse DA's are available, and most high quality video DA's can be used as linear subcarrier DA's.

DISTRIBUTION AMPLIFIER SELECTION

As a signal traverses a typical path through a typical system, it may pass through ten or more DA's. Therefore, the performance of a single amplifier must be, for many parameters, an order of magnitude better than the system specification. Table 1 lists the performance specifications of a representative unit, the Grass Valley Group 3402V-A. This amplifier is pictured in Fig. 6.

Selecting the right DA for a particular application involves determining the desired level of performance and the features and functions required. Since most manufacturers offer a range of models which all fit into the same mounting tray, it is

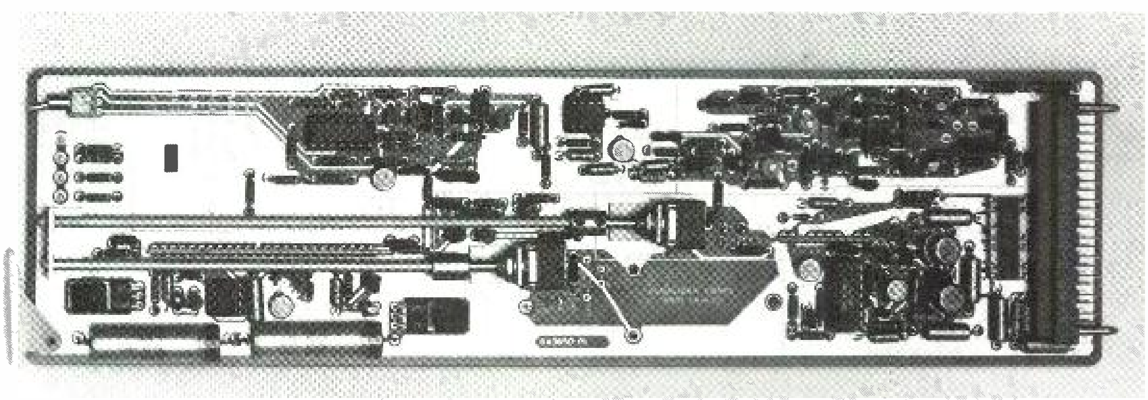

Fig. 6. A modern video distribution amplifier, the Grass Valley Group Model 3402. (Photo Courtesy the Grass Valley Group, Inc.)

wise to look at the sum total of all requirements and then choose the manufacturer who can best accommodate those requirements.

DA's fall into the "install it and forget it" category; low maintenance and long service are important factors. When selecting distribution amplifiers, consider the quality of the components which the manufacturer has used; will they hold up for ten or fifteen years? Is the design a conservative one, or are devices being run at their limits? Thermal stress will not only lead to equipment failure, it will cause drift as well.

VIDEO SIGNAL SWITCHING

Video signals can be switched by a variety of methods. Simple monitor selectors may be nothing more than passive mechanical switches. In some cases, electromechanical devices such as relays are employed. Most modern switching systems employ solid state switching devices.

Fig. 7 is a simplified block diagram of a video switching system. Input amplifiers buffer the input signals and provide drive to the switching elements (crosspoints). The crosspoints are bussed in horizontal rows (output buses) to an output amplifier. If the switcher has more than one output bus, the buffered input signals are bussed in vertical columns (input buses) which run across the output rows.

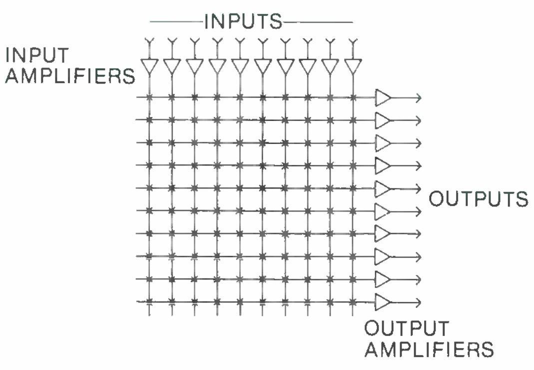

Fig. 7. Simplified block diagram of a video switching system.

Not shown in the block diagram is the crosspoint control logic. In single bus switchers with ten inputs or less, control is generally accomplished using a single wire per crosspoint. In single bus switchers with more than ten inputs, an encoding scheme such as BCD is generally used to reduce the number of wires required. Larger multiple bus switchers generally use some sort of serial control scheme.

Single Bus Switchers

Single bus switchers run the gamut from simple to elegant. For noncritical applications such as picture monitor input selection, a mechanical switcher is probably adequate. Most mechanical designs are entirely passive; that is, they have no input or output amplifiers. As such, they have terminating inputs and require dedicated (non loop-through) input feeds.

A slightly more sophisticated design adds an output amplifier only. Switching is still done using mechanical switches, but the bus is made high impedance to allow looping inputs. This approach makes it possible to "stack" units to provide an economical multiple input/multiple output configuration. In terms of performance, this type of switcher is no better than the passive type. If the input loop-throughs are not compensated, its performance may actually be worse.

Any single bus switcher which is in a program path should be designed more along the lines of Fig. 7. In other words, it should have input buffer amplifiers, solid state crosspoints, and an output amplifier. If it is being used for "live" switching, the buffer amplifiers should clamp or dc restore the input signals to provide bounce-free switching. If the unit is used in an actual program loop, the design should permit servicing with minimum disruption; that is, all active circuitry should be mounted on plug-in printed circuit boards.

Single bus switchers can be combined to make multiple bus switchers, but in most cases the result will cost more and lack the flexibility and performance of a true multiple bus design.

Multiple Bus Switchers

There are two basic architectures for multiple bus switchers. An output oriented switcher is one with all buses physically separated in the switcher electronics. A matrix switcher is one in which the circuitry for several output buses is packaged together.

Output oriented switchers are generally larger and more expensive than comparable matrix switchers. The reason for this is that there is more duplication of circuitry and more interconnect required. The chief advantage of an output oriented switcher over a matrix switcher is that circuit cards can be removed from the output oriented switcher without disturbing multiple signal paths. This is an important consideration in large facilities such as network switching centers.

Matrix architected switchers are smaller and less expensive than comparable output oriented switchers. They offer several other advantages as well: reduced power consumption, slightly higher reliability due to fewer mechanical interconnec-

Fig. 8. An example of a modern single-bus routing switcher, the Grass Valley Group TEN-X. (Photo Courtesy the Grass Valley Group, Inc.)

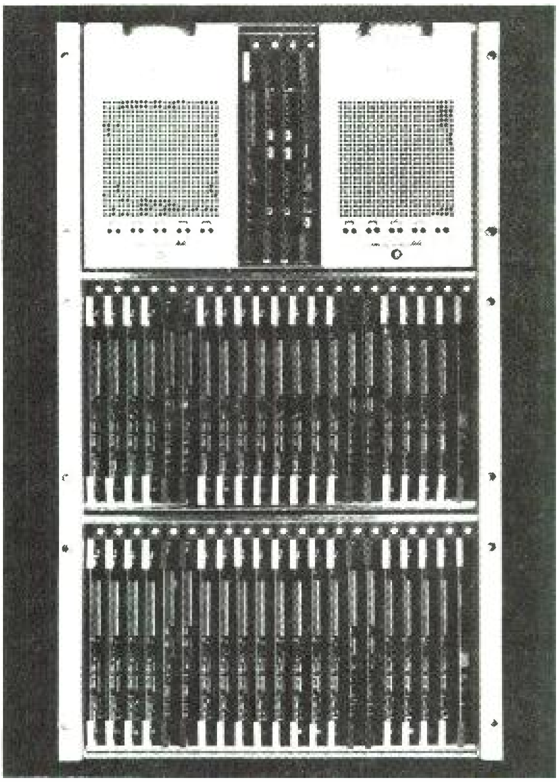

Fig. 9. A multiple-bus routing switcher system, The Grass Valley Group model HX-64. (Photo Courtesy the Grass Valley Group, Inc.)

tions, and shorter signal path. Matrix switchers are generally designed around a relatively small square or rectangular building block, making input and output expansion in bite-sized increments a simple matter.

Input signal distribution in both types of switchers can be handled using either looping techniques or distribution amplifiers. The cost objectives of most matrix designs suggests the use of looping while the large physical size of the output oriented switcher makes the distribution amplifier approach almost a necessity.

As is the case with single bus switchers, if "live" switches are made on a multiple bus switcher, the input signals must be clamped or dc restored to ensure transient-free switches. This signal conditioning may be done external to the switching system, but better overall performance will result if it is done in the switching system. The reason for this is that a common precision dc reference can be used for all inputs if the clamping or dc restoration is done inside the system.

Switch timing is also important if glitches are to be avoided. Most multiple bus switchers have a sync input which is used as a reference for a master trigger pulse generator. The trigger is set to occur during the vertical interval, thus ensuring glitch free switching between synchronous sources.

Timing "scatter" is an important parameter for any switching system which will be used in a studio environment. It is the measure of the delay difference between paths in the switcher. In a system which uses looping techniques to distribute the input signals, there will be a fixed amount of delay between output bus groups. But on a given bus, there should be very little path length difference from input to input; less than plus or minus one degree of subcarrier is a typical figure. In a system which uses DA's to distribute the input signals, it should be possible to achieve the same timing accuracy from any input to any output.

With respect to the use of routing switchers in video systems, two basic schools of thought prevail: the master grid approach, and the cellular approach. With the master grid approach, one large switching system is used with all switched sources in the plant appearing as inputs. This provides maximum flexibility, but at considerable expense. All sources are not needed at all destinations, so many crosspoints in the system are "wasted."

In the cellular approach, the plant is subdivided into functional areas (studios, master control, vtr/telecine pool, news editing, etc.) and a small switching system placed in each area. Each system has only those area sources it requires plus a few "master" sources common to the other areas (color black, bars, etc.) To accommodate occasional special setups, tie lines may be run between the systems.

The cellular approach can be considerably more cost effective than the master grid approach. But the incidence of special setups must be carefully considered. Since multiple switching operations may be required in the case of specials, the chance for error increases significantly. This must be weighed against the additional cost of the larger matrix required for a master grid system.

Switching System Control

Switching system control has advanced considerably over the past few years. Most multiple bus switchers have microprocessor-based control systems. Control panels communicate with the switcher over twisted pairs or coax. Sources may be selected using their familiar names (eg "BLACK," "BARS," or "VTR 5") instead of input numbers. Simultaneous selections, or "salvo" switches, can be easily made.

A recent development in control system capability has been made possible through the availability of a semiconductor device known as an EEPROM (Electrically Erasable Programmable Read Only Memory). This device can be programmed, erased, and reprogrammed just like an ordinary PROM. But, unlike a PROM, the EEPROM does not require a special programmer; more importantly, it can be erased electrically, in circuit.

The system pictured in Fig. 9, a Grass Valley Group HORIZON series, uses EEPROMS in its control system for non-volatile storage of installation dependent information. This information is loaded into the system using an ordinary ASCII data terminal.

The HORIZON system is a representative state of the art routing switcher. The unit pictured in Fig. 9 is a 64 input by 64 output video matrix complete with dual power supplies and dual control cards. The unit occupies 18RU (31.5″) of vertical rack space. It is a matrix architected system. Performance specifications are listed in Table 2.

VIDEO SIGNAL TIMING

Timing requirements stem from the fact that a video signal is a serial stream of analog data which is used to electronically reconstruct a picture. Horizontal timing information is used to trace out each line of the picture on the face of a display tube in exact synchronism with the camera tube beam which scanned the original scene. Vertical timing information is used to signal the end of each "still" picture in the frame-by-frame reconstruction process. Color timing information in the form of a brief burst of subcarrier each line is used to ensure that the color at each point on the screen is accurately reproduced.

Timing in the television studio is the responsibility of a ubiquitous black box called a sync generator. In the early days, a monochrome sync generator filled two six foot equipment racks. Today, color units which are several orders of magnitude more accurate and stable can be packaged on a single printed circuit card.

The size and complexity of the early generators prompted the "master" generator approach to plant timing. Since it would have been impractical to include a sync generator with each source, a set of signals was defined which would provide the information necessary to generate and process a television picture. The responsibility for generating these signals fell to the "master" sync generator. These signals were then distributed to each piece of equipment requiring them. They included sync and blanking (used mainly in video circuits) and horizontal and vertical drives (used to drive the sweep circuits in cameras).

The emergence of color meant that the 3.58 MHz subcarrier had to be generated by the same master generator and distributed throughout the plant. In addition, another pulse called burst flag was needed to gate the color burst. Six separate signals, just to keep everything locked to the same time reference.

The advent of solid state devices led to a dramatic reduction in the size of the master sync generator. At the same time, integrated circuit logic made it possible to easily derive horizontal and vertical drives and burst flag from sync. Before long, timing input requirements dwindled to sync, blanking, and subcarrier on most equipment.

The capabilities of Large Scale Integrated circuit (LSI) technology has made it possible to reduce most of the circuitry required for pulse generation to a single chip. For this reason, most modern equipment has its own sync generator built in. In addition to making standalone operation possible, this reduces the input timing reference requirements to a single wire signal such as color black.

Fig. 10 is a simplified functional block diagram of a sync generator. A precision oscillator is used to drive a set of complex logic which generates all of the required timing signals. Buffer amplifiers convert the logic levels to the required 4 volt negative-going pulses and 2 volt peak to peak subcarrier.

The FCC requires that subcarrier frequency be maintained within 5 Hz of 3.579545 MHz. Therefore, an ovenized or temperature compensated crystal oscillator is generally employed.

Most modern sync generators utilize a single LSI integrated circuit as the "heart" of the pulse generation logic. Numerous off the shelf and custom parts exist which have 90% or more of the logic required. External parts are often needed to provide additional features such as variable pulse width.

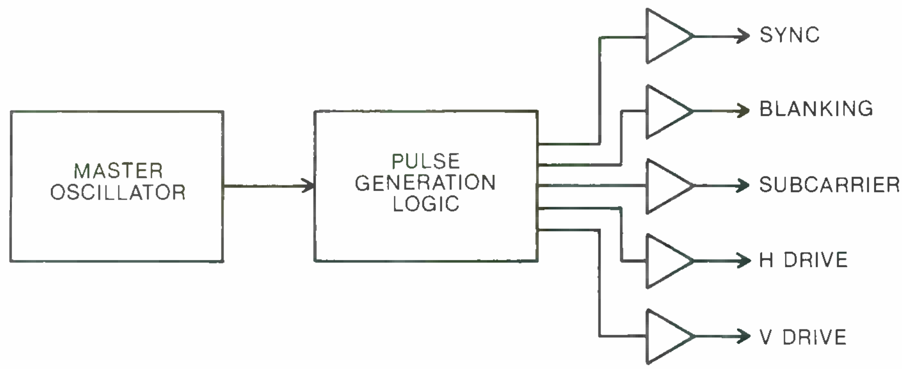

Fig. 10. Simplified block diagram of a basic sync generator.

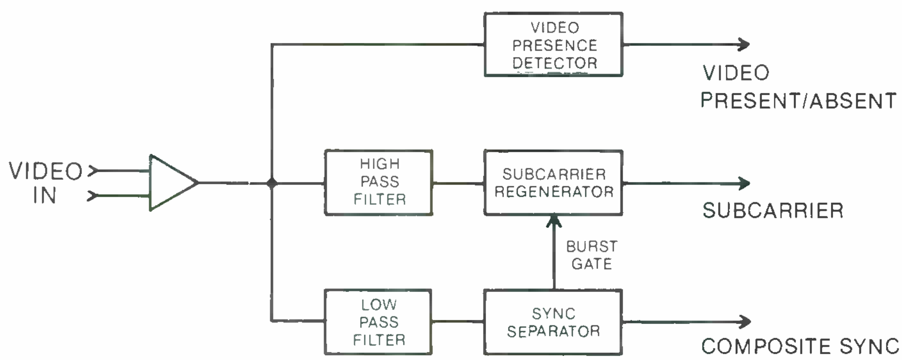

Fig. 11. Simplified block diagram of basic sync generator genlock circuits.

All IC logic families produce pulses with very fast risetimes. So, in addition to providing the proper impedance and voltage levels, the output buffer stages often include shaping or filtering circuits to control the risetimes of the output pulses. Controlled risetimes help to ensure the reliable and predictable operation of pulse detection and regeneration circuits in the equipment being driven by the sync generator.

GENLOCK

A considerable amount of additional circuitry is required to enable a sync generator to lock to a composite video signal such as color black. Fig. 11 shows the basic functional blocks: an input amplifier, subcarrier regenerator, sync separator, and video presence detector.

The input locking signal may contain common mode hum or noise which could upset the operation of the circuits which follow. For this reason, a differential input amplifier is generally used, followed by a clamp.

Since the locking signal may not always be present, the genlock circuitry includes a video presence detector. The output of this detector is a logic level signal which tells the sync generator whether to select the "free run" or "locked" mode of operation.

The buffered and clamped input signal is low pass filtered and then fed to a sync separator which outputs sync and burst gate. The burst gate, along with high pass filtered video, is used to regenerate the 3.58 MHz subcarrier impressed on the input locking signal. The separated sync and subcarrier drive phase locked loops in the generator. Adjustments on these locking circuits permit the generator phase to be varied with respect to the phase of the input signal.

The generator may be genlocked to the input signal in one of two basic ways: by independent locking of sync and subcarrier or by dependent locking of sync and subcarrier. With the independent method, the generator's sync and subcarrier locking circuits are continuously referenced to the input signal's sync and subcarrier. With the dependent method, lock is initiated in the same manner as the independent method. But once full genlock is achieved, the input signal's subcarrier is used as a reference for both loops.

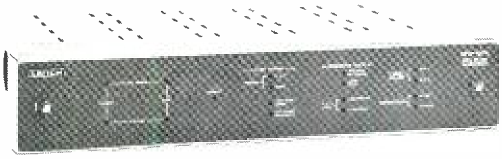

Fig. 12. A common sync generator feature or option is the capability to lock the generator to an external signal so that any equipment using the generator's output will also be in sync with the external signal. This model uses the chroma burst of the external signal to lock a 14.3 MHz oscillator from which it then derives its chroma and pulse outputs. (Photo courtesy of Leitch Video Limited.)

An independent locking generator will track the SC/H (Subcarrier to Horizontal) phase of the input signal. It will also lock to a non-standard color signal (one in which the subcarrier and sync are not phase locked at all).

A dependent locking generator will maintain correctly SC/H phased outputs regardless of the SC/H phase of the input signal. It will also lock more reliably to signals with occasional noise hits or with sync-only timebase error.

Genlocking the master generator to a non-synchronous source such as network has given way to the use of frame synchronizers. However, genlocking is still used in "master/slave" sync systems where individual sources or groups of sources are driven off "slave" generators which are locked to the plant's "master" generator. And, as was previously mentioned, many of today's cameras and vtr's contain their own (genlockable) sync generators.

As a result, it is not uncommon to find situations where three or more sync generators are cascaded. This makes it very important that the first generator in the line (usually the plant's master generator) has very good timebase stability. It is equally important that all slave generators exhibit very low "jitter." A figure of 5ns or less is typical for timebase stability, while 2ns or less is a reasonable jitter spec.

Where genlocking to an outside source such as network is done on a regular basis, a feature called "protected genlock" is highly desirable. With protected genlock, the local generator is locked to the outside source using the independent locking method. Any horizontal timing jumps on the locking signal will not disturb the generator; it will remain frequency locked to the locking signal's subcarrier. If the input signal disappears or becomes monochrome, the protected genlock logic will instruct the generator to make a smooth switch to the free run mode.

TIMING CONCEPTS IN THE MODERN STUDIO

In the early days of color, the system designer's attention became focused on the importance of color timing. While timing errors as large as several hundred nanoseconds could be tolerated in monochrome systems, color systems demanded accuracy at least two orders of magnitude better. The vectorscope took its place alongside the waveform monitor as an indispensable measurement tool.

As monochrome systems were converted to color, these new timing tolerances had to be addressed. The required degree of precision was not difficult to attain if reasonable care was exercised.

Unfortunately, the importance of the interrelationships between sync and subcarrier, especially SC/H (Subcarrier to Horizontal) phase, were not fully appreciated at the time. It wasn't until color video tape recording and editing came along that these issues came to light.

When the NTSC color system was devised, the frequency of the color subcarrier was chosen to be an odd multiple of half the line scanning frequency. This was done so that the subcarrier peaks would occur in alternating positions on adjacent scan lines, thus minimizing the visibility of the subcarrier on monochrome receivers.

The multiplier which was finally chosen was 455/2, or 227.5; this is also the number of cycles of subcarrier per scan line. Since there are 525 lines per frame, there are 227.5 times 525 or 119437.5 cycles of subcarrier per frame. The extra half-cycle of subcarrier means that it takes two frames (or four fields) for the color sequence to repeat.

When a video tape is edited, the machine must ensure that the continuity of this sequence is preserved. If it is not, there will be a noticeable horizontal shift in the picture at the edit point. Many machines incorporate "color framers" to prevent this from occurring. They operate by matching the color framing of the signal recorded on the tape with that of the house reference.

Reliable operation of these color framers is dependent upon the maintenance of consistent SC/H phase throughout the plant. Furthermore, when tapes which have been recorded in another facility are edited, the SC/H phase of the off tape signal must match the local plant's SC/H phase. For this reason, an industry-wide standard has been established which defines correct (or "zero") SC/H phase as the coincidence of the 50% point on the leading edge of horizontal sync and the zero-crossing of subcarrier. (See Fig. 2.)

Fig. 13 shows two methods of displaying SC/H phase. A third method, used with the GVG 3258, provides a digital readout of SC/H phase in degrees or nanoseconds plus time base error measurement and comparative SC/H phase between two sources.

Maintaining SC/H phase depends upon careful selection of equipment and periodic measurements of actual performance. Sync generators with low SC/H phase drift are key elements in any SC/H phased plant. It is equally important to avoid system design techniques which might lead to SC/H phase problems (such as the use of regenerative pulse or subcarrier distribution amplifiers).

SC/H phase should be checked as routinely as video levels or color phase.

Timing Tools

Numerous approaches to plant timing exist, but they all fall into one of two general categories: adjusting the relative phase of the source timing reference, or delaying the source video. In general, it is always better to move the timing signal around. This is because even the best video delay systems introduce a certain amount of signal distortion. Obviously, video delay is not a viable solution where a source must be advanced in time with respect to other sources.

Video delay lines are often used when a single source must feed different timing points. This is quite often the case in a system where a source must appear as a direct input to a studio production switcher, but the same source may be selected on a routing switcher which, in turn, feeds the production switcher. This is illustrated in Fig. 13.

In general, it is better to use a delay DA than a passive delay line/video DA combination. This is because a properly designed delay DA will include circuits which compensate for response errors in the delay elements. Where passive delays are used, they should be of the highest possible quality and the shortest electrical length required to get the job done.

Other devices which can be useful in plant timing are regenerative pulse and subcarrier DA's. They should be applied with caution, however, since they can create system SC/H phasing problems.

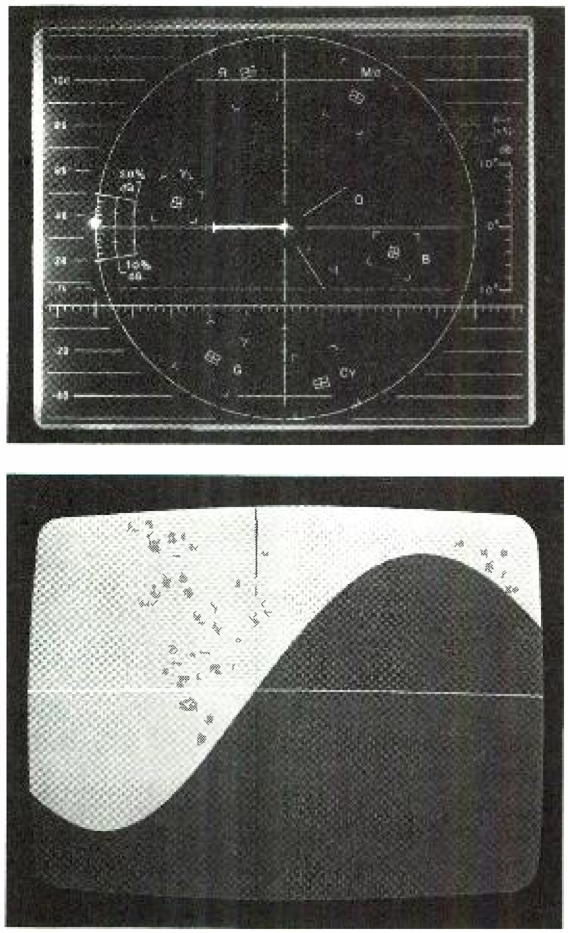

Fig. 13. Two methods of displaying SC/H phase relationships. Top: The Tektronix 1750 waveform and vector monitor shows the relation between the burst phase vector and the "sync dot" as either an absolute value or relative to an external reference. Phase may be read out from the graticule or by adjusting and reading calibrated controls. Bottom: The Lenco Videoscop generates an output to a picture monitor showing the relationship of the leading edge sync in line 1, field 1 (vertical line), the 50% point (horizontal line), and one cycle of the subcarrier (sine wave). (Photographs courtesy of Tektronix, Inc. and Lenco, Inc.)

GUIDE TO PLANT TIMING

At first glance, timing a television plant may appear a formidable task. Like anything else, though, the task can be whittled down to size through a systematic method comprised of analysis, synthesis, and execution.

Analysis entails the careful examination of the timing requirements in each individual operating area of the plant. This information, along with a clear understanding of the capabilities of the equipment, can be used to synthesize a plan for overall plant timing.

The demands placed on most plants are constantly changing and ever-increasing; today's special setup may be S.O.P. tomorrow. It is important to realize, therefore, that the "plan" which has been put together so carefully may change. If this happens, it may be necessary to add or replace equipment or perhaps even alter the system design.

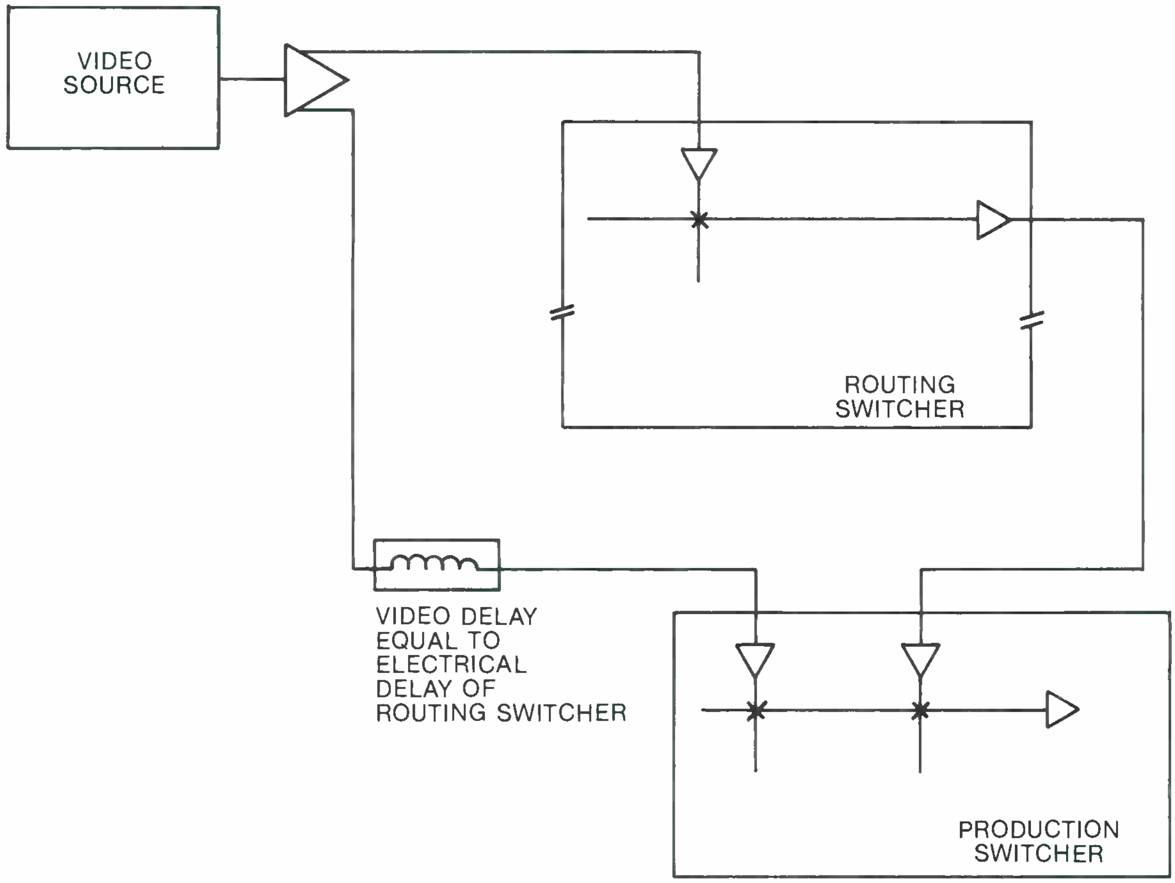

Fig. 14. Timing a single video source to two different timing points.

Studio Timing

The studio (or edit suite) is an excellent place to begin a discussion on system timing, since it is the place where proper source-to-source timing relationships is most important. This is due to the fact that the studio is the primary area of the plant where multiple sources are combined. If visible picture jumps and/or color shifts are to be avoided when making mix, wipe, or key transitions, all studio sources must be carefully timed.

Timing the inputs to a studio switcher is a very simple, straightforward process:

1. Identify those sources which do not have timing adjustments or whose timing can only be adjusted over a very limited range. These will more than likely include the color black and background generators in the production switcher itself and MAY also include test signal generators, character generators, and older studio cameras or film chain cameras.
2. Using delay DA's or delay lines, match the timing of all of the sources identified in item 1 above to the "latest" in the group of fixed sources. Designate one of these sources as the reference source for setting the timing of all of the adjustable sources. Note that if there is only one fixed source, it should be used as the reference source.
3. Adjust the timing of each of the adjustable sources to match the timing of the reference source.

All timing adjustments should be made by selecting each source, in turn, on the production switcher. An externally referenced waveform monitor, vectorscope, and (optionally) SC/H phase meter connected to the output of the production switcher provide the necessary monitoring capability. Timebase and/or gain magnification should be used on the test instruments to ensure the accuracy of timing adjustments.

The output SC/H phase of most equipment will change when timing adjustments are made. If the plant is to be SC/H phased, it is imperative that this parameter be checked whenever source timing is adjusted. On equipment which uses separate pulses and subcarrier for timing, care should be taken to maintain consistent phase relationship between the input sync and subcarrier once the output SC/H phase of the source is established.

If the plant is equipped with multiple studios, the procedures outlined above should be repeated in each studio. On those sources which feed more than one studio, extreme care should be taken to ensure that cable lengths are made equal. If this is not done, it will be impossible to achieve proper timing in all locations.

Timing in Master Control

For all intents and purposes, master control is just another studio and can therefore be timed using the above procedure.

One timing problem which is fairly common in master control is that of studio delay. It generally shows up as a slight timing disturbance when master control switches from a given program source to that same program source through a studio switcher (a common occurence going into or coming out of a live program such as news). An easy solution to the problem is to re-synchronize the output of each studio (see "CASCADING STUDIOS," below).

Timing Through an Assignment Switcher

If routing switcher buses are used as input preselectors for a studio switcher, the propagation delay through the routing switcher must be taken into account when timing the plant. This is usually accomplished by setting up both direct and delayed paths for any sources which feed a studio switcher both directly and through a routing switcher (see Fig. 14).

In plants where the routing switcher is used for source assignment, it is a good idea to designate the output of the routing switcher as the zero timing point for the plant. If this is done, a bus on the switcher can be designated the plant "quality control point"—one location (with one set of accurately calibrated test instruments) where the signal levels and timing relationships of all sources in the plant can be observed and compared. The generalized timing procedure for studio switchers can be used to time sources into the plant routing switcher.

Cascading Studios

Occasionally, it may be necessary in some plants to cascade studios. In order to preserve overall system timing, the delay through each studio must be taken into account. One approach to keeping everything in time is to time all sources into the first studio "early" by an amount equal to the studio delay. However, if more than two studios are involved and timing into a routing switcher and/or master control are a factor, altering source timing may not be practical.

A far better approach is to re-synchronize the output of each studio using a frame synchronizer. The timing of each studio can then be set to match that of all other sources in the plant.

A word of caution is in order, however: since nearly a frame of video delay is required to re-synchronize a source which is only a hundred nanoseconds or so "late," the video from the studio when passed through a frame synchronizer will be out of sync with the audio by approximately 33ms. While this amount of time difference might not easily be perceived by most individuals, the cumulative effects of several passes probably would be. Therefore, it is a good idea to insert an audio delay line in the program output from the studio.

TABLE 1. Typical video DA performance specifications.

Frequency Response	±0.05 dB to 5 MHz, -0.3 dB at 10 MHz
Differential Phase	<0.1 degree
Differential Gain	<0.15 %
T-Pulse to Bar	<1.0 %
Tilt	<0.5 %
Chrominance/Luminance Delay	<10 ns
Hum	>60 dB below 1 volt p-p
Noise	>70 dB rms below 1 volt p-p
Common Mode Rejection	>60 dB
Input Return Loss	>40 dB to 5 MHz
Output Return Loss	>40 dB to 5 MHz
Output Isolation	>40 dB to 5 MHz

TABLE 2. Typical video switching system performance specifications.

Frequency Response	+0.1 dB to 5.5 MHz, -0.5 dB at 8 MHz
Differential Phase	<0.1 degree
Differential Gain	<0.1 %
2T-Pulse to Bar	<0.25 %
Tilt	<0.5 %
Crosstalk	−60 dB at 5 MHz, worst case
Noise	>75 dB rms below 1 volt p-p
Input Return Loss	>40 dB to 5 MHz
Output Return Loss	>40 dB to 5 MHz
Switching Transients	<30 mV
Timing Scatter	± 1 degree max. input to input on any one output bus

5.5

Video Tape Recording

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HISTORY

The introduction of the first practical video tape recorder at the 1956 National Association of Television and Radio Broadcasters Convention substantially changed the operational aspects of television broadcasting and eventually made substantial changes in how material is produced for broadcasting.

The first recorders, known as Quadruplex recorders, were crude by present standards, but over a period of more than 20 years, continual improvements were made which exploited nearly all of the abilities of the recording format. By the late 1970's, in spite of all the improvements, quadruplex recorders still suffered from several major shortcomings; they were physically quite large, expensive to purchase, and expensive to operate. Head costs were about $4.00 per hour, and tape consumption amounted to 30 square inches per second. Finally, because the format divided the picture into 17 horizontal bands of 17 scan lines each, it was often difficult or impossible to eliminate differences between the bands that made them visible.

By the mid 1970's the users had made known through various bodies, such as the European Broadcasting Union and the NAB, that smaller, better recorders using less tape and costing less to operate and to maintain were needed. Four major manufacturers of television equipment responded to the pressure and introduced four different recorders employing four completely different formats, all of which employed the helical scan technique in one way or another.

The helical scan format had been thought of and extensively tested during the early development of the video tape recorder by the Ampex engineers. It was obvious that it was a simpler format and that a VTR using it would be less expensive in several areas. It was also obvious, however, that there were two major problems which at that time seemed insurmountable. In a helical recorder the video track is at a very small angle with respect to the direction of tape motion, thus instability of the tape movement caused very large time base errors that caused wild swings of several microseconds of the picture when it was displayed on a monitor with a long time constant in the horizontal AFC circuit. At that time most receivers did employ very long time constant AFC circuits. It seemed unlikely that the scanner with its large mass and inertia could be made to respond to these rather rapid variations, and of course the first electronic time base corrector had not been invented. Because in quadruplex format the tracks are almost at right angles to the tape motion, only insignificant amounts of the longitudinal speed errors are introduced into the video signal. This problem plagued the helical format for professional use until wide range digital time base correctors were introduced much later.

Additionally, a great deal remained to be understood about the tape-to-scanner and guiding

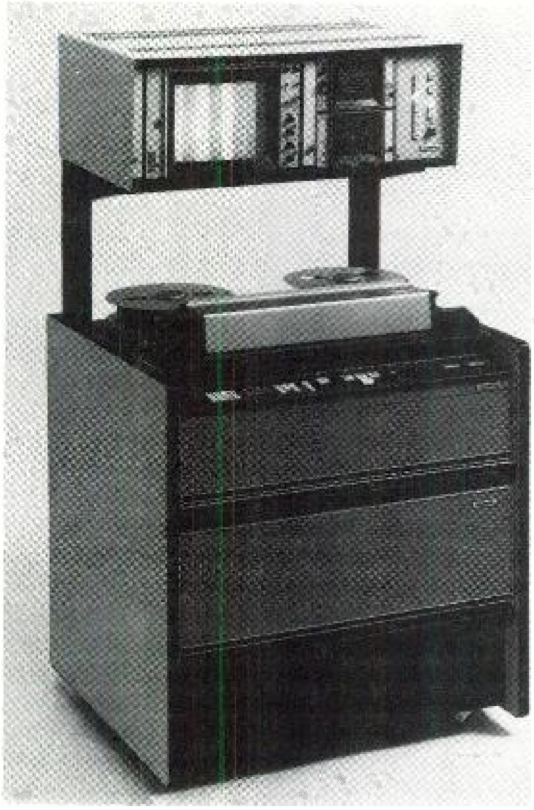


Fig. 1. Ampex AVR-2 Quadruplex Recorder.

elements relationship. Because of the large area of tape rubbing against the scanner and the tension build-up as the tape passed around stationary guides, many, many problems were encountered. In extreme cases tape would stick to the scanner and refuse to move. In other cases tape was deformed by the excessive tension needed to pull the tape past the guides and around the scanner. Only gradually were these problems understood and solved through the forming of new tape surfaces and development and use of new materials and design techniques for scanner and guides.

One of the new recorders introduced fell by the wayside fairly early on, but three remained. In Europe Bosch introduced the BCN series of recorders that employed a small scanner, a 180 degree wrap of the tape around the scanner and which segmented the picture as had the quadruplex format. It used 1'' tape and consumed 10 square inches of tape per second. In the United States Ampex re-introduced an older format it had employed for some years on its line of industrial recorders and introduced the new VPR-1 recorder at the NAB show in 1976. In Japan Sony Corp. developed a recorder employing a format that superficially resembled the Ampex format and introduced its BVH-1000 recorder at the 1976 NAB show as well.

The Sony and Ampex recorders were close enough to each other that there was some hope that they could be brought together in a common format. Both used 1'' tape, an omega wrap in which the tape was almost completely wrapped around the scanner, and one television field was recorded for each revolution of the scanner. A small gap of about 10 television lines existed where the wrap was incomplete. The Sony recorder employed an extra set of heads on the scanner to record the missing scan lines near the bottom of the tape, but the Ampex format simply ignored the missing lines. The scanner diameter, the tape speed and other characteristics were tantalizingly similar. The Bosch format, however, was strikingly different from the other two.

The Society of Motion Picture and Television Engineers, (SMPTE), formed an Engineering committee to see if a common format could be found to which everyone could agree. It became quickly apparent that the Bosch format was too different and that Bosch could not see its way clear to change. The other manufacturers did at least seem willing to consider the possibility of a change. In about one year the SMPTE committee had drafted a format that borrowed from each of the two formats, added a few embellishments of its own, and to which everyone on the committee had agreed. Many technical experts and users participated in the committee's work, and the expertise of these people was recognized by the manufacturers. In a relatively short time there were four suppliers of equipment that conformed to the new format that had been assigned the arbitrary letter C as a designation by the committee. Thus was the C Format born. The Bosch format was given the designation of B. The A designation was never assigned permanently.

The characteristics of the two formats are spelled out in several ANSI Standards and SMPTE Recommended Practices that are reproduced later in the text. Copies of these documents may be purchased from the Society of Motion Picture and Television Engineers, 862 Scarsdale Ave., Scarsdale, New York 10583.

QUADRUPLEX RECORDERS

Many quadruplex recorders are still in service and are expected to be around for a number of years, so a brief overview of their characteristics is in order. The previous edition of the NAB Handbook contained a complete description of the format and is recommended for those who wish to know more about it.

The tape transport closely resembles similar mechanisms found in high quality audio recorders of that day. The tape is pulled from a supply reel and across the erase, record, and playback heads by a rotating capstan and then wound by a motor-driven takeup reel as shown in Fig. 2. The primary longitudinal tape speed is 15 inches per second, the secondary is 7.5 inches per second. Not many people use this secondary speed.

As the tape is drawn past the erase head and idler it is deformed to form a canoe shape as it passes by the rotating video head wheel upon which are mounted four video heads spaced precisely 90 degrees apart around the wheel. The tape is positioned exactly by the vacuum guide as illustrated in Fig. 3.

In 525 NTSC operation, the head is rotated at 240 rps. The result is a series of video tracks written across the tape at almost 90 degrees to the direction of tape travel. Audio and control tracks are written along the two edges of the tape. Servos control the rotation of both the capstan and the video head wheel so that the signals are replayed with minimal timing errors. A 240 Hz Control Track (4 × Field Rate) is recorded along a longitudinal track concurrently with the video and audio recording. In playback the recovered 240 Hz signal is compared to a reference frequency to position the video heads exactly over the recorded video tracks. In recording, the head-wheel motor is driven by the same 240 Hz signal recorded on the control track. In playback the off-tape sync signal is compared to the reference sync signal and the drive to the headwheel motor modulated to vary the speed of the motor and thus the timing of the reproduced sync and video signal. Residual time base errors (amounting to about .15 microsecond peak-to-peak at this time) are removed by time base correctors that reduce the residual errors to just a few nanoseconds.

The first timebase correctors were analog devices that employed voltage-variable delay lines and had a correction range of about a microsecond or so. The earlier variants usually had first a unit to correct gross errors in which off-tape sync pulses were compared to reference pulses. A second unit compared the off-tape burst phase to a reference color subcarrier, and a correction voltage, proportional to the phase error, was applied to the delay line. Finally a third unit corrected for velocity errors. Near the end of quad's life, digital time base correctors were used that combined all the time-base correction functions as well as video processing into one unit that became known generally as the TBC. Video processing prior to the advent of TBCs was accomplished in a separate unit that cleaned up the sync signal, clamped the signal, inserted new burst and clipped noise and switching transients.

The quadruplex video head assembly was a complex affair that was relatively expensive. The position of the vacuum guide with respect to the center of rotation of the head wheel was critical and had to be adjusted both vertically and horizontally if hooks and other geometric distortions were to be avoided in the reproduced picture.

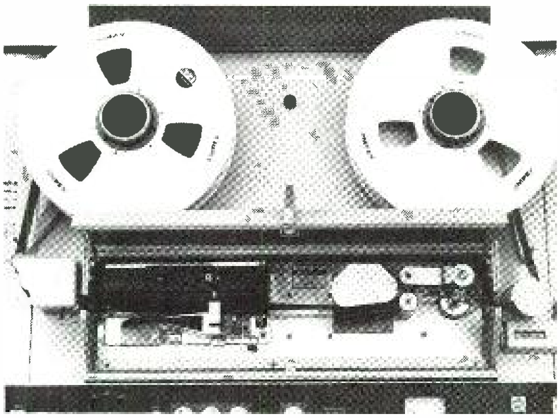


Fig. 2. An early quadruplex tape transport showing the tape path.

THE C FORMAT HELICAL RECORDER

The Tape Transport

In general the tape transport can be described in a way similar to that for the quadruplex transport; tape is pulled from a supply reel and across the erase, record and playback heads by a rotating capstan. In detail, however, the C format transport is very different from a quadruplex transport. The familiar full-width erase head, control track head, and audio heads are still there, but the scanner replaces the quadruplex head wheel. A diagram of the tape path of an Ampex VPR-3 is shown in Fig. 4.

Tape unspools from the supply reel and is looped around a tension arm that senses the tension in the tape as it leaves the supply reel. Data from the tension arm is used to adjust the hold-back torque of the supply reel. The tape then passes over the full width erase head, past a guide and approaches the scanner and is about to enter the most critical part of its journey to the take-

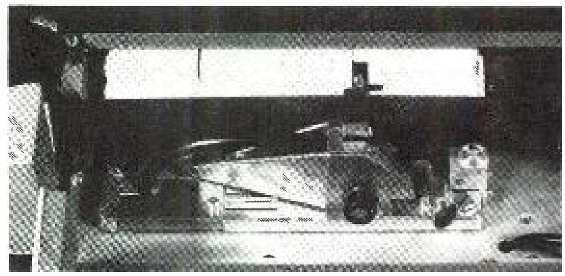


Fig. 3. Head assembly of AVR-2.

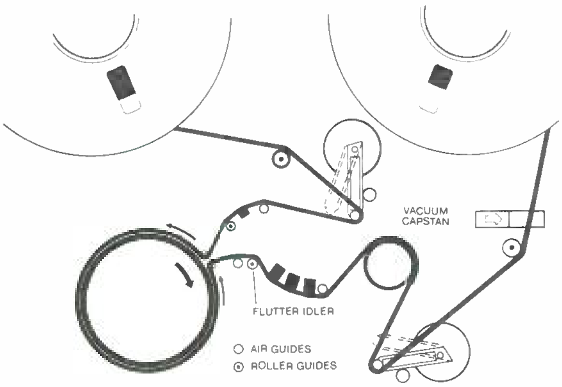


Fig. 4. Tape path of Ampex VPR-3.

up reel. The entrance guide very near to the scanner deflects the tape upwards as it starts around the scanner, and the adjustment of this guide is critical. The tape describes a helix as it climbs around the scanner and finally exits the scanner at the exit guide, thus the name of helical recording.

The upper half of the scanner in which the various video heads are mounted rotates and not only sweeps the heads past the tape but also pumps air between the tape and the scanner to reduce friction between the two. In the C format there can be six heads mounted on the scanner. The position of these heads is defined in ANSI Standard V98.18 and is shown in Fig. 6.

Because the entrance and exit guides must have a small space between them to allow the tape to pass, there is a brief period during which video is not recorded. Upon playback there will be a gap of about 10 video lines where no information is available. This gap is positioned in the vertical interval, starting approximately at line 6 on odd numbered fields. Since the missing information contains known, defined information, it can easily be reconstructed. At this time there is no use of these lines for transmission of data, such as VITS, other than vertical synchronizing pulses.

Provision is made, however, for the recording and recovery of the missing information by employing a separate group of heads, referred to as Sync Heads. Each of the two groups of heads contains an erase head, a record head, and a playback head. The C format defines the Sync head group as optional, but if it is not used, dummy heads must be installed to minimize velocity errors. These errors can arise because as a video head travels across the tape it produces a bow wave just as a ship does, and just as a wave does when it strikes the edge of a canal the bow wave from the head is reflected back into the path of the following head. If the wave were present during recording on one machine but were missing during playback on another machine, velocity errors would be generated, and for this reason six heads or three heads and three dummy heads must generate six bow waves.

After leaving the scanner the usual array of audio erase and record/playback heads is encountered. The capstan meters the tape, and

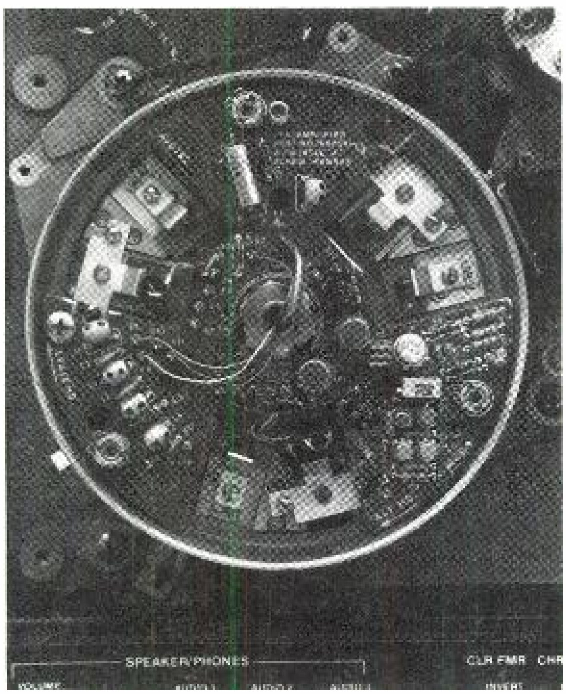


Fig. 5. C format scanner showing six heads.

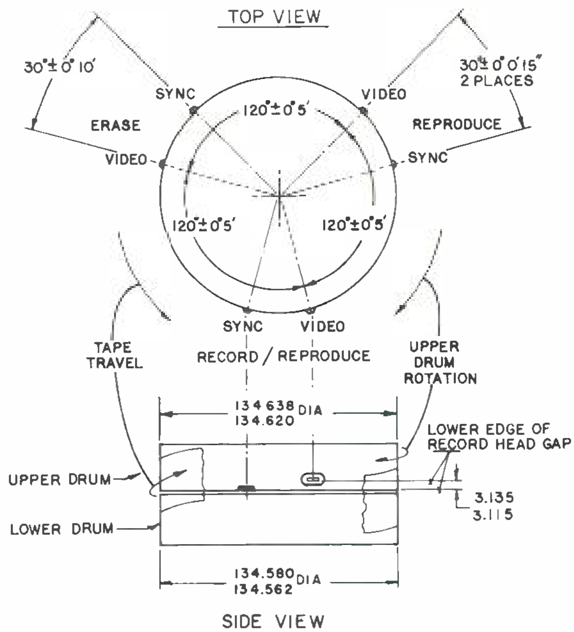
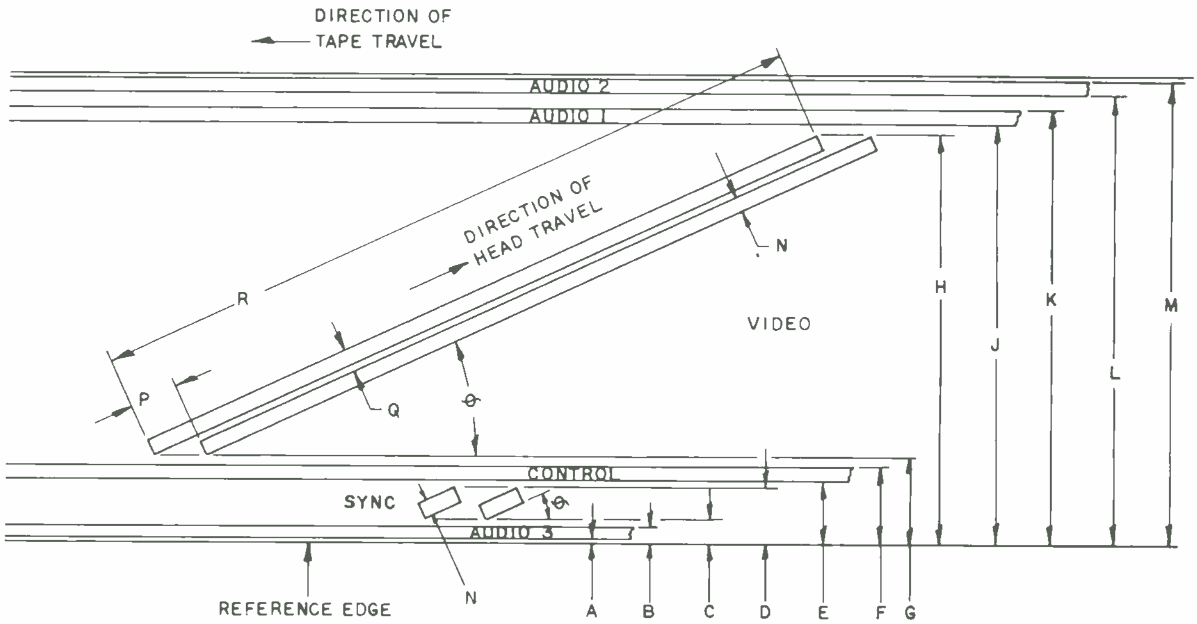
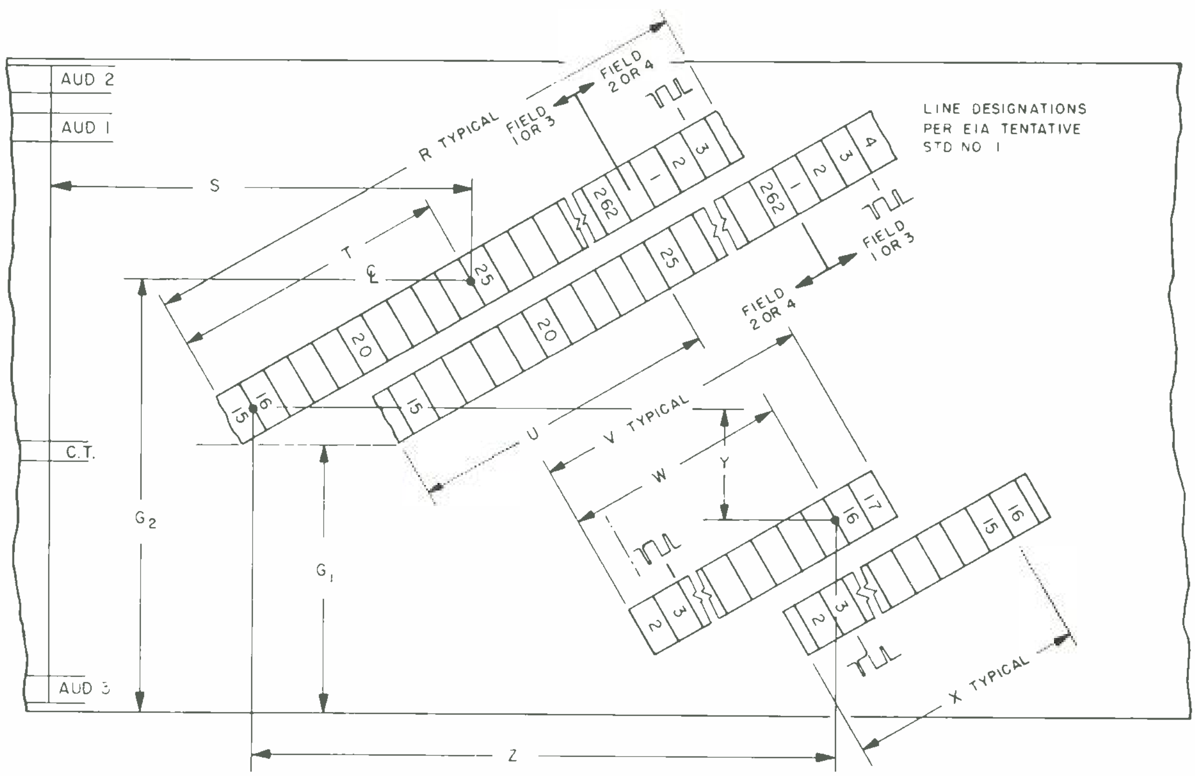


Fig. 6. Pole tip locations and drum dimensions.



Record Location and Dimensions



Video and Sync Record Location

Fig. 7. Video and sync record location.

Record Locations and Dimensions

	Dimensions	Millimeters Minimum	Millimeters Maximum	Inches Minimum	Inches Maximum
A	Audio 3 lower edge	0.000	0.200	0.0000	0.0079
B	Audio 3 upper edge	0.775	1.025	0.0305	0.0404
C	Sync track lower edge	1.385	1.445	0.0545	0.0569
D	Sync track upper edge	2.680	2.740	0.1055	0.1079
E	Control track lower edge	2.870	3.130	0.1130	0.1232
F	Control track upper edge	3.430	3.770	0.1350	0.1484
G_1	Video track lower edge	3.860	3.920	0.1520	0.1543
G_2	Video line 25 start	4.650	4.710	0.1831	0.1854
H	Video track upper edge	22.355	22.475	0.8801	0.8848
J	Audio 1 lower edge	22.700	22.900	0.8937	0.9016
K	Audio 1 upper edge	23.475	23.725	0.9242	0.9341
L	Audio 2 lower edge	24.275	24.525	0.9557	0.9656
M	Audio 2 upper edge	25.100	25.300	0.9882	0.9961
N	Video and sync track width	0.125	0.135	0.0049	0.0053
P	Video offset	4.067 (2.5H) ref		0.1601 ref	
Q	Video track pitch	0.1823 ref		0.00718 ref	
R	Video track length	410.764 (252.5H) ref		16.1718 ref	
S	Control track head distance	116.23	117.03	4.567	4.607
T	Vertical phase odd field	16.270 ref (10.0H)		0.6406 ref	
U	Vertical phase even field	17.080 ref (10.5H)		0.6724 ref	
V	Sync track length	25.620 (15.75H)	26.420 (16.25H)	1.0087	1.0402
W	Vertical phase odd sync field	22.360 (13.75H)	23.170 (14.25H)	0.8803	0.9122
X	Vertical phase even sync field	23.170 (14.25H)	23.980 (14.75H)	0.9122	0.9441
Y	Vertical head offset	1.529 ref		0.0602 ref	
Z	Horizontal head offset	35.350 ref		1.3917 ref	
θ	Track angle	2°34′ ref			

Fig. 7. Video and sync record location. (Con't.)

a second tension arm senses the take up tension and supplies data to the take up reel servo.

The pattern of the various tracks on the tape is defined in ANSI Std V98.19 and is shown in Fig. 7.

Guiding of the tape around the scanner is particularly critical because of the very long length of the video track. Ideally, the track should be perfectly straight. If it is, a rectangular burst of modulated carrier will be obtained on interchange playback. If it is not, then shapes such as those in Fig. 8 will be obtained. There are a number of factors that influence track straightness of which, the most critical that can be adjusted by the operator are the following.

1. Supply and take up tensions. These tensions can be most easily measured by use of a tensions gauge such as the one shown in Fig. 9. The manual for the particular machine being adjusted will explain how to set the correct tension.

2. Entrance and exit guides. A standard reference tape is used as a practical way to check for correct alignment. If adjustments are provided, they are used to playback a burst of carrier whose envelope is as rectangular as possible. If severe problems in interchange are encountered, the manufacturer of the equipment should be consulted.

As scanners wear after many, many hours of use it may be necessary to replace either the entire scanner or at least the upper rotating half of it. A note here that one manufacturer provides for replacement of individual heads on the scanner while another requires that the upper, rotating half of the scanner be replaced.

There may be adjustments as well for positioning of the vertical drop out by moving the entrance and exit guides circumferentially around the scanner, and for positioning longitudinally of the control track head. As always, the operating manual for the equipment should be consulted and followed, and also as always the transport should be kept clean and free of oxide build ups. Generally the transport can be cleaned with xylene except for the capstan pressure roller which, if used, may be damaged by the use of such a solvent. Instructions by the manufacturer should be followed carefully to avoid such damage.

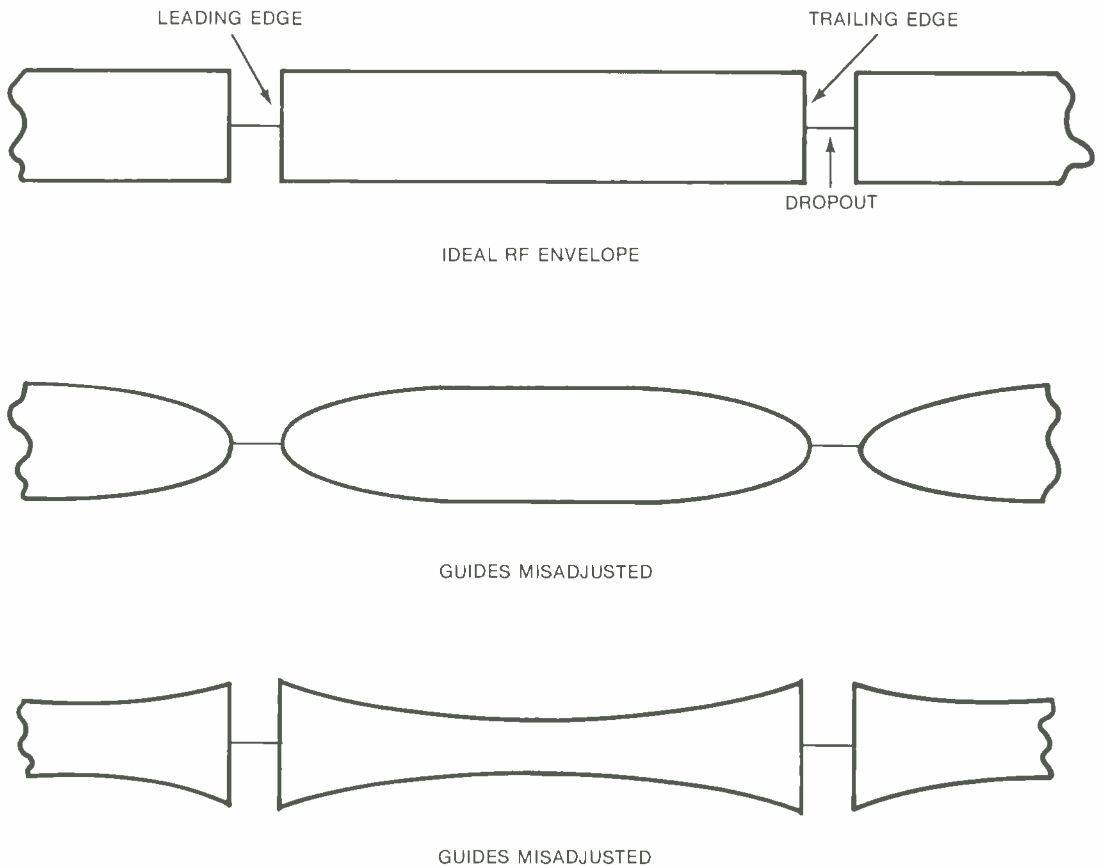


Fig. 8. Drawing of RF envelopes.

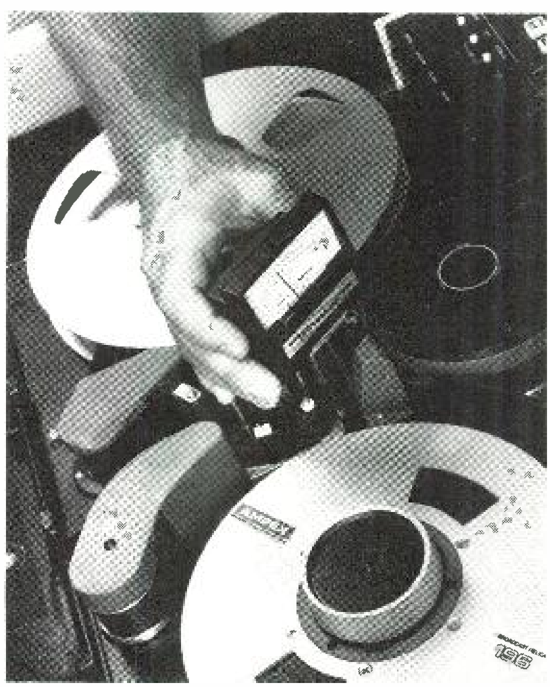


Fig. 9. Use of tape tension gauge.

Servo Systems

Capstan and Scanner Servos

Of the several servo systems within the recorder, the capstan and scanner servos are most interdependent, and in broad terms they are similar to the quadruplex capstan and headwheel servos. During the recording process these two servos are referenced to the synchronizing information in the video signal to be recorded. Capstan speed, and thus longitudinal tape speed, is directly related to the vertical synchronizing signal, and a control track signal derived from the video synchronizing signal is recorded on the tape. This signal's waveform is defined in SMPTE Recommended Practice RP-85, and its location in Standard V98.19 illustrated earlier. The position of the scanner, and thus the rotational position of the video record head, at the instant of the vertical synchronizing pulse is controlled by the scanner servo that compares vertical sync against a pulse generated by the scanner and whose location is known with respect to the video record head. Usually there is a maintenance adjustment that assures correct positioning of the record head, and the service manual should be consulted for instructions on how to make the adjustment.

During playback the servo system most often uses a reference video signal such as color bars or sync and burst, but lacking those, an internal oscillator can be used. Vertical sync is compared to the recovered control track signal, and the speed and phase of the capstan motor adjusted. The recovered vertical sync signal off tape is also compared to the reference sync and the speed and phase of the scanner adjusted to provide coincidence in time between reference and off-tape vertical sync. Additionally, reference and off-tape horizontal sync signals are compared and the error signal used to modulate the drive to the scanner motor in order to minimize errors in the timing of off-tape horizontal sync signals. Residual timing errors should be a few microseconds peak-to-peak.

The major component of these errors is relatively large as noted above, but because the large mass of the rotating upper scanner and its drive motor limits the scanner servo's ability to compensate for time base errors that change value rapidly, several smaller, rapidly changing errors are also left along with the large peak error just mentioned. Mentioned earlier were velocity errors that could be caused by a bow wave if six heads were not included on the scanner. Even with six heads, the effects of the bow wave may not be exactly compensated for. Additionally, scrape flutter cause by passage of the tape over guides or over the stationary lower scanner where the tape is not supported by an air film can result in small but significant high frequency time base errors.

Tape Tension Servos

If tension on the tape around the scanner is not the same as it was during the recording process, a major error results that is evidenced by the familiar hooking at the top of the reproduced picture when displayed on a monitor with a long horizontal oscillator AFC time constant.

To be sure that tape tension is the same during record and palyback, both the supply and take up reel motors are servoed during record and playback. The components sensing the tensions are the tensions arms near the supply and take-up reels.

Fig. 10. Tape tension arms on modern C format VTR.

Attached to these arms are sensors that translate the position of the arms (which is an indication of the tension) into an error signal that can be used to control the drive to the reel motors. Tape tension should be checked on a periodic basis, for if allowed to drift too far from the norm it can seriously affect interchangeability.

Video Head Position Servo

A significant event in the maturing of helical recording technology was the development of a method of mounting a video head on a flexible beam that could be deflected by electrical means. The system was developed by Ampex engineers and is now employed by all manufacturers of type C helical VTR's as well as on some of the more sophisticated 3/4" U-Matic recorders. Ampex chose to call the system Automatic Scan Tracking or AST for short. The other manufacturers have chosen other names, but the principal is the same in all cases.

A thin metal plate is anchored firmly at one end, and the video head is mounted at the far,

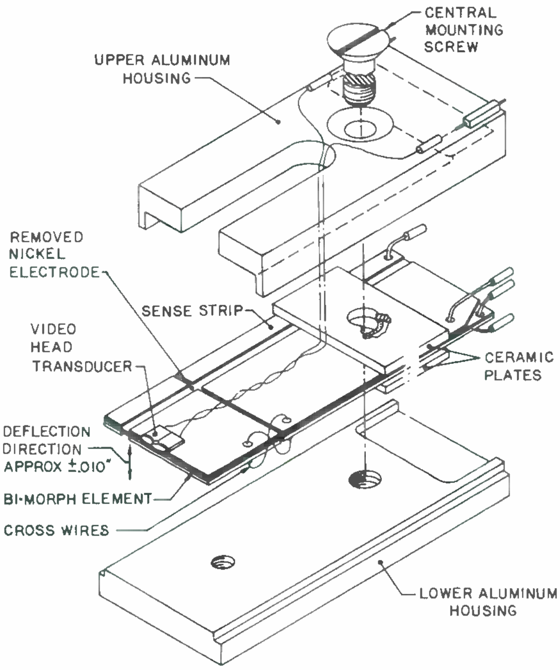


Fig. 11. Drawing of AST head.

unsecured end. a material exhibiting piezo-electric qualities, such as barium titanate, is attached to the top and bottom of the plate. When a voltage (about 200 volts) is applied to the piezo-electric material, the material will contract or expand, depending upon the polarity of the applied voltage, and cause the plate to bend up or down.

A supplemental piezo electric element is attached to the plate to act as a transmitter to send a voltage back to the servo to tell it how far the plate has been deflected. When calibrated, a closed loop servo is obtained that can move the head at the end of the plate to a precisely defined position. The amount of movement is relatively large and is enough to deflect the head over two video tracks, approximately 0.020 inches. Fig. 12 shows an AST head from a VPR-2B.

The introduction of such a head made several outstanding features possible on helical VTRs. With proper care there is no reason there should be interchange problems with VTRs conforming to the C format, but every user knows that proper care is not always a reality; curved, non-standard tracks do occur in practice. The use of moveable heads was first seen as a way to allow playback of such non-standard tapes without degradation in quality.

During playback a sine wave voltage is applied to the piezo electric material at a frequency of 450 cps, causing the head to cross back and forth over the video track as it travels along the recorded track. The result is a modulation of the envelope of the recovered FM carrier. If tracking is perfect the phase and frequency of the modulation will be one thing. If tracking is not perfect the modulation pattern will be different, depending upon which side of the track is experiencing the greatest loss of tracking. This difference can be used as an error signal to generate a correction voltage that will deflect the head back to the center of track, even if that center is not where it is supposed to be.

When longitudinal tape motion is stopped on a helical recorder and the scanner is allowed to continue rotating, a recognizable picture is obtained because one sweep of the track reproduces one field of the picture. Tracking is not perfect, however, and a noise bar is usually evident in the picture. When the AST servo is engaged, however, and given the ability of the head to be deflected enough, the track can be followed and a noiseless picture reproduced. The time to travel the track will be slightly wrong, but a properly arranged TBC can correct that problem. It can also be quickly seen that if the tape is moved slowly in the longitudinal direction, and if the head has sufficient mechanical deflection ability, slow motion playback is possible. Time base errors remain, but with proper coordination between the VTR and the time base corrector, the errors can be eliminated and a perfect slow motion picture obtained. The coordination between VTR and TBC is relatively simple inasmuch as only direction and tape speed need to be given to the TBC, but the extra circuitry needed in the TBC to sort out the proper timing can become complex and add to the cost of the TBC.

With the introduction of VTRs that can provide slow motion capability, the older disc slow

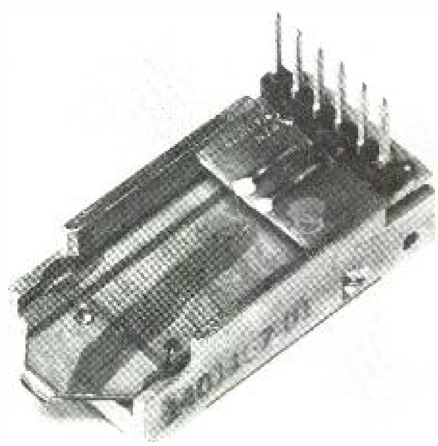


Fig. 12. AST head assembly.

Fig. 13. HS-100 slow motion recorder.

motion recorders were rapidly retired from service. The Ampex HS-100 slow motion recorder introduced in 1967 used two double-sided, hard-coated discs that provided 30 seconds of continuous recording time. Because analog recording, rather than digital, techniques were used, heads were flown very close (about 12 microinches) to the surface of the disc, and head crashes were frequent. The HS-100 was endured (and hated) only because there was nothing better. Very few, if any, remain in service at this time.

Video Heads

General

Quadruplex recorders have, for the most part, used video heads with metal tips attached to ferrite around which the coils are wound. It was recognized very early that if the pole pieces themselves were made of ferrite, wear would be better since the ferrite was much harder than the metal; however, there was one major problem that prevented the use of ferrite. When the heads of a quadruplex recorder encountered the tape, they struck it relatively suddenly and immediately dug deeply into the tape and stretched the tape within narrow confines. Shock and pressure were high. When an attempt was made to use ferrite under these conditions, the ferrite simply could not withstand such abuse and broke up and chipped away. Only in the latter days of quadruplex recording were techniques developed that allowed the use of ferrite.

Helical recorders are much easier on heads since there is no vacuum guide as in quadruplex recording to confine the tape in the area around the head, and the head encounters the tape in a much gentler fashion. Accordingly, ferrite heads have been used on helical recorders for many years. The hardness of the ferrite and the gentleness with which it encounters the tape contribute to the large number of hours of use that can be expected from helical heads in contrast to quadruplex heads.

Because ferrite is brittle, rough handling and carelessness when cleaning helical heads can destroy the head. Follow instructions given by the manufacturer to the letter. In general, cleaning should be done with circumferential motion rather than motion across the head.

Head Construction

The actual transducer is small and is usually mounted on a larger supporting structure. Fig. 14 shows a record/playback head from a VPR-2B. The actual transducer is mounted at the very end of the large structure. This particular head assembly can be replaced by the user in the field. The helical head is made up of two pieces of ferrite whose inner surfaces are carefully lapped. The two pieces of ferrite are brought together with a non magnetic material between the surfaces in the front part of the transducer to produce the head gap.

The Scanner Assembly

Signals are brought to and from the video heads by rotary transformers. Unlike quadruplex head assemblies, the reproduce preamplifiers are mounted within the rotating scanner and send

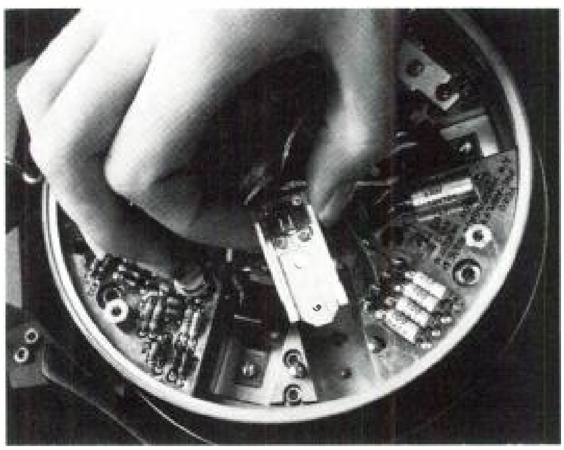


Fig. 14. Record-playback head from Ampex VPR-2B.

Fig. 15. Top, rotating part of scanner, including pre-amps.

highlevel playback signals to the signal system through the rotary transformers.

The scanner assembly is reasonably complex. It must contain the drive motor, the rotary transformers, and slip rings which bring control voltages to the AST head and to the preamplifiers. It also contains bearings to allow rotation of the upper scanner, and the preamplifiers, themselves.

Video Head Life

Measurement of Head Wear

The tips of the video heads protrude a few thousandths of an inch beyond the circumference of the scanner and thus actually dig into the tape. The amount of protrusion is an indication of the amount of life left in the heads. When new, the heads protrude about 0.0036 inches. At end of life the protrusion is about 0.0012 inches. End of life usually comes when enough material is worn away to allow the back of the gap to break through.

Most manufacturers provide as an accessory a dial indicator and holding fixture that can be attached to the transport and used to measure the projection of the head tip beyond the surface of the scanner.

Factors Affecting Video Head and Tape Life

Video head life and video tape life are related. Video head wear is affected by the characteristics and condition of the tape used, the cleanliness of the equipment, the environment of the operational area, adjustment of the VTR tape transport, and the general handling procedures of the VTR and the video tape. Similarly, the video tape life and condition are affected by the condition of the video head, the tape path surfaces of the video recorder and all of the environmental conditions mentioned in conjunction with head life. Additionally, the condition of reels and the circumstances of packaging, shipping, storage, and above all by the video tape handling procedures exercised by the VTR operator.

Video head wear is of great concern in relation to the cost of operation as well as providing high quality performance of the video recorder. Helical recorders generally exhibit lower rates of video head wear than do quadruplex recorders because head-to-tape mechanical pressure is less and the abrasion by the tape is less. One can reasonably expect head life on a helical recorder to be at least three times the head life of a quadruplex recorder.

All magnetic recording tapes present an abrasive surface to the video head tips, and head wear begins the moment the first tape is recorded or reproduced. The abrasiveness of tape is one of the critical characteristics and should be determined before large quantities of a brand are purchased. The rate of head wear produced by good quality tape is finite but relatively low; damaged tape is quite another matter. Head wear is also related to the moisture content of the tape binder. The higher the moisture, the higher the rate of head wear. The moisture content is determined by the humidity at which the tape is stored and by the humidity of the operating environment. There is a marked increased in wear when the humidity is much above 40%; therefore, the storage environment as well as the operating environment relative humidity should be maintained in the range of 35 to 45%.

A real killer of video heads, helical or quadruplex, is damaged tape. The head and tail ends of tape become creased and frayed with use, and

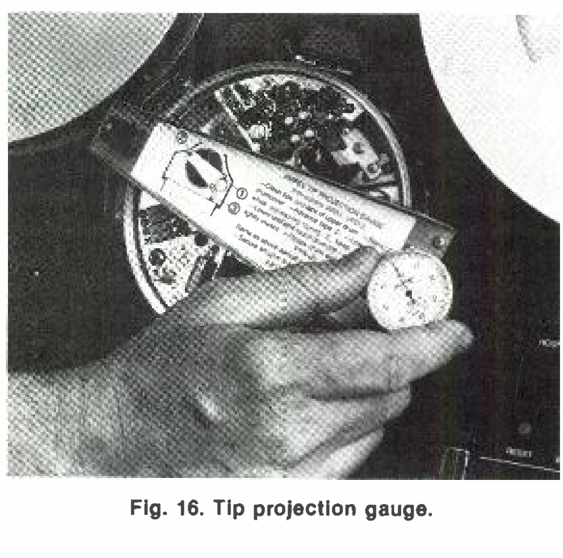


Fig. 16. Tip projection gauge.

if these damaged ends are not cut off they can and will destroy catastrophically the video heads as the tape end flies by the heads when tape is being rewound or spooled off a reel.

Tape that has been physically damaged by creasing, crumpling, or scratching causes high wear rates. And sharp deformation or scratching of the oxide coating interrupts its continuity and uniformity and results in an action somewhat akin to that of a file. It also loosens individual particles of oxide that may be picked up and retained by the video head tips, the scanner surface or any of the other guiding surfaces or the audio head stacks.

As a minimum, before each day of use, closely inspect all metal surfaces such as tape guides, idlers, compliance arms, scanner, and audio head stack, which touch the tape for an accumulation of oxide and dirt. If any is present, remove it by using a cotton-tipped swab moistened with head cleaner. On quadruplex recorders there is a tendency for oxide to build up on the head wheel, and the wheel should be cleaned before every recording or playback.

While tape is being manipulated by hand, use care to avoid creasing or crumpling. Do not handle the tape any more than necessary, for in addition to mechanical damage, oil from the skin will cause difficulties.

The environment in the area where the tape is actually used should approach as closely as possible, while still being practical, that of a "cleanroom." Such an area is free of normally expected airborne dust and lint. Various airconditioning filtration systems are available to accomplish the cleaning.

Surrounding walls and ceilings should be painted and free from material that will attract and hold dust. Air intakes should be well-filtered to eliminate outside dust and dirt from entering. Anything, such as empty boxes or packing materials, that tend to collect dirt and dust should not be brought into the area. Video tape equipment, racks, shelves, and video tape reels must be kept clean and free from dust and dirt.

The design of the equipment area should be such that control of temperature and relative humidity can be exercised. Variations of temperature should be held to ±5 degrees Farenheit and the relative humidity should be kept constant to within ±10%. The general recommendation is to have an environment that is comfortable for the operating personnel, for such an environment is also good for tape. Generally this means a temperature in the low 70's and a relative humidity of about 35%. Cigarette ashes are not good for tape; therefore, smoking in the area should be discouraged, and food and drink should be restricted. Small food particles can be easily transmitted to the video tape and recorders from the operator's hands and result in excessive drop out activity.

Aside from the direct benefits gained from a clean, well maintained, temperature and humidity controlled environment, the psychological effect on the employee is also of great importance. Operators will exercise more care and be concerned about quality when working in such an environment. Cleanliness in video tape areas is absolutely essential to the long-term life of video tape material stocks and to minimum video head wear.

Tape should be stored in clean, temperature and humidity controlled areas. Humidity should be 35-45% and temperature should be 60-65 degrees F. It should be stored in boxes and on edge, not flat. When brought from storage into an operating environment, the tape should be spooled at slow speed (100-150 ips) back and forth and allowed if possible to sit for several hours.

When shipping tapes it is best to use high-impact, plastic cases available from most manufacturers of tape.

The Signal System

General

All video recorders, both professional and consumer, use essentially the same type of signal system since the function is identical; a varying amplitude video signal is converted to a frequency modulated carrier for recording on the tape. A carrier is used for recording on the tape because the tape can only accommodate a limited number of frequency octaves and a video signal requires many octaves. If the same video signal is modulated onto a high frequency carrier, the number of octaves that must be reproduced is small and can be accommodated. Frequency modulation is used because of its performance in the presence of noise. This is important because the recovered RF signal is not of constant amplitude or signal-to-noise but is varying as a function of mistracking and head-to-tape conditions. Both of these are quite variable, and observation of the recovered RF envelope shows that it is constantly varying in amplitude.

The frequency modulated carrier is amplified and split into the required number of channels to match the number of record heads employed by the format. On quadruplex machines this is four channels, and on C format this is only one channel. The amplified signal is coupled to the rotating scanner or head wheel by rotary transformers. Each head on a quadruplex recorder receives the signal continuously and records the signal while the head is in contact with the tape.

In C format helical recorders, the current to the head is turned on and off at precise times to insure the recording is made only where desired on the tape.

During playback the very weak signal from the heads is coupled to preamplifiers mounted as closely as possible to the heads. On quadruplex recorders the signal is first sent to the rotary transformers for coupling to the preamplifiers mounted on the head assembly. On professional helical recorders the preamplifiers are mounted inside the scanners and rotate with the video transducers. The amplified signal is then coupled to the equalizers, limiters, and demodulators through the rotary transformer(s). The bandwidth of the signal system extends from approximately 2 MHz up to approximately 18 MHz, or just slightly over 3 octaves.

Modulation

There are two basic types of modulators used, multivibrator and heterodyne. The Ampex VPR-2B recorder, for example, uses a multivibrator operating at twice the desired frequency to avoid second harmonic problems. It is divided later by two to the desired frequency.

Usually the video input signal to the modulator is band limited by a low pass filter to prevent spurious sidebands generated by frequencies above those normally encountered in the standard being used. It is then clamped to provide a stable reference point defined by the back porch of blanking. All professional recorders, both B and C format as well as some U-Matic recorders, now use High Band frequencies in which the blanking carrier frequency is 7.9 MHz, peak white is 10.9 MHz and tip of sync is 7.06 MHz for 525/60 systems. Quadruplex recorders before 1964 used low band frequencies, one set for color, and another set for monochrome. Both high and low-band frequencies are defined in SMPTE Recommended Practice RP-6. Some of the old, archival recordings that might be encountered may use these older low-band frequencies, and of course some of the home VTRs and lowband U-Matic VTR's also use lower carrier frequencies.

Some degree of pre-emphasis is applied to the video signal before it is delivered to the modulator in order to improve the signal-to-noise ratio of the reproduced signal. This is a well known technique and its use here is completely conventional. The amount of pre-emphasis employed is also defined in the SMPTE Recommended Practices just mentioned in association with carrier frequencies.

The choice of carrier frequencies is determined by the level of performance desired. Generally speaking, the carrier is made as high as possible consistent with the ability of the VTR to recover the carrier in order to minimize moire or spurious frequency components caused by folded sidebands generated when significant sidebands are produced by the modulating signal and these sidebands approach zero frequency and then fold back to produce sidebands at the "wrong" frequency and thus cause spurious signals in the demodulated signal.

The Playback System

The frequency modulated signal recovered by the head is transferred by rotary transformer to a further amplifier. After amplification to a reasonable level the signal is applied to an equalizer that compensates for frequency response characteristics of the video head to provide a flat response. Trim controls are also provided to minimize differential phase and gain errors. If more than one head is used in the format, such as in the B format or in quadruplex recorders, each head output is equalized separately, and then a switcher, timed by the position of the scanner or head wheel, switches between the heads just as one head leaves the tape and another enters the tape. Thus a continuous signal is provided to the limiting amplifiers.

There are a number of ways the video can be recovered from the frequency modulated carrier. Early quadruplex machines employed a simple slope detector, but over the years many methods have been employed. Recent Ampex VTRs employed carrier crossing detectors. In other VTRs the manufacturers have produced integrated circuits represented as a simple block on the circuit diagrams and the inner workings of the IC remain a mystery to the user.

After demodulation the video signal is clamped on the back porch employing conventional techniques, and the vertical dropout, if there is one, is clamped to remove noise caused by the loss of signal as the head transists the gap between the exit and entrance guides to the scanner. If separate heads are employed to recover all of the vertical sync signal, the recovered synchronizing signal is inserted into the 10 line gap. Finally the signal is amplified and supplied at the output of the system at the customary level of one volt peak-to-peak at an impedance of 75 ohms.

A glance at the schematics of a variety of 1″ and 3/4″ recorders reveals a variety of circuits and techniques used to recover the signal and to process it before the output of the recorder. The user must judge the entire system of signal recording and recovery by the results obtained in terms of signal-to-noise of both the luminance and chrominance signals, and of the residual spurious signals (moire) in the video signal and, of course, by such other criteria as sine squared response, differential gain and differential phase.

Time Base Correction

General and Historical

If a video signal is to be replayed with each picture element positioned in time with respect to any other picture element exactly as it was in the original signal, the head to tape velocity on playback must match exactly the head to tape velocity at all instants during the recording. In spite of the use of servo mechanisms to control the rotation of the drum or scanner as precisely as possible, only so much can be done. Not only may there be instabilities in drum or scanner rotation, but there may also be instabilities in the longitudinal speed of the tape. Additionally, there are other disturbing things that tape has a habit of doing such as fluttering in local areas. The net result of the instabilities is that the reproduced signal has time base errors that must be removed for professional use of the signal. Home class VTRs usually have substantial time base errors, but newer receivers usually can ignore them and various schemes are used to recover the unstable chrominance information. Quality suffers considerably using these techniques. For professional use, however, where highest quality is required and editing and other post production techniques are to be used,the recovered signal must be time base stabilized.

Before discussing time base correction it is well to remember that there is no continuous, coherent signal that can be used as reference in the composite television signal. Even the chrominance subcarrier is suppressed and not present at all times. Two recurring signals are, however, present at all times, the scanning synchronizing pulses and the color burst that is present after each sync pulse other than those in the vertical interval. It is these two signals that are compared to a reference, stable signal in order to measure the amount of time base error present.

Time base correction occurs in four stages. The first step involves control of the rotational velocity of the scanner, or head wheel, and of the longitudinal motion of the tape. This step is discussed separately under Servo Systems. The residual jitter or time base stability after this step in quadruplex recorders is about ±0.07 microseconds and in helical recorders, such as the Ampex VPR-2 series, is about 7 to 10 microseconds. Any following correction circuit must be able to accept this magnitude of variations.

Earlier quadruplex recorders such as the VR-1200, VR-2000 and TR-70 employed analog techniques. The heart of the system was a delay line of about one microsecond in length which was made up of pie sections of L and C. The capacitors were varactors whose capacity was changed by varying a dc voltage across them. As the value of the capacitors changed, the delay of the line changed. About the practical limit of delay variation that could be obtained was ±0.5 microseconds. This was more than adequate, however, for quadruplex recorders. Off-tape sync pulses were compared to reference sync pulses and an error voltage created proportional to the time difference between them. This voltage was applied to the voltage variable delay line just described.

The time base stability at the output of such a device was about ±30 nanoseconds, still not good enough for direct color recovery. It has been determined that about 5 degrees of phase error in the 3.58 MHz color subcarrier is just detectable to the eye. Since the period of one cycle of the subcarrier is 280 nanoseconds, the time base error must be reduced below 3.9 nanoseconds. This was done by employing a second, shorter voltage variable delay line which was controlled by an error voltage derived by comparing off-tape burst to a reference 3.58 MHz color subcarrier (related of course to the synchronizing pulses used earlier.) The output signal from such a device was usually about ±2 nanoseconds and was limited by the signal-to-noise of the recovered burst signal off-tape.

The final step corrected for velocity errors. Because the sync and burst signals occured only once every line, head-to-tape velocity changes that took place during the line sweep time were still uncorrected. If a field of one full color were reproduced, these errors showed up as hue shifts from left to right on the screen. Velocity compensation consists of looking at what has happened to the time base errors over a period in the past and predicting what they will be in the future and then applying a correction throughout the scan line. Velocity errors do exist in helical recorders and must be dealt with.

Digital Time Base Correctors

So far only older analog time base error correction techniques have been discussed, but modern TBC's, as they are called, employ digital techniques. The first digital, wide range time base corrector was introduced by the ADDA Corporation in 1973 at the NAB show in Washington D.C. Ampex introduced its first digital TBC as part of its AVR-2 quadruplex recorder at the next NAB show in 1974. As one might expect, the first units were large and expensive. Present units are very small and reduced in price.

The first step in a digital TBC is to convert the analog video signal into a digital signal. The signal is sampled at a rate well above the Nyquist frequency for the highest video signal to be passed by the system. The first TBC's used a sampling frequency exactly 3 times the color subcarrier or 10.737 MHz. Each sample was then converted to an 8 bit binary word. Through the work of the SMPTE a standard has been proposed for composite NTSC color signals that involves a sampling frequency of four times the color subcarrier, or 14.32 MHz. Most but not all modern correctors now employ this higher sampling rate. A few use nine bits to define each sample. The first Analog-to-Digital converters (A/D converters) were made up of discrete components, but VLSI chips are now available to perform the same task. At the output of the TBC, the task must, of course, be reversed and the digital signal converted back to an analog one to be compatible with the rest of the TV plant. When the day of the all-digital studio arrives these A/D-D/A steps will not be necessary.

The digitized, off-tape video signal is clocked into a random access memory, time base errors and all, by a clock derived from off-tape sync and color burst. This same clock is used to derive the sampling frequency in the A/D process. A short time after the video signal is clocked into the memory and while it is still being clocked into a spot farther along in the memory, a second stable clock can be used to bring the signal out of memory at a constant rate and free of time base errors. This is an obviously oversimplified version of what happens, but it is in fact the heart of what happens along with certain embellishments to accommodate velocity errors and slow and fast speeds.

If a video signal is sampled and each sample defined by eight bits, the serial bit rate will be about 100 megabits per second. Circuitry to handle such a rate will be extremely expensive, so the digital signal is not processed in a serial fashion but is handled in parallel with maximum bit rates of either 3 × 3.58 MHz or 4 × 3.58 MHz which are quite reasonable bit rates to be handled by fairly conventional techniques.

Velocity errors are compensated for in the clocking-in process. Remembering that there is no continuous reference signal during a TV line and that high frequency time base errors may occur during the line it should be obvious that a clocking-in frequency derived from burst at the start of the line may clock errors into memory at the wrong time. As in analog TBCs discussed earlier, several previous bursts and horizontal sync pulses are examined and a prediction is made about what probably will happen during the line currently being clocked into memory. This prediction is used to modulate the timing of the clocking signal during the TV line.

If a TBC were to be used only to time base correct a signal from a VTR in normal play mode, the TBC would be complete as described; however, users expect a TBC to also provide full color-stable signals during reverse, stop, slow, normal, above normal speed (possibly 3 times play speed) and to provide recognizeable pictures during fast shuttle in both directions. Circuitry to accomplish these tasks becomes complex. Information must be sent to the TBC from the VTR that tells the TBC which direction the tape is moving and whether the tape is above or below normal play speed. Numerous clocking oscillators in the TBC must lock to the incoming off tape sync and burst through a wide range of frequencies and provide clocking frequencies for moving data into the memory.

Dropout compensation is also considered a part of the TBC's duty today; fortunately it is relatively easy in a digital TBC. Sensing of a dropout is easily accomplished by looking at the RF envelope recovered from tape. Once a drop out is sensed, information from a previous horizontal line of video information can be substituted for the missing information. Since digital TBCs store several lines of information, no particular problems are evident. Some TBCs have separate memories for storing two lines of information just for drop out compensation. While in some cases information from a line twice removed may not be suitable, most of the time it is, and DOCs based upon this principle are very effective.

Finally, TBCs must deal with playback signals from helical VTR's that use color-under techniques. The C and B format machines do not use such techniques, but often the TBCs from these machines may be used to stabilize the outputs of such color-under machines. Some such VTRs allow an unstable color subcarrier to be fed back to the VTR upon which the color information can be remodulated, but when this is not the case, the chrominance signal is demodulated in the TBC and remodulated upon a stable subcarrier after time base correction. A front panel switch usually allows normal or heterodyne (color-under) type of correction.

Control Systems

One has only to remove the trim panels from some of the older and less expensive audio and video recorders to encounter a collection of mechanical, sheet metal levers and springs and latches that at one time provided a cheaper but maddening way of controlling tape transports. When remote control of equipment was required, as was the case in most professional service, pushbuttons and relays were used. The early quadruplex recorders were full of relay logic, and the practice continued for many years until the advent of integrated circuits that provided inexpensive logic chips. The changeover occurred about 1970 when Ampex introduced the AVR-1 and continued until 1982 when microprocessors were introduced by at least two manufacturers. The technician of today must have a good understanding of microprocessors and of software. So far

Fig. 17. Early razor blade editing.

all of the software associated with control of VTRs is provided via PROMs (Programmable Read Only Memory.) The use of such techniques adds greatly to the complexity and problems of the repairman when something is wrong, but it allows sophistication of control not dreamed of a few years ago. The caveat is that technicians expected to service such equipment must be chosen with care.

VIDEO TAPE EDITING

The reader will find this subject covered in another chapter in this book, so the intent here is only to provide an overview and a bit of the history so the reader can appreciate the flexibility available today.

Historical

The first attempts made to edit video tape were made early in the development of quadruplex recorders and made use of a razor blade to cut physically two pieces of tape in a defined way and to place and join the two pieces with splicing tape. An edit pulse was added to the control track signal to mark which field (odd or even) was adjacent to the pulse and to guide the editor's razor blade. The tape was developed with a mix of carbonyl iron and a solvent that evaporated and left the iron particles attracted to the recorded areas. Fig. 17, a view of early razor blade editing, shows that it was a crude arrangement at best, and very soon a method of turning video heads off and on as well as a full tape width erase head were developed.

Next, a programmer was developed to control the electronic editor with its operation based upon counting edit pulse (or frames) and marking the edit points by placing cue tones on a low quality cue track. This was a great improvement, and the method served the industry well for a number of years. Ultimately, however, the need for a more sophisticated system of editing was needed, and it became obvious that every frame of a TV signal should have a unique identification if video tape was to be a flexible as film is with its edge numbers and sprocket holes. Several companies developed codes, and ultimately SMPTE again was asked to standardize the code. The result was the now familiar and (improperly identified) SMPTE Time Code that is properly identified as ANSI Standard V98.12.

The SMPTE Time Code

Each television frame is uniquely identified by an 80 bit digital address. A few of these bits are used for synchronizing purposes, a second group identifies each frame by hour, minute, second and frame. 32 bits are left over in groups of four that can be used as desired by the user. The distribution and use of the bits is shown in Fig. 18 which is a reproduction of ANSI Standard V98.12.

When first used the code was recorded on the longitudinal cue track on the quadruplex recorders. The pratice was carried over to C and B format helical recording, but it is now also inserted into the vertical interval. This practice is defined in SMPTE Recommended Practice RP-108.

Once a TV frame is uniquely identified, all things are possible. An exact frame can be looked for and found, two tapes can be run in synchronism with a defined offset in time code, audio and video recorders can be synchronized, and film and video transports can be locked together. The time code is video tape's sprocket hole.

Editors

Almost all modern VTRs have some sort of built-in editor that allows programming of a simple cut at a specified point defined. Shown in Fig. 19 is the editor section of the control panel of the Ampex VPR-80, the least expensive model in Ampex's product line.

When truly sophisticated editing is desired, however, and external controller must be used that will control not only the actions of the VTRs but also of switchers, special effects units, graphic generators and other parts of the video system. These units are covered elsewhere in this volume.

OTHER FORMATS

As with Editors, other significant formats will be covered elsewhere in this volume, but they will be mentioned here to give perspective to the position of professional formats such as the B and C formats.

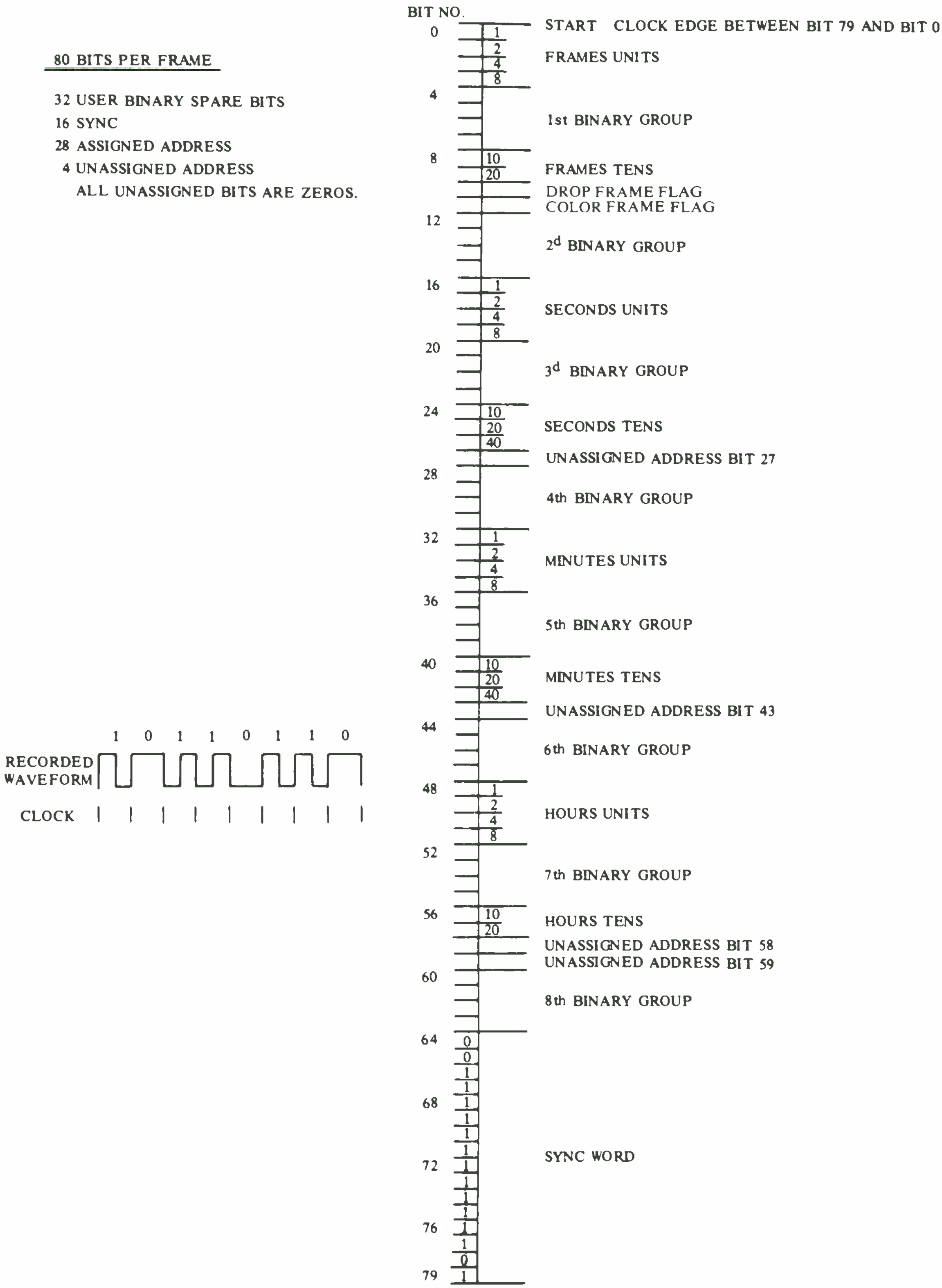


Fig. 18. Partial reproduction of ANSI Standard.

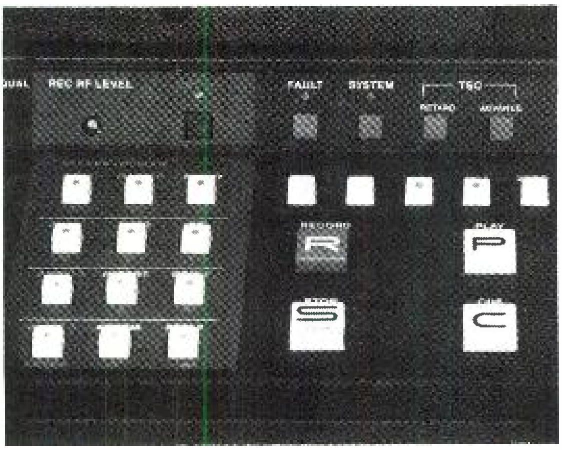


Fig. 19. Editor section of VPR-80 control panel.

During the evolution of helical recorder there have been dozens of formats, some intended for home use and some for professional use. The first of them to gain reasonable acceptance were the IVC 1″ format exemplified by the model 960 and the Ampex models VR-1500 and VR-660. Sony corporation as well as a number of other manufacturers produced a number of formats too numerous to mention here. It was with the U-Matic format that Sony found the magic formula, and this format is still in wide use today.

It is unfortunate that a major split came when a recorder was designed for the home market. Today we have both the Beta and the VHS formats, and the unsophisticated user is subjected to a number of conflicting claims through advertising. Both formats have been adapted for professional use, and at this point it is not clear which, if any, will be successful in this role. Many of these Beta and VHS machines are in use in the professional world for off-line use in reviewing material to be edited or for having a quick-look at dailies or programs being offered for sale.

At this time 8mm and 1/4″ formats are on the horizon as possible replacements for the Beta and VHS formats for both home and professional use, but it is too early at this time to predict which, if any, will survive. Digital and possibly component recording are on the horizon, and it is certain that digital recording will be a reality soon, but at this writing it is impossible to make predictions about what the format will be.

MAGNETIC DISC RECORDERS

Historical

There were a number of experiments with the use of discs for the recording of TV frames, particularly in West Germany, in the early sixties, but the first two practical uses did not occur until much later. Memorex developed an editing system, the CMX-600, in 1970 that used computer disc drives and packs to store monochrome television signals in short segments. The result was an editing system that resembled film editing. Unfortunately, as good as it was, it was ahead of its time. Analog recording was used, and the resolution of the reproduced signal was poor and was monochrome. It also was not very cost effective since a number of relatively expensive disc drives was used, thus making the system very expensive.

Ampex used two hard discs with four flying heads that could be moved radially in and out on the discs at variable rates to achieve the first practical slow motion recorder. This was the HS-100 Instant Replay recorder that revolutionized the telecasting of sports. It also was an analog device but was able to reproduce color directly by dint of violating the rules of the computer world where heads were flown over hard discs at an altitude of about 50 microinches and flying the HS-100 heads at about 12 microinches. The low altitude made the system very susceptible to dirt and contamination, and the units were touchy and difficult to maintain. They quickly became indispensible to the broadcaster, however, so their crankiness was tolerated. About 30 seconds of action could be recorded on the four disc sides.

Today, discs are widely used for the storage of still images and are known generically as Still Stores. Several manufacturers supply such units, and Fig. 20 illustrates one of the models available. Conceived of originally as a simple store for still images and a slide replacement system, it soon became obvious that they could be used as a creative production tool as well. The units offered today allow the creation of montages from images stored on the discs, and at least one has built-in titling capability as well.

Fig. 20. ESS 3 Electronic Still Store.

5.6

Radio Studio Audio

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In spite of what sales managers and others may say, the hub of activity in any radio station is the studio or studio-control room. The quality of the station's sound, its only product, is determined largely by what is going on in the studio. Even the best air talent, when burdened with inadequate equipment, will produce only marginal results. The point is that the right equipment, properly configured in the studio, translates to AQH numbers and rate cards.

TYPICAL STUDIO LAYOUTS

No two studios are exactly alike. Each design is affected by such factors as programming needs, available space and the ideas of the current and previous engineers, not necessarily in that order. Some stations, particularly older ones and those in very active markets, have gone through many changes as formats, managers, engineers and equipment have changed. The variety of equipment available and the number of ways all this equipment may be connected means that the number of studio designs is, if not infinite, certainly very large.

It is, nonetheless, possible to catalog most radio studios under one of a few general categories, depending on whether the emphasis is on music and other pre-recorded program material or on live material such as news, sports and telephone call-in.

The music-talk format is probably the most common, particularly in smaller stations. The basic setup includes an audio console, two turntables, and two or three cartridge machines arranged on a "U-shaped" desk. Assorted other equipment might include patching and transmitter monitoring and control, telephones and so on. This is in response to the programming needs which are fairly simple. The combo announcer-engineer on duty plays records, talks and reads announcements, plays commercials on carts, and switches in hourly the satellite news feed. Announcements prerecorded by the station may be produced in this same studio after signoff or in a second studio.

A variation on this theme is found in more competitive markets where records are transferred to cartridge before being aired. A separate studio is usually used for this as well as for production of spots. In this case, the air studio may have four, five or more playback cartridge machines and no turntables. The production studio has the turntables and record-playback cartridge machines, its own mixer console and an assortment of equipment to alter the characteristics of the recorded sound during transfer: multi-band frequency response adjustment, reverbation and so on. The production studio is also usually arranged so it can go on the air as a backup.

In a station where live or fairly immediate materials, such as news, sports and call-in, are a major part of the programming schedule, the sys-

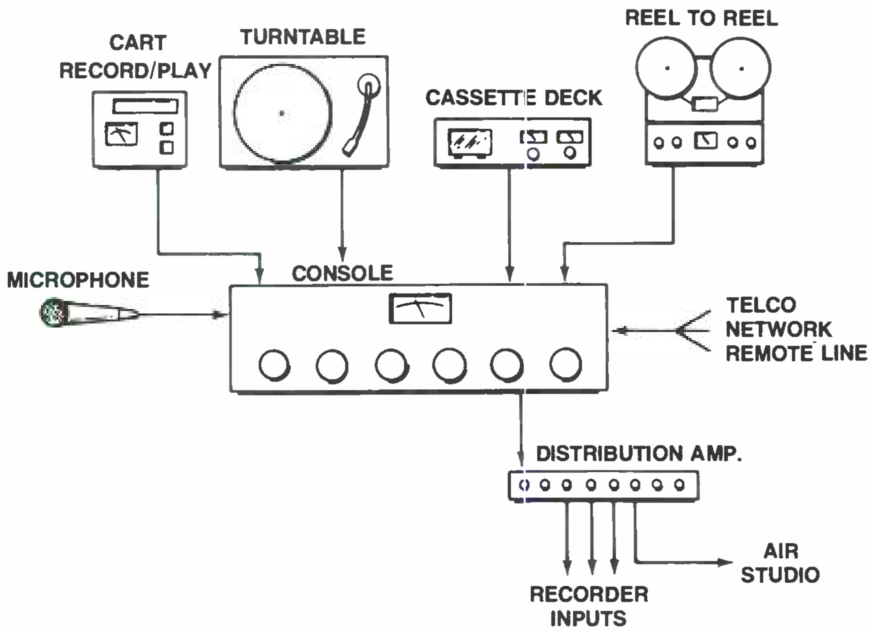


Fig. 1. Typical small production/news studio.

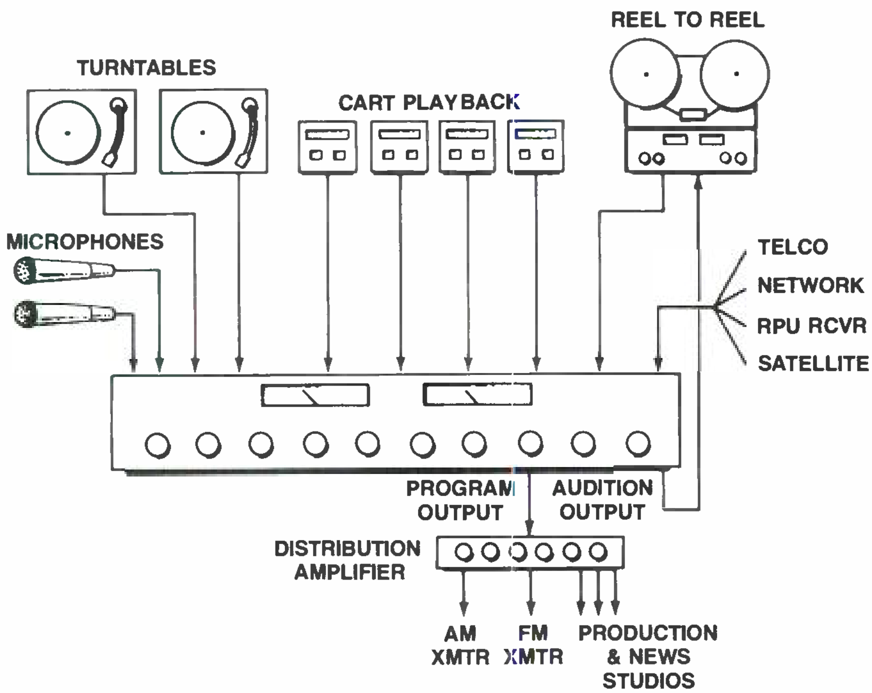


Fig. 2. Typical on-air studio.

tem must support some combination of live studio announcements, prerecorded announcements, various types of local, and network feeds. It may be necessary at times to operate with separate announcers and engineers due to the tight timing and fast switching between numerous sources. Here the studio must be arranged to accommodate several people at the same time. Alternatively, the studio and control room are separated, and there may be several studios and announcement booths controlled from one control room. Here, the control room can serve as a master control for announcement booths with small mixers and, perhaps, a playback cartridge machine. All these variations impose different requirements on the audio console and its internal and external switching and patching capabilities.

THE AUDIO CONSOLE

Centered in the control room, just in front of a "disc jockey" and the corresponding worn spot in the carpet, will be the audio console. This can be anything from a pristine BMX to a twenty year old RCA, but there will be at least one, of some nature, in every station. The smaller four to six mixer consoles are typically found in news rooms and production studios and on the air in smaller stations. The number of available mixing channels limits the number of audio events which can occur simultaneously or in rapid succession, and this can be a problem when a small unit is used as the main air console to save money. Also, many of the smaller units will not have an audition buss, the lack of which can severely limit versatility. One important feature to look for in a console of this size is a greater number of switchable inputs for each mixer; otherwise, the money saved in buying a small console will be spent on the patch panels necessary to get all of the inputs to the mixers.

As part of the planning process for building a new facility, buying a new console or just moving things around, it would be advisable to list all possible sources and assign some sort of priority depending on how often or how quickly they would have to be accessible to the console operator. This rating would help evaluate the number of mixing channels and switched inputs the console should have and what should be on a control room patch panel or master control patch panel or routing switcher.

The console found in most stations has eight to ten mixers. These will generally be of the rotary type, but slide attenuators are rapidly gaining popularity. For heavy on-air use, step attenuators are generally preferred as they require only occasional cleaning to maintain silent operation. Attenuators which depend on a contact sliding over the resistive element (usually carbon) are subject to wear and a build-up of dirt. Wear changes the element's resistance and the build-up of dirt (including the worn-off carbon) will cause erratic changes in the contact between slider and element. The results are uneven operation and noise. In step attenuators, attenuation is by a series of fixed resistors permanently wired to metal contacts over which a slider moves. The steps are in small increments and the slider makes the next contact before it breaks the last contact, resulting in smooth, quiet, and stable operation.

	Direct	Switched	Patched
Microphones			
Control Room			
Studio A			
Studio B			
News booth			
Other & Spare			
Total			
Recorded Program			
Turntable			
Compact Disc			
Cartridge Player			
Cassette Player			
Reel-to-Reel			
Automation			
Other & Spare			
Total			
Live Sources			
News Wire			
Satellite			
Telephone			
Telco Loop			
Production CR			
CB Monitor			
Studio			
RPU Receiver			
Other & Spare			
Total			
Special			
Filter			
Reverberation			
Oscillator			
Other & Spare			
Total			

The trade-offs are generally in initial cost, size and maintenance. The continuous carbon potentiometer is typically less expensive, smaller, and requires more frequent cleaning and replacement. On the other hand, a good quality unit, not the

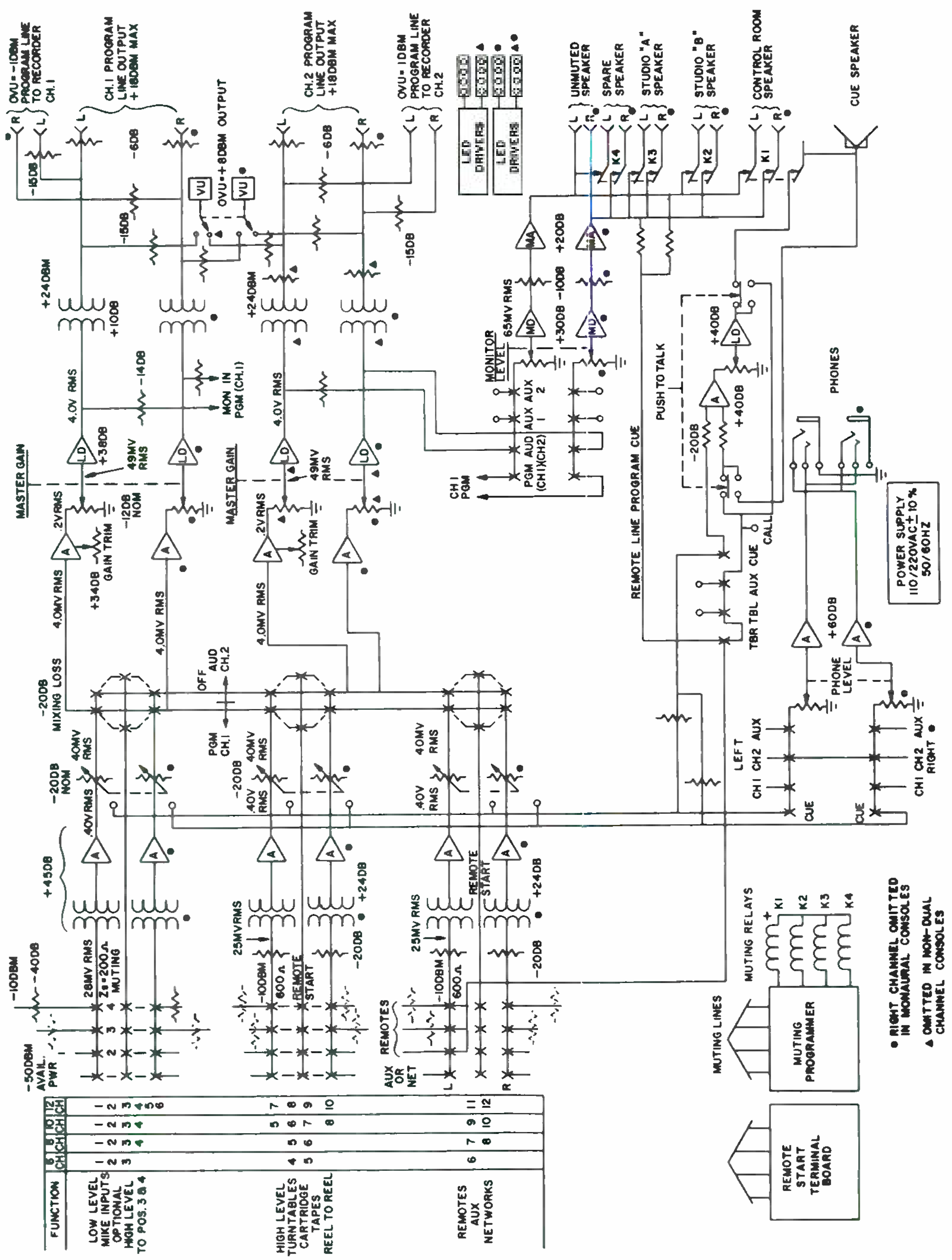


Fig. 3. Functional block diagram of AMPRO LC Series Console. (Courtesy of Television Technology Corp.)

"volume control replacement" type, can be expected to give reliable service for a long time. It is well-suited to small, portable remote units.

The step attenuator has a potential operational drawback when used by someone not trained in proper board operation. If the operator sets the level too high on either the master control or the output of some source such as a cart player, the mixer will bring the source up in only a few steps. When used this way, the jump in level between steps becomes noticeable and riding gain is difficult.

A relatively recent alternative is the use of voltage controlled amplifiers (VCAs). With this system, the audio signal does not pass through thetiometer but, rather, through the VCA. The gain of the VCA is, in turn, controlled by a voltage which is adjusted by the potentiometer. The audio is isolated from potentiometer noise. In the worst case, level control may become erratic, but listeners will not be subjected to scratchy audio. As with any amplifier, however, the VCA will introduce some thermal noise and distortion into the signal; how much is often a function of price. A quality console should have an overall distortion figure of 0.05% or less.

Fig. 4. Microtrak 6510 Console.

For most air work, an audition or cue channel is necessary to allow the operator to cue music, set up remotes, or listen for cues from network feeds. Even if the station has it's music library on cart, a cue circuit is an absolute must for previewing program material for use on the air and receiving cue signals from the news network and programs off "The Bird".

In an air studio the announcer's microphone is generally put on mixer number one. The turntables would be on mixers two and three. Cart playback would fall on four, five, six and seven. Eight, nine and ten would be assigned to reel to reel playback and the variety of remote feeds that the station would have, EBS, news network, remote line, etc. Autogram, Broadcast Electronics, LPB, Howe Audio and several other manufacturers have good medium size audio consoles.

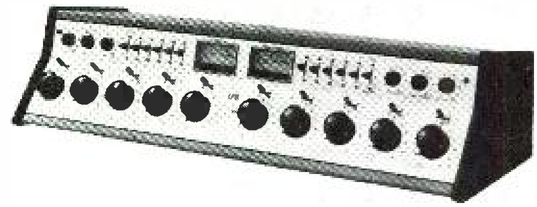


Fig. 5. LPB S-20 Console.

Console Features and Options

In medium and large market stations the consoles are generally ten channels or more and mostly the slider attenuator type. These larger units give the operator the luxury of not having to do any switching, as each audio source is assigned it's own mixer. They will also have a larger number of options available which make them more "user friendly" and at the same time "idiot proof". Clocks and timers are available on most consoles in this class, and are even standard equipment on some models such as the Radio Systems ESA 10. Monitor amplifiers in these are mostly of higher power than in the smaller consoles. Some manufacturers, as Howe in their 7500 series, treat the monitor amplifiers as an option. They assume that the station might want to use more than the eight watts found in some consoles to drive their JBL 4411's and opt for a Crown or other more powerful amplifier mounted in the equipment rack. One feature which should be added is a mono/stereo switch on the monitor amplifier to allow the operator to check for out-of-phase program material. This is easily added by the engineer with a switch at the monitor gain control.

A well designed console will have interchangeable input cards so that input sources can be moved to different mixers or changed later to allow for modifications in the studio or format changes. Remote starts for cart machines and turntables is a fact of life on the top-of-the-line models. This allows the operator to start the equipment by turning on the appropriate channel on the console and in some cases the equipment can be stopped when the channel is turned off, allowing for almost automatic operation using logic circuits. One important feature to look for with remote starts is switching of the start function with the switching of the input selector. This eliminates the annoyance of having a cart machine start everytime another piece of equipment wired to the same mixer is used.

The more expensive consoles offer a choice of standard analog VU meters or an LED "Light Show" in at least two colors which gives a more accurate tracking of audio peaks. Some manufacturers claim that this latter type of meter reduces the number of times the operator runs the board with the meters "buried in the red" but there seems to be no solid support for this, at present. There are several models of LED VU meters available for retrofit into consoles which do not offer this as an option. Some models replace existing VU meters in the console and at least one model show peaks and RMS values on both left and right channels on the same display.

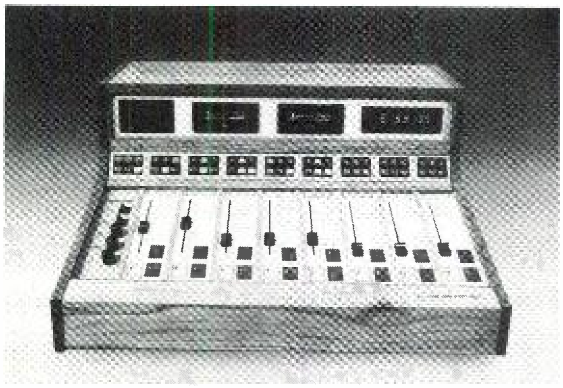


Fig. 6. Howe Audio Series 9000 Console.

Planning and Wiring

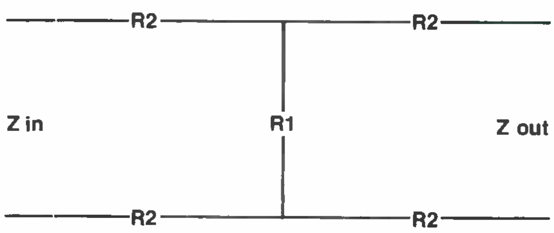
With the coming of multitrack recorders in broadcast production, there has come a need for consoles with more than single left and right outputs. Those consoles in the medium and large class can provide four output channels using the Program and Audition outputs to feed the inputs of a four track recorder. For eight tracks or more, some patching will be necessary or the station will have to consider a recording studio type console. This will give the necessary outputs, individual equalization on each channel, pan pots and a ration of other features for added versatility. There is, however, generally a trade-off on other features as the console was not designed for radio station use; and the cost is very much higher.

Before the new console arrives, the planning for its wiring should have been completed. This is something that cannot be trusted to trial and error because one the wiring is in place and the studio is being used those corrections that will be made "later" never seem to get done until a serious problem arises. The first step is to determine which audio source will be assigned to the various mixer inputs and the wiring for the patch panel and console documented so that trouble shooting and changes made later will be easier. In most cases it is best to run all the wiring, except the microphone inputs, through a patch panel to increase the versatility of the system. This is particularly important in the case of smaller consoles and in production studios to allow equipment to be brought in for special effects and emergency situations. Always bring the output of the production studio console up on the patch panel in the air studio so in case of the failure of the main console the production console can be patched into the audio chain and the station can go on the air from the production room.

Part of the planning for a new console is determining that the levels from all of the audio sources are compatible with the input levels required by the console. If this is not done the operating positions of the mixers will be inconsistent, making it difficult for the operator to run the board properly. In extreme cases you may not be able to open the pot more than a small fraction of a turn before driving the meters to the pin, or you may not be able to get enough gain from the channel even with the control fully open. Neither situation is acceptable and in most cases results in poor signal to noise figures. All studio sources should be set to their normal output operating levels, and H pads should be installed as needed between the source and the console input. In many cases these can be mounted inside the equipment. Pads of 10 dB or greater value are also useful in correcting impedance mismatches. The accompanying chart gives resistor values for 600 ohm H pads.

Patch panels, or jack fields, come in two basic types. The tip and sleeve which consists of one conductor and one shield, and the tip, ring and

Loss (In DB)	R1 (Ohms)	R2 (Ohms)
3	1.8K	51
6	820	100
10	430	160
15	220	200
20	120	240
30	39	270
40	12	300



H-PAD

Z in = Z out = 600 Ohms

Fig. 7. Typical H Pad values.

sleeve type which consists of two conductors and a shield. The tip and sleeve dates back to the days of the first jack field and is mostly obsolete. With this system it would be necessary to use four individual cords to patch one piece of stereo equipment. The most practical system is the tip, ring and sleeve jack which allows stereo connections to be made with one double plug cable, mono systems would use a single plug cord. The engineer has the choice of wiring the jack field and its terminal block or buying one that has been prewired, easily saving about two day's work and a soldering iron tip. The prewired patch panels are available with several types of terminal blocks, the old "Christmas Tree" solder terminals, wire wraps, telco punch blocks and a new split cylinder terminal developed by ADC that operated on a punch-down principle. This split cylinder design was developed for broadcasters and the voltage levels found in audio systems rather than for the 48 volts that the #66 telco blocks were designed to carry.

Another option available with prewired patch panels is that of having the "normals" brought out to the terminal. Normals are jumpers which connect the circuit of the upper jack to the one below it in the jack field when no cords are plugged into either of the jacks. This jumper circuit is broken and the audio transferrred to the patch cord when it is plugged into a jack. In a typical application the console output is wired to the upper jack and the input of the next piece of equipment, usually a limiter, is wired to the jack below. With no patch cord plugged in, the audio from the console is normally connected to the input of the limiter; thus the term "normals". However, sometimes this feature is not wanted, and having the normal jumpers at the terminal block makes it easy to change.

Stealing a little of the patch panel's thunder is the audio routing switcher. This unit replaces the jack field in that it accomplishes the same function, except with a routing switcher the audio is moved around by a series of mechanical or solid-state switches rather than by cords and jacks. The system is useful because it eliminates looking for patch cords in an emergency and with some units audio can be routed to more than one feed at a time. These are, however, active devices subject to component failures and they do add an amount of noise and distortion to the audio chain. The main reason why patch panels will be around for some time to come is economics; some routing switchers cost ten times as much as a prewired patch panel.

TURNTABLES

The most often used audio source in a radio station is the turntable. Until recently, at least, the most common broadcast turntable has been the idler drive type which uses a motor that turns at 1800 RPM driving an idler wheel which then drives the platter. There are two variations of this system. With the rim drive turntables the idler wheel runs against the outer rim of the platter. These are cheaper to manufacture as a smaller motor can be used to drive the heavy platter. In the better system, the idler wheel drives a hub near the center of the platter. This design minimizes wow and flutter caused by fluctuations in the motor speed. Also, vibration generated by the motor is not transmitted across the entire width of the platter as would be the case if it were driven from the rim. This type of drive system requires a large, powerful motor for the rapid starts necessary in broadcast operation.

The idler wheel type of turntables require periodic adjustment to insure that the idler is positioned properly with respect to the multi-diameter motor shaft. The idler wheel hardens with age and if left engaged while the motor is stopped, will develop flat spots. If either condition exists the turntable will be noisy and the idler will have to be replaced.

Finding increasing use in the broadcast station is the direct drive type of consumer turntable, in which the platter is the rotor of the motor. These turntables all but eliminate wow and flutter and make speed control more accurate. Their speed control circuits constantly monitor speed and make adjustments, keeping it more accurate than if it were left to depend on line voltage and frequency. This makes the practice of "speed enhancing" a science by giving the operator the ability to dial up an exact speed. By eliminating the idler wheel there are no adjustments to make and maintenance is eliminated. Repairs are much more difficult because of the complex circuitry in these models, and with some models, no schematics or service information is supplied to the purchaser. Remote starts are also difficult with some models and require additional outboard circuits to accomplish this function. The situation may change, however, as at least one manufacturer is building a direct drive turntable for broadcasters.

Interfacing directly with the turntable is the tone arm. There seem to be relatively few true broadcast tone arms as most are designed for consumer use. These arms track quite well and are easily adjusted but are not designed for the constant use that they receive in a broadcast studio. The easy adjustments are often misadjusted by well-meaning operators, especially on weekends. The ideal broadcast arm is set up once for its cartridge and then left alone, unless someone steps on it. It has happened!

When installing a tone arm, use the template from the turntable manufacturer and follow the instructions supplied with the arm, otherwise the tracking error will be greater than optimum and even the best cartridge will not sound good. The manufacturer's recommended tracking weight for the cartridge and stylus being used should be strictly adhered to even though it may take some effort in getting the proper amount of weights in place. This job should not be rushed; a tone arm is a precision device.

The choice of a phone cartridge depends on the level of quality that you want to obtain. The less expensive models are generally rugged and give the greatest life in on-the-air use, but better separation and high frequency response is gained by moving up to more expensive, though generally less rugged, models. The majority of cartridges are designed for the consumer market and will not stand up to continuous use and back cueing that they will be subject to in a radio station. The stylus needs to be cleaned and inspected daily and replaced when it shows wear or if a defect is suspected. The engineering department needs a supply stashed away so regular replacement is not a problem.

The next element in the turntable "system" is the preamplifier. This item, like the tone arm, is often not given proper attention. It is easy to buy the cheapest one, since it will never be seen again once it is bolted into place and levels are set. The important specifications to look at are the noise and distortion figures. This task is not as easy as it sounds as the many manufacturers express these figures in as many ways. Noise will be referenced to different output levels and may be stated in decibels or RMS voltage.

The more expensive models offer features including filters for warp and resonance, high frequency cut and boost, and adjustable cartridge loading. These are nice to have, but should be kept out of reach of tinkering programmers. The best designed preamps put the RIAA equalization circuit in the second stage to eliminate non-linear loading of the cartridge by interaction with the filter components. The most important concern for the engineer is the preamp's resistance to RF. This is a high gain amplifier and one of the first places that RF interference will make itself known in a studio. Some units are designed to operate in an RF environment, others are not; take this into consideration when you buy. The general rule of thumb, you get what you pay for, applies to preamps. With it being the first, and sometimes the weakest link, in a station's audio chain this is not a place to economize. A poor preamp can kill a station's air sound, as the only thing that goes through a preamp is *all* of your music.

Although not a conventional turntable, the compact disc player is serving the same purpose in many radio stations. Its use began in classical music stations but spread to others as record companies issued a wider variety of titles on compact disc. The discs, themselves, are one-sided, 12 centimeters in diameter and will reproduce up to 75 minutes of program material.

Sound is encoded as digital "bits" which are recorded as minute, etched marks on the surface of the disc. These bits are read, in the player, by focusing a light beam, from a semiconductor laser, on the disc. There is no stylus contacting the disc as with conventional recordings so there is no wear. A transparent plastic coating protects the surface so that only the most severe accumulations of dirt or scratches will affect playback quality. In general, compact discs are much less susceptible to mistreatment than are conventional recordings.

In the recording process, audio is sampled at brief, regular intervals. An analog to digital converter measures the voltage at each sampling point on the waveform and generates a digital code representing that voltage. These codes are recorded; and, with appropriate digital equipment, may be used to mix and manipulate the sounds they represent as has been done with conventional analog recording.

In playback, the codes are used to drive a digital to analog converter which reconstructs the original audio waveform by generating a series of voltage steps. These steps are smoothed out by filtering.

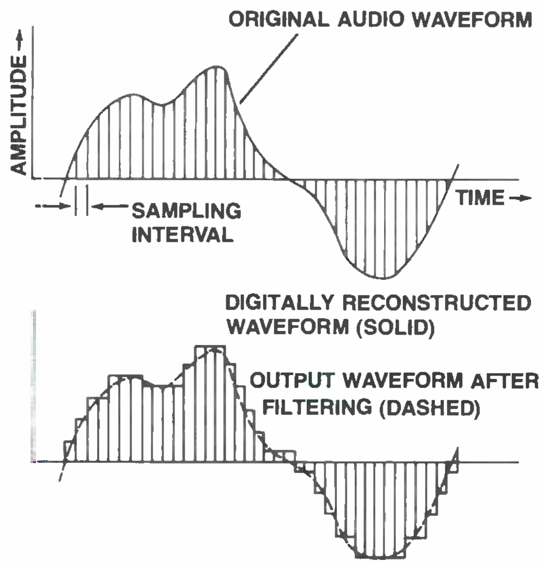


Fig. 8. Digital recording and reproduction of analog signal.

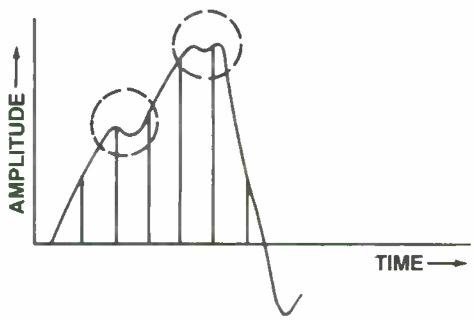


Fig. 9. Errors resulting from too low a sampling rate.

Although much of the noise and distortion of conventional tape and disc recording is eliminated by digital techniques, there two sources of errors which affect digital recordings. Both are related to the sampling rate, and in both cases, a higher sampling rate is preferable. Since the amount of information any disc (or tape) can hold is limited by the nature of the medium, the trade-off is between noise and recording time. As of this writing, most master recording is done with sampling rates in the range of 44 kHz to 50 kHz while playback discs use 44.1 kHz.

If the audio signal is changing very rapidly with regard to the sampling rate, it is possible for the sampling device to miss or inaccurately measure certain changes. The reconstructed waveform will reflect these inaccuracies. As an extreme example of inaccuracy, consider a 10 kHz frequency being sampled 10,000 times per second; the apparent result would be a continuous, dc voltage.

Upper harmonics may be missed or, in being reproduced inaccurately, they may appear in the output as new components not harmonically related to the original sound. This is called *aliasing*. To reduce this source of noise, sampling should be done at a rate at least twice the frequency of the highest harmonic to be reproduced accurately. In addition, a low-pass filter should cut off system input above the highest desired harmonic.

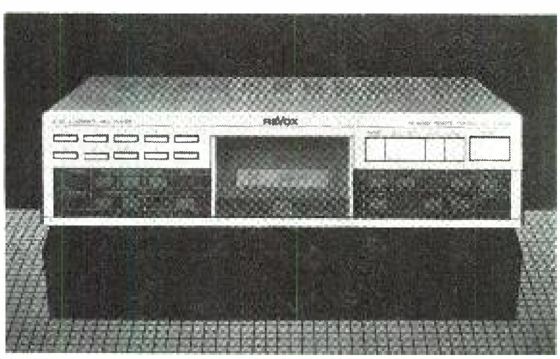


Fig. 10. REVOX B225 Compact Disc Player.

Quantizing error is another source of distortion, which also appears as noise in the output. The analog to digital converter samples voltages at discrete levels. If the waveform voltage at a given sampling point falls midway between two levels, it is uncertain which will be selected by the circuit. This means that, at any point in the reconstructed waveform, there is a possible error of one voltage step. The result is random distortion which may be masked by the introduction of a small amount of random noise.

MICROPHONES

The choice of a microphone for use in a broadcast studio is unfortunately often left to chance, and quite often in a smaller market station it turns out to be whatever is lying around. The choice is not always an easy task as there are a vast number from which to choose from many good manufacturers. For the radio studio we can disregard the lavalier types. This is also true of the wireless systems, except for use on remotes and events outside the station. Again, shotgun mikes are mainly the property of the news people and found on TV sets. This narrows the choice to the general purpose studio microphones of which there are several choices.

The dynamic microphone is the most common type found in use today. Its construction consists of a coil of wire attached to the diaphragm, which moves in the field of a permanent magnet as the sound waves strike the diaphragm. A voltage is generated as the coil is set in motion. This design lends itself to the construction of a very durable microphone which will tolerate an amount of physical and acoustic abuse, such as rock and roll disc jockeys. The quality of this and any other type of microphone depends on the finesse exercised by the manufacturer in its design and construction. The more intricate the design and construction, the wider and more linear the frequency response. In the case of directional microphones, this results in a more consistent pattern and smoother off-axis response.

At one time, having the very best studio microphone meant having a ribbon mike. The ribbon microphone is constructed using a very thin corrugated ribbon of foil positioned in the field of a large powerful magnet. A voltage is generated across the ribbon as it moved with the sound waves striking it. Since the ribbon was very thin and light, this type of microphone is very sensitive and picked up many of the subtle sounds missed by a dynamic mike. Unfortunately because of the delicate nature of the ribbon, this type of mike is easily damaged by shock or a blast of air. Some models, such as the RCA 44BX and BK11, are rather large and heavy due to the size of the magnet required. With the improvements made in dynamic mikes and the introduction of condenser microphones, the ribbon mikes are waning in popularity.

Taking over from the ribbon microphones are the condenser mikes, originally made famous by Neumann. Their construction consists of two plates, one fixed and the other moveable, forming a variable capacitor. As sound strikes the moveable plate, the distance between them changes. Since the plates are charged from a power supply, or battery, the movement of the one plate causes a change in the capacitance between them and resulting in a current flow from one plate to another. This current flow results in a voltage being generated across a resistor positioned electrically between the plates. A variation on this involves the use of a permanent electric field device, an electret, to provide the field. This eliminates a power supply circuit.

In another variation, the plates are used as part of an 8 MHz oscillator which swings in frequency as sound waves vary the capacitance between the plates. This results in an FM signal which is then demodulated to produce an audio output. This is known as an RF condenser microphone. Both of these microphone systems are relatively expensive and are found more in recording studios rather than in radio station control rooms.

The dynamic microphones, as well as the other types, can be constructed to exhibit directional characteristics. This design is best for a radio studio as it eliminates unwanted noise from equipment or people other than the announcer that may be in the room. The "cardioid" pattern is the most common which results in a heart-shaped pickup pattern in front of the microphone. In a dynamic mike, this is usually done with an acoustical phase-shift network which feeds sound waves from the rear of the microphone to the back of the diaphragm out of phase with the same sound arriving at the front so that they cancel out. A ribbon mike, by its construction, is bidirectional, as sounds from the sides, striking the edge of the ribbon, are not picked up. It can be made unidirectional by blocking the path of sound to the rear of the ribbon. Condenser microphones can be made directional by using more than one element in the microphone case and electrically combining the outputs to cancel voltages generated by sounds from one side of the microphone.

One drawback to using a directional microphone is that the off-axis pickup is not linear over the audio spectrum and the user (speaker, singer) needs to be positioned in front of the microphone for the smoothest response. This is easy enough in a studio situation but with news interviews, passing a microphone back and forth, it is not so simple. Also, directional mikes will not pick up a lot of the background sounds desirable in doing news coverage. Omnidirectional types are best suited for remote news and sports, except when noisy crowds can keep the announcer from being heard, and in soundproof studios. Directional types also suffer from what is known as "proximity effect". The low frequency sensitivity increases as the sound source gets closer to the mike. Many announcers use this to make their voices sound deeper on the air by working the mike close. Most directional microphones have adjustable bass roll-off circuits to compensate for the boost in low frequency response.

If it has not been done in recent memory, microphone wiring should be checked to assure that all microphones and connectors are wired uniformly. Since microphones, cables, mixers and accessories are often acquired by a station at different times from different sources and repaired by different people, a systematic check of all connections is a good way to bring all equipment into conformity with some standard.

Microphone polarity is defined in the Electronic Industries Association (EIA) Standard RS-221-A in terms of the microphone having terminals connected so the voltage is "in-phase" or "out-of-phase" with the sound wave producing that voltage. There are devices available for testing microphone polarity by producing a sound pulse; however there is a practical method which also gives good results. It will not identify the phase relation between sound and electrical waves but will permit uniform phasing of connections.

When using two or more microphones in the same studio, care must be taken to have all mikes phased alike. To check this, open the mixer channel for the first mike and adjust the attenuator for the proper level when the microphone is spoken into. Move the second microphone next to the first and adjust its attenuator to a similar setting as the first. When speaking into both mikes the overall level on the VU meter should stay the same or rise slightly. If the level falls, the microphones are out of phase as their outputs are cancelling each other out. Reverse the leads on one of the mike connectors.

For microphones having connectors meeting EIA Standard RS-297, the connections are as follows:

Out-of-Phase Terminal	3
In-Phase Terminal	2
Ground	1

Other connectors should have the in-phase terminal marked with a red dot or stripe. Cable connections are red (or color other than black) for in-phase and black for out-of-phase.

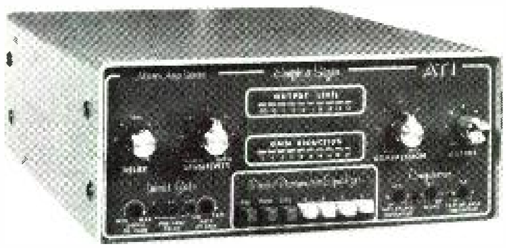


Fig. 11. The ATI Emph' a Sizer provides compression, equalization and gating for enhancing microphone performance.

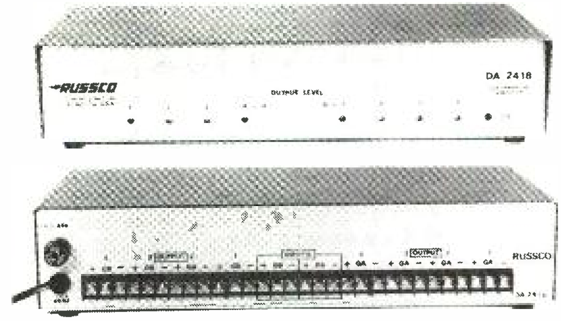


Fig. 12. RUSSCO DA 2418 Distribution Amplifier.

Microphone Processing

The addition of compression and equalization to the announcer's mike can make his other voice more compatible with the processed program material played on the air. The result is a more consistent air sound when the transition from program material to live announcer is made. Some consoles offer this processing as a built-in option, others provide jumpers between mike preamps and the mixing buss of that external equipment can be easily wired into the circuit.

Many stations featuring morning teams on the air will "pan" each person's mike slightly to one channel augmenting the interchange between the two announcers with a slight stereo effect. This should be limited to about 3 decibels of shift or the artificial effect may be objectional to some listeners. *AM-FM Broadcasting* by Harold Ennes contains more information on microphone use and placement in studio situations and should be consulted for unusual situations.

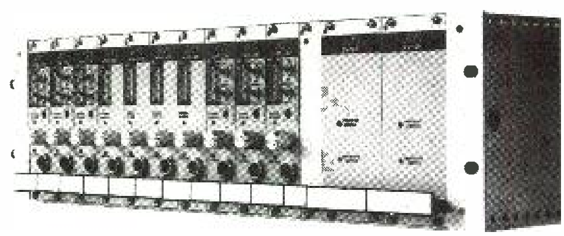


Fig. 13. ATI DA 10000 Distribution Amplifier.

STUDIO ODDS AND ENDS

The job of a distribution amplifier is to provide audio from a single source, such as a console output, to a number of inputs. This cuts down on time-consuming patching and switching in the production room. In the air studio it routes the console output into dual audio processing chains, one for AM and the second for FM in simulcast situations, and into recorders or monitors. The majority of the DA's in use are the straight forward type which are simply amplifiers with multiple outputs.

Output Choices

The main choice to be made is whether to have individual level controls for each output and the number of outputs. Several manufacturers have begun combining other features into the distribution amp which can cut down on the amount of equipment required in a studio. Options may include metering, compression, loss of signal alarms, and redundant power supplies. This system would find use in a large production facility in either radio or television stations.

Equalization

Equalizers are available in two basic types. The best known is the graphic in which the audio spectrum is divided into a series of bands represented by a group of slider controls on the front panel that the operator adjusts up or down to vary the level in that particular band. This is the easiest to use as the controls give a picture of the approximate frequency response that is being created. The parametric equalizer is more versatile in the hands of a trained operator. With a parametric equalizer a specific frequency can be dialed in and the amount of cut or boost can be set. The parametric also allows adjustment of the selectivity of the change being made. The operator can "notch out" a hum or buzz or give a gentle sloping boost to a band of frequencies to make them stand out. Equalizers can "brighten" a station's air sound and help in doing creative production.

Noise reduction systems can help reduce tape noise for stations having the majority of their programming on tape. With the systems made by Dolby or DBX the audio is encoded, recorded on tape, and then decoded to reproduce the original audio with a lower level of noise than if the system had not been used. The companding type of noise reduction is also finding some use on auxiliary mediums such as on unprepared phone lines and SCA service. These systems cannot remove noise that already exists on the source material. There are other systems which attempt to reduce residual noise that exists on recorded material. These are basically gating devices which turn off all audio when it falls below a certain level. The better of these devices are frequency selective and target their action to the frequency range associated with most tape hiss. The theory of operation being that any audio below a fixed level must be noise and should be ignored. All of these noise reduction systems have some merit, but in professional broadcasting it might be more practical to spend the additional money on better equipment and tape that does not need help. These systems, for the most part, are designed more as consumer electronics.

AUDIO PRODUCTION

The most challenging use of equipment and talent in broadcasting is that of production. Whether it be a thirty second spot or a series of programs, it demands the most from the station's resources. On a small scale this may mean reading copy over a music background and getting both to end at the same time. One of the common problems is the agency tape which was produced on a machine aligned to no standard known to man. The simplest fix to this situation, other than calling the station engineer every time, is to extend the azimuth adjustment on the playback head out to a knob which can be tweaked by production people as necessary for the tape being used. With this adjustment in place all that is required is to "tune" in the playback alignment. Often tapes from outside the station will need "brightening up" with an equalizer in order to keep them consistent with the quality of the station's air sound. Those with hum or noise can be saved with a parametric equalizer being used as a notch filter tuned to the objectional noise.

The use of multitrack reel to reel recorders makes adding tags or reading copy into "doughnut" tapes a lot easier than the old method of using two recorders. The music bed or agency tape is put on one track and the copy is overdubbed or recorded on another track or tracks. If a mistake is made, the voice track is simply re-recorded until perfection is reached. To do this the machine will have to have the selective synchronization feature. This switches the record head to the playback amplifier on the tracks that contain the music bed, otherwise the bed and the voice track would be separated by the amount of distance between the play head and the record head on the recorder making this type of use impossible. When all tracks are successfully recorded the finished product of music bed, voice track, and sound effects is created by mixing all tracks together onto a cart or another reel to reel.

Most professional reel to reel recorders have a variable speed option which takes those 64 second spots and fits them into a 60 second space of air time. If only a few seconds of adjustment are required this will not cause any noticeable effect on the audio quality, but as more adjustment is required a device such as an Eventide Harmonizer will be necessary to restore the normal pitch to the audio. Eventide also provides a system that automatically adjusts the speed of the deck and adjusts the pitch of the audio the required amount.

A stereo effect can be created using a mono music bed and a stereo equalizer. The mono source is fed to both channels of the graphic equalizer and a boost on one band is set up opposite a cut on the opposite channel so that some frequencies are predominant on the right channel and others on the left.

Phasing or flanging is created by some of the new electronic devices on the market or can be done by playing back the same material on two reel to reel decks and varying the speed of one deck slightly by dragging your finger on the flange of the supply reel to slow it down and by giving the take-up reel a push to speed it up. The resulting phase differences will result in the spacey effect needed to accent production for special effects.

Reverb and echo effects are generated by electronic reverb units constructed with springs or large metal plates which are set into motion with the audio and the delayed vibrations picked up and mixed back into the audio at the required level. For adding reverb to program material the amount mixed back in should be about twenty decibels down from the normal program level or

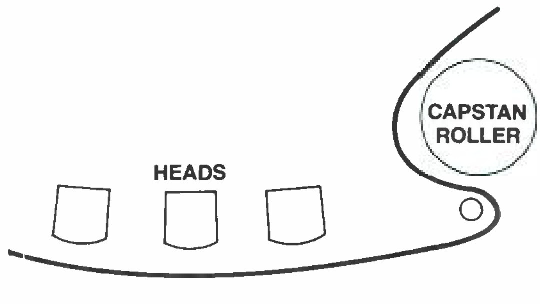


Fig. 14. Reverse tape path on an open reel recorder.

the effect will be distracting. If a compressor or limiter is used, the reverb should be mixed back in after the output of the limiter. This procedure will prevent the percentage of reverb in the program chain from varying with the operation of the limiter. Echo effects can also be added in production with a reel to reel recorder by simply opening the playback pot on the console slightly as a recording is being made. Reel to reel decks can be made to play backwards to create reverse echo and other special effects by simply threading the tape backwards around the capstan roller.

A good sound effects library is a definite asset to a production studio and there are several complete libraries on the market. Often an effect is necessary that has not been recorded, this is when the creative production man stands out in being able to create what he needs. How do you create a "Star Wars" light saber? By waving a hand-held bulk tape eraser in front of a Sennheiser MD 421, of course! But don't let the station engineer catch you doing this! The list of production tricks goes on and they were created by someone such as Jack Elliot in Los Angeles or Jim Stanley in Virginia Beach, who are interested in doing a better than average job in creating a spot. There is additional incentive in doing good production in that a good production man is worth money to station management. He can be the reason why advertisers buy the station, to get his services. Many people specializing in this field have gone on to opening their own studios selling their services to advertisers and radio stations.

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5.7

Magnetic Recording Media

Part I: Care and Handling of Magnetic Tape

Part II: Principles and Current State of the Art

Part I: Care and Handling of Magnetic Tape

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INTRODUCTION

Magnetic tape recording plays an important role in many information storage applications. Those people involved with the recording and storage of this type material should be aware of the conditions and procedures that will help ensure the best overall results. The purpose of this section is to give some of the guidelines to achieve good tape life and tape performance. Although this information is aimed at the video tape recording field, it is basically applicable to other magnetic recording areas.

BASIC INFORMATION

Magnetic recording has proven to be a very viable, versatile, and reliable form of storing information. The electromagnetic media used in recording tapes are capable of retaining the signal for indefinite periods of time. The recorded signals will remain virtually unchanged for decades unless they are intentionally erased. Those problems that have been encountered with magnetic recording are mostly physical in nature, due to improper handling or storage conditions. These problems can be caused by malfunction of machines, improper operator care, or poor environment in either the operations area or storage area.

The following information is intended to help in achieving optimum tape performances by hitting upon certain key points with as little rhetoric as possible to simplify the reader's future reference to parts of this chapter.

The Recording and Operations Area Environment

Foreign debris from the environment is one of the reasons for the cause of dropouts, scratches, machine parts wear, head life, and tape life. This warrants important consideration today, and is becoming more important as the recording density on tape increases.

1. The area in which tapes and machines are used should approach "clean room" conditions, such as those found in main frame computer tape installations. This degree of air filtration, though ideal, is not realistically achievable in many video tape operations. It is recommended that at a minimum the filtration used must at least meet the efficiency rating of 90 percent based on the National Bureau of Standards Dust Spot Efficiency Test—Atmospheric Dust. Qualified air conditioning and air filtration firms are aware of the requirements to satisfy this need. The airflow system should be designed to maintain a positive pressure in the recording area.

This prevents dust particles from floating into the room from other locations. In addition the duct work sizes and placement, leading to this area, should allow the necessary amount of air flow to maintain the proper temperature and humidity control, without having air velocities at a level high enough to blow settled dust and debris around the room.

2. The temperature should be controlled at approximately 70 °F plus or minus 5%, and the relative humidity at 50% plus or minus 20%. Without this control the risk of head clogging, stiction, and higher headwear are increased. The old cliche is still true "what is comfortable for humans is a good environment for the tape."
3. To avoid inadvertent contamination of both the tapes and machines, smoking, eating, and drinking should be restricted to certain areas and preferrably out of the recording locations.
4. Special attention must be given to the floors in or near the recording area, as they are a source of debris due to pedestrian traffic. Cement flooring must be sealed and tile floors should not be waxed. Both should be mopped clean on a routine basis. The use of indoor-outdoor type carpeting that contains anti-static material is acceptable and may be most desirable, depending on the needs of an operation. The carpeting helps reduce room noise from equipment, and affords an atmosphere conducive to good operator practices. The use of industrial vacuum cleaners is recommended on carpeting or floors, and it should be done on a regular basis. The exhaust from these units should be located outside the recording area.
5. To aid in the reduction of debris due to pedestrian traffic, the recording area should be located such that it is not a normal passageway to other parts of the operation. This not only improves the cleanliness of the area but avoids continual distraction of the tape machine operator's function.
6. During times of construction in any area of the facility, the frequency of cleaning should be increased substantially. The area under construction should be sealed off with plastic sheeting as best as possible. Surprising as it may seem, cement dust can get through filters that water will not.
7. The tops of equipment and other exposed surfaces should be cleaned on a periodic basis to maintain the integrity of the operations area. This procedure will also give clues as to how clean the envrionment is in the area. The more dust and debris found during this routine is an indication of the amount of dust and debris that has found its way onto the tapes and into the recording equipment.

Operating Practice Recommendations

The video tape operations in a facility are usually involved with more than one type of tape machine; namely reel to reel type and cassette type.

The differences from a tape life or reliability point of view are only the questions of who or what is handling the tape during thread up, how exposed is the tape when on or off the machine, and the tape path conditions. The cassette approach virtually eliminates the human handling of the tape, and reduces the exposure to atmospheric contamination; however, the mechanical threading mechanisms require attention as well as the condition of the tape path. The recommended environmental conditions apply to both type systems. Before proceeding, the next comments may help put things in a better perspective.

People involved in any video tape recording operation, at some point, may ask whether all the emphasis put on cleaning, handling, and environmental control really accomplishes anything; that is, in other terms, "familiarity breeds contempt." Fig. 1 gives a good picture of the relative sizes of what is being dealt with. Not only is this debris capable of causing large drop-outs in the area they are located on the tape; but some of these particles, when wound into a tape pack, may cause impressions into adjacent layers which in turn will result in additional drop-out activity.

Wallace's law on signal loss due to head to tape spacing theoretically indicates that compacted debris contained in a one pint jar, if equally distributed, would be sufficient to put every video tape ever produced out of manufacturers' drop-out specifications.

Good housekeeping and operation practices are very important to a successful video tape operation.

Following are some of the operational recommendations that apply to the handling of reel to reel and cassette type video tape recording formats.

1. The components in a video tape machine that contact either the front or back side of a tape, should be cleaned before each tape is threaded to make a recording or playback. The recommended cleaning fluids are Freon TF (a DuPont trademark), Genesolve D (an Allied Chemical Trademark), isopropol alcohol, or other cleaners recommended by

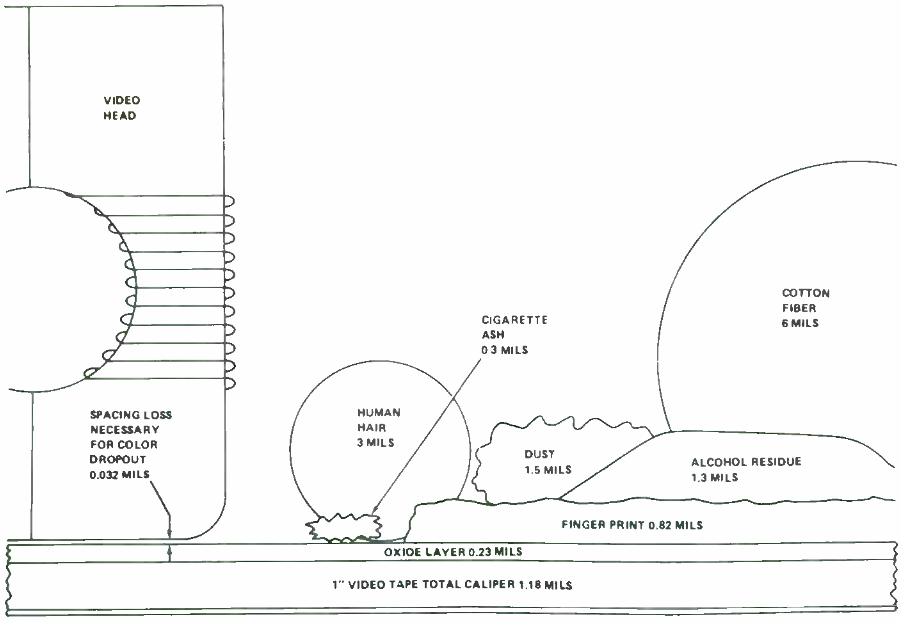

Fig. 1. Debris perspective on 1" video tape.

the machine manufacturer. The cleaning should be accomplished by applying the liquid to a lint free cloth, such as Texwipe (trademark), and gently rubbing the tape path surfaces with the dampened material. During this procedure extreme care should be exercised in cleaning the video heads. They can be easily broken or damaged with excessive pressure, particularly in the case where ferrite materials are used. Ferrite is the head material used in most all current day helical machines and in many rebuilt quadruplex heads. Since the video heads will not withstand a strong scrubbing action, compared to the other transport areas during cleaning, the stronger solvents such as isopropol alcohol, or machine manufacturers recommended solvents (usually Xylene type), are the best choices for removing debris build up on the video heads. They tend to help soften material buildup in addition to acting as a wash.

2. The cleaning of the capstan and the capstan pinch roller are especially important for two reasons. Any dirt or debris build up on these areas will cause impressions in the tape on each revolution that can cause drop-outs through a long period into the roll. Also the accumulation of material in this area may reduce the frictional pressures on the tape which are needed to maintain proper linear tape speed control. One word of caution: the material used in the pinch roller can be adversely effected by some cleaning solutions. The machine manufacturers recommendations should be followed for cleaning the pinch roller.

3. Cleaners, other than Freon TF or Genesolve D are relatively slow in evaporating, so give ample time for the transport to dry before threading the tape. This usually means about 30 seconds.

4. The video drum (scanner assembly) on helical machines have tape edge guiding surfaces that may accumulate debris. These surfaces need extra cleaning attention to ensure smooth tape handling and good interchange with other recordings.

5. The tape reel is specially designed for transporting magnetic tape. The reel should always be handled by the hub, which is the strongest portion of the reel. (See Fig. 2.) The reel flanges are designed to protect the tape edges, not to guide the tape. A reel should never be carried by the flanges. Handling the tape by the reel flanges or dropping an unprotected reel can bend the flanges. If the tape is rubbing against the flange of the reel, either the reel flanges are bent, or the reel pedestal or guides require adjustment. Avoid squeezing the flanges of a reel when putting on or taking off the transport.

6. The take-up reels should be cleaned at the start of each day. The tape winding surface of the hub and the inside surfaces of the

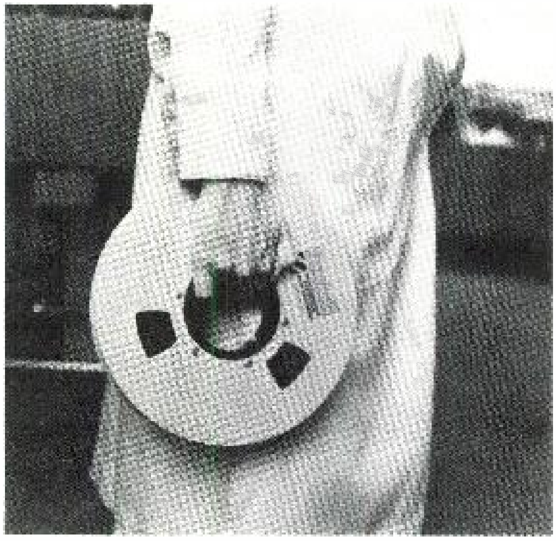

Fig. 2. Proper handling of a video tape reel.

flanges are the main areas of the reel needing this attention.

7. The sudden stoppage of a spinning reel of tape should be avoided in order to prevent the possibility of interlayer slippage that results in cinching or windowing of the tape. See Fig. 3. Newer machines have built in control that slows the spinning tape motion before stopping or reversing wind modes. Machines without this feature require the operator to slow the winding action down before stopping or changing wind directions. The break tension adjustments on the machine should also be checked according to the machine manufacturers operating manual to minimize problems of this type.
8. The condition of the pinch roller and its engagement with the capstan are very important in preventing creasing or edge damage to tapes. After much usage the pinch roller may become barrel-shaped, non uniform, hardened, or misaligned with the capstan. A straight-edge applied against the pinch roller, along with proper lighting, is one method of determining the flatness of its surface. One alignment check, proven to be helpful, is watching the vertical direction the tape travels when the pinch roller engages with the capstan. During repeated stops and starts, as the pinch roller is engaged, the tape should not show any signs of edge buckling on adjacent guides, if these parts are properly functioning.

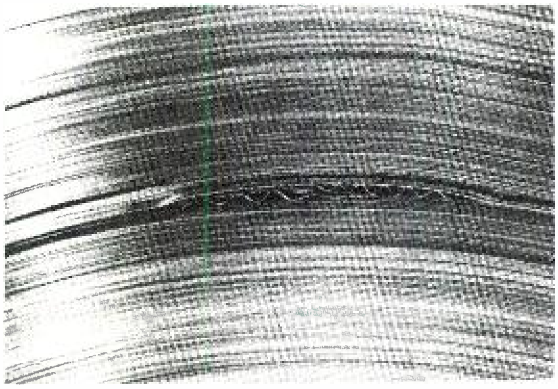

Fig. 3. Cinched tape. Note the complete fold over of one tape strand within the corrugated area.

9. The wear on capstans, pinch rollers, audio stacks, erase stacks, fixed guides (either front or backside), roller guides that don't turn, and fixed drum assemblies are key elements of the helical transport to consider if stiction or runability problems occur during editing or other modes of operation. As the stationary components wear, the land area the tape passes over increases. This increased surface area and a worn polished drum all create additional drag on the tape movement through the transport. How often these parts need to be replaced will vary with the type of video tape operation, but for machines dedicated to heavy editing service, based on field input, these fixed parts will require replacement more often. As a guide, they should be considered for replacement after approximately 1500 hours of use. An exception is the pinch roller which may require replacement on a more frequent basis.
10. The measurement and adjustment of the capstan pinch roller pressure against the capstan should be done after the machine has been running for approximately one hour.
11. The tape should not be touched or handled except at the ends, which is necessary for thread up on reel to reel machines.
12. The tapes and cassettes should be kept in their containers when not being used.
13. The cardboard debris from ripping open the master shipping cartons is a common source of debris. This unpacking should be done outside the operations area.
14. The outer wrap of reel to reel tape, when not being used, should be taped down with hold down tab material designed for this use. Proper hold down tab material will not leave adhesive residue on the tape after removal.
15. When threading the reel to reel machines, do not let the end of the tape touch the floor or table tops. This helps avoid pick up of dust and debris. Also any wrinkled or

creased ends of the tape should be cut off. Do not fold the tape on the take up reel when threading.

16. The tape edge guiding surfaces in the transport should be periodically inspected for wear. If grooves are visibly noticeable, the guides should be replaced.
17. The tension gauges used for making machine tension adjustments must be checked for proper calibration. The calibration check should be done with the same piece of tape that will be used during the machine tension measurement and adjustment. Tape thicknesses and other properties will effect the accuracy. Also, the calibration should be done with weights that will check the low and high ends of the scale.
18. Before attempting to use tapes that have just been exposed to environments considerably different than the operating environment, they should be given approximately 24 hours to stabilize in the operating environment.
19. The use of commercially available video tape cleaners on 2″, 1″, and 3/4″ formats has proven to be effective in extending tape life on old and used tapes. If these type units are being used, they also must be checked for proper alignment, tape tensions, and guide wear.
20. Reel to reel tapes and cassettes should be given a full length rewind to achieve a uniform and even wind before going to storage or preparing for shipment.
21. The recommendations for cassette type operations are basically the same as for reel to reel except for the obvious differences in the human tape handling aspects.

Storage Environment

1. At a minimum, the temperature and humidity should be controlled within the limits previously specified for the operations area; namely 70°F plus or minus 5% and a relative humidity of 50% plus or minus 20%. This has been adequate for most operations' library needs; however, if archival storage times are being considered, it is recommended that the relative humidity be kept at 45% or lower and the temperature below 70°F.
2. The tape reels or cassettes should be kept in their original containers and stored on end to assure the tape is being supported by the hub.
3. The air to this area should also be filtered to the same degree as previously recommended for the operations area.
4. When tapes are removed from the library, the containers should be inspected for accumulation of dust or debris and wiped clean, if necessary, before being taken to the operations area.
5. The storage environment must be controlled, but it should be pointed out that most tapes are designed to withstand relatively short periods of extreme environments, such as those encountered during shipment; namely, −30°F to 120°F.

Accidental Erasure

Because magnetic tape has the advantage of being erasable and then reusable, the question regarding the threat of accidental erasure arises. The following information should alleviate those fears as the chances of accidental erasure are extremely remote.

1. The coercivities of video tapes are in the range of 300 oerstads to over 1000 oersteds. Therefore magnetic field strengths below 50 oerstads have little effect on the recorded tape. Most sources of magnetic fields are well below this level. As an example the earth's magnetic field is approximately 0.6 oersteds. The field from an electric hand drill under full load is about 10 oersteds at the surface of the drill case.
2. The strength of a magnetic field drops dramatically as the distance from the field source increases. At a distance just 3 inches away from a 1500 oersted source the field strength is below 50 oersteds. Therefore, the mere spacing provided by a shipping container in separating the tape from external magnetic fields offers considerable protection.
3. Permanent magnets used in holding various objects to steel surfaces can have field intensities of about 1500 oersteds at the surface of their pole tips. These would present an eraseability problem only when held extremely close to the reel of tape itself.

Magnetic Recording Media

Part II: Principles and Current State of the Art

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INTRODUCTION

The broadcast of audio and video programming is greatly facilitated by the ability to store and recover information, and sometimes erase or alter it. Despite the availability of photography and other means of storing images, sounds and symbols, magnetic recording has enjoyed a long pre-eminence in these applications because of a number of features:

1. Easy recording by inexpensive transducer (head)
2. Stable long-term storage
3. Simple playback process with good signal-to-noise ratio, possibly using the same head as for recording
4. Easy erasure and editing or updating, especially important in broadcast studio environment
5. Relatively high surface information density
6. Thin flexible media, which can be rolled up (unlike phonograph record or video disk) for extremely high *volume* information density
7. Inexpensive media—high-quality magnetic video cassette tape is available at a lower cost per square inch than the transparent tape used to mend books!

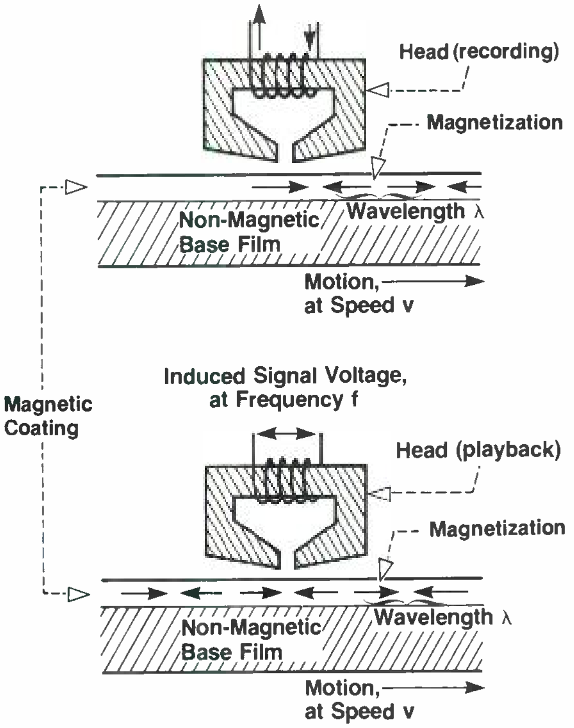
NOTE: Superscript numbers refer to references at end of the chapter.

This article will attempt to give a brief overview of the basic physical principles involved in magnetic recording, some of the leading types of recording materials, and the directions of current research and development. More thorough discussions of various aspects of recording theory and practice are available[1,2].

BASIC PRINCIPLES

Audio and video signals are handled as electrical quantities (voltages and currents); some fundamental physical laws that relate electrical and magnetic phenomena, and therefore make magnetic recording possible, are shown in Fig. 1. The changing flow of electrical current in the recording head gives rise to a changing magnetic field, which imposes a magnetization pattern on the recording medium (e.g., tape). During playback the reverse process occurs, as the changing magnetic field experienced by the head, due to the passing recorded magnetic surface, induces a signal voltage.

The inductive nature of the playback head dictates that only a *change* in the magnetic field experienced by the head leads to an output signal. Further, a faster change produces a greater output amplitude. A consequence of this is a frequency dependence in the output amplitude. Fig. 2 shows this dependence, for sinusoidal signals recorded at a constant depth and magnetic ampli-



$$\lambda = \frac{v}{f}$$

Fig. 1. The elements of magnetic recording and the relationship between signal frequency, transport speed, and recorded wavelength.

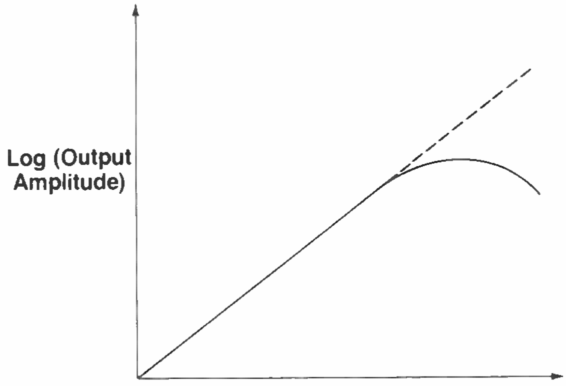


Fig. 2. Logarithmic plot of output amplitude vs. signal frequency, for sinusoidal magnetic recording at constant magnetization amplitude and depth. The slope of the linear rise is 6 dB per octave.

tude in the medium. The output is proportional to frequency over a large range; the fall-off at high frequencies (high recording densities or short recorded wavelengths) is due to signal loss from a number of sources which will be discussed later. The frequency dependence of the output is largely compensated for in many applications by the use of a frequency-dependent electronic gain; this process is called equalization.

The performance of a magnetic recording system depends, to a large degree, upon the magnetic properties of the material used to make the recording medium. The most significant properties are explained in Fig. 3, which shows a plot (called a hysteresis loop) obtained by placing a sample of the material in a varying magnetic field H and measuring its magnetization intensity M. The points at which the plot crosses the vertical M axis define the quantity known as the *remanent magnetization* M_r, which describes how much magnetization intensity the material is able to retain in zero field after being magnetized by a saturating field. The quantity usually specified (in the cgs system) is $4\pi M_r$, also designated as B_r, which is called the *retentivity*. The ratio of M_r to the *saturated magnetization* M_S is called the *squareness;* it helps to define the shape of the loop and therefore the recording properties of the material. The points at which the loop crosses the horizontal H axis define the *intrinsic coercivity,* often referred to as simply the *coercivity* and designated by H_c. This is the field needed to reverse half of the magnetization after saturation in one direction, resulting in zero net magnetization. The steepness of the plot as it goes from one direction of magnetization to the opposite one relates to the ability of the material to record a signal with sensitivity and precision. A quantitative measure of this steepness is the *switching-field distribution* (SFD), described in Fig. 4.

The response of a magnetic recording material to an applied field is clearly non-linear (see Fig. 3). In many applications, it is desirable to remove

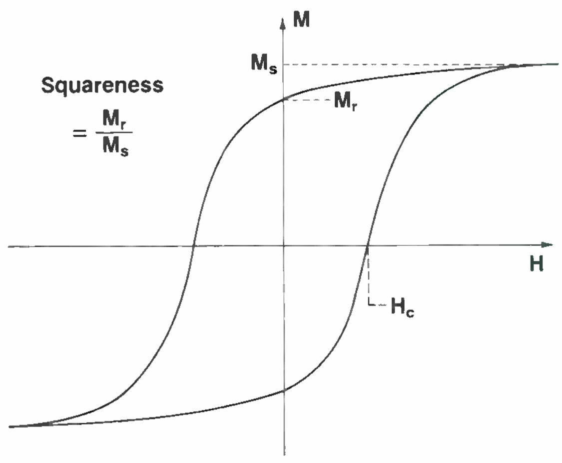


Fig. 3. Plot of magnetization M vs. applied magnetic field H, for a typical magnetic recording material.

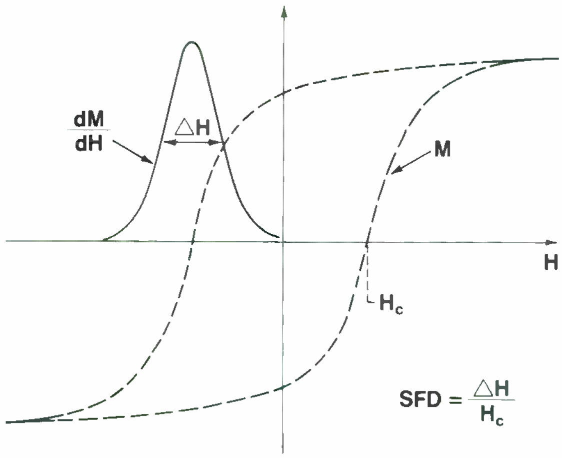


Fig. 4. Plots of magnetization M (broken line) and first derivative dM/dH (solid line) vs. applied field H. Width ΔH, used in the definition of the switching-field distribution (SFD), is measured at half the peak height.

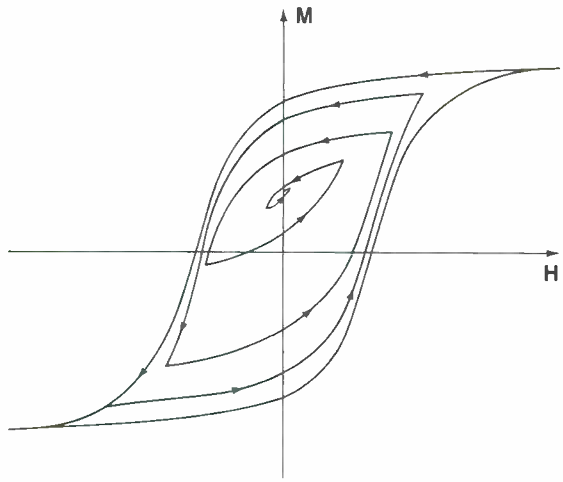


Fig. 5. The principle of ac-bias. As a specific small area on the tape moves away from the recording head gap, it is subject to a high-frequency bias field of decreasing amplitude. This causes its magnetization to vary as shown by the arrows, and to be left in a final state proportional to the signal field upon which the high-frequency field is superimposed.

this non-linearity so that the recorded magnetization can be made proportional to the signal current. The needed linearization of the response is accomplished through the technique known as ac-bias, in which the desired signal current is added to an alternating current of much higher frequency and amplitude before being applied to the recording head. The process and the result are diagrammed in Fig. 5; more complete discussions are available elsewhere[2,3]. The recording material is taken through a number of progressively smaller hysteresis loops as it leaves the vicinity of the recording head gap and is left in a state in which the remanent magnetization is very nearly proportional to the signal. The use of ac-bias is important in applications where the waveform read back must be an accurate reproduction of the input waveform, particularly in audio recording.

In some applications, the recorded information is contained in the timing of the zero-crossing points of the signal, rather than in the shape or amplitude of its waveform. Two of these applications, which do not usually utilize ac-bias, are digital recording and frequency-modulation (FM) recording. Digital recording is very simple in concept, although the applictions can be very sophisticated. Digital recording seeks only to convey reliably the strings of binary digits (abbreviated as "bits"), 0's and 1's, that make up the information used in computers and other data-handling devices. Fig. 6 shows a simple scheme for doing this. In practice, more complex procedures are used; some additional magnetic transitions are used for timing purposes and some convey information that is redundant but needed for the detection and correction of errors. Errors may occur through excessive random noise or through defects in the surface of the tape or disk.

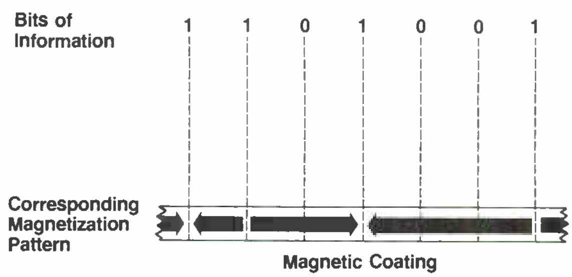


Fig. 6. A simple form of digital recording, in which 1's are encoded as magnetic transitions and 0's as the absence of transitions.

Digital recording, in addition to its obvious computer applications, is increasingly being used to replace analog recording techniques in audio and video applications[4,5]. The original signal is measured at precise intervals and the measurements expressed as a series of numbers that are then recorded digitally like any other data. The recovery of the signal consists of reading back the series of numbers (again, at precise time intervals), converting them into values of an analog quantity (e.g., voltage), and smoothing these discrete values into a waveform.

The first design consideration of a digital recorder is the frequency of sampling, which must be at least twice the highest frequency present in the signal to be recorded, according to the Nyquist theorem. Thus, for a high-fidelity audio signal containing frequencies up to a 20 kHz, sampling might be done at a rate between 40,000 and 50,000 times per second. Next, the dynamic range of the final reproduction is determined by the precision of each conversion to digital form. If a final dynamic range of 60 dB (an amplitude ratio of 1000) is required, then the binary code for each sampling must be able to express integer numbers from 0 to 1000. This requires ten bits per sampled value, since $2^{10} = 1024$. Similarly, a dynamic range of 90 dB requires a minimum of 15 bits per sampled value. The dynamic range is essentially the signal-to-noise ratio of the reconstructed signal, since the steps between adjacent numerical values provide the only noise inherent in the final output.

The advantages of a properly designed digital system include an excellent dynamic range, an essentially perfect time-base (absence of "wow" and "flutter"), and the ability to do repeated editing or dubbing operations without signal degradation. The tenth-generation audio reproduction made on a studio digital recorder is virtually indistinguishable from the original. This resistance to degradation of the recorded signal is intrinsic to digital technology and results from the fact that the information is encoded only in the timing of transitions. As long as these can be reliably detected and are not shifted, substantial degradation of the amplitude or shape of the digital waveform can occur with no effect on the result.

Digital magnetic recording, while not requiring linear reproduction of a waveform, makes great demands upon the recording system because of the drastically increased bandwidth. If an audio signal of 20 kHz bandwidth is to be reproduced with the modest dynamic range of 60 dB, frequencies up to at least 225 kHz are needed to accomodate its digitally encoded representation. An actual studio recorder might operate beyond 600 kHz, depending upon sampling rate, dynamic range, and the digital code used. Such frequencies dictate very short intervals between the magnetic transitions on the recording surface if tape usage is not to become prohibitive. On the other hand, the signal-to-noise ratio of the digital waveform that is read back need not be as great as that required in an analog recording. In general, the lower the signal-to-noise ratio of the digital waveform the higher the probability of mis-read bits (bit errors), but these can be detected and either corrected or concealed through the use of encoding schemes that involve a certain amount of redundancy in the recorded information. These schemes can overcome not only isolated single-bit errors but also the effects of recording surface defects that delete a number of adjacent bits. (See, for example, reference 6).

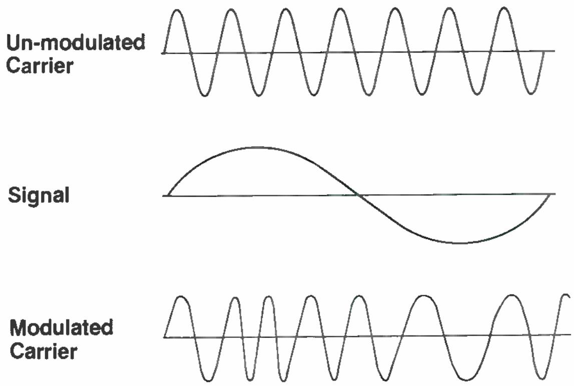


Fig. 7. The principle of frequency modulation (FM).

Error correction techniques make possible the use of very high densities in digital recording, provided that the heads and media can operate with the required resolution.

Another mode of magnetic recording that does not require linear waveform reproduction is frequency-modulation (FM). This is primarily used in video recording[7], but can also be applied to audio recording and is in fact used to give high-fidelity sound in some video cassette recorders. The principle of FM is that a carrier wave is generated whose *frequency* varies with the *value* of the signal to be transmitted (Fig. 7). It is this frequency-modulated carrier that is recorded on the magnetic medium. As in digital recording, the information is contained in the intervals between the zero-crossings of the recorded waveform; its amplitude is not significant as long as the signal-to-noise ratio is adequate for reliable detection. Unlike digital recording, FM recording contains an analog relationship between frequency and the value being transmitted. The benefits of FM, as compared to direct analog recording, are two-fold. One is the insensitivity to amplitude variations in the recorded waveform. The second, which is crucial to magnetic video recording, is that FM reduces the ratio of the highest to the lowest frequencies that need to be recorded; this is shown schematically in Fig. 8. Suppose that it is desired to record a video luminance (brightness) signal that contains frequencies from 30 Hz to 4.5 MHz. The upper and lower frequency limits have a ratio of 150,000 or 17 octaves. (By comparison, a full-fidelity audio system is required only to reproduce frequencies from 20 Hz to 20 kHz, a ratio of 1000, or 10 octaves.) A frequency span of 17 octaves (ratio of 150,000) is impractical in magnetic recording. First, the ratio of highest to lowest frequencies cannot exceed the ratio of the physical length of the playback head pole-pieces to that of the gap (reference 1, pp. 99-119). This ratio cannot in practice be as great as 150,000. Even if it could, the 6 dB per octave frequency dependence (Fig. 2) would require a dynamic range of more than 100 dB for direct video recording, too much for successful equalization. Frequency modulation, as indicated in Fig. 8, reduces the ratio of highest to lowest frequency; this ratio after modulation is as low as 1 octave in some video recording systems.

The frequencies involved in video reproduction are much higher than those in audio. Since there are practical lower limits on the recorded wavelength, this means that much higher head-to-tape speeds are needed. To attain these speeds, all video systems use heads mounted on a spinning drum. The tape is transported around part of this drum at a speed much lower than that at which the heads are moving across the tape surface. The various systems in current use employ different FM carrier frequencies as well as different schemes for conveying the color information, which in practice has a lower resolution and therefore requires a smaller bandwidth than that of the luminance information. In all systems the hue and saturation information is conveyed by amplitude and phase modulation of a special color sub-carrier (at 3.58 MHz in the U.S.A.). In the one-inch, open-reel tape format commonly used in broadcasting, the modulated color sub-carrier is simply added to the luminance signal and the resulting sum is the signal used to frequency-modulate the video carrier. In 3/4-inch U-matic systems and in non-professional one-half-inch cassette systems, the color sub-carrier (with its amplitude and phase information) is heterodyned down to a lower frequency (about 0.6 MHz) which is then recorded directly, using the frequency-modulated luminance carrier for ac-bias. Some recently developed one-half-inch cassette systems for professional applications utilize FM recording of both luminance and color information with a separate head, tape track, and carrier frequency for each. This achieves broadcast-quality reproduction using the same cassettes used in consumer systems, but at the cost of much lower recording densities (higher tape consumption).

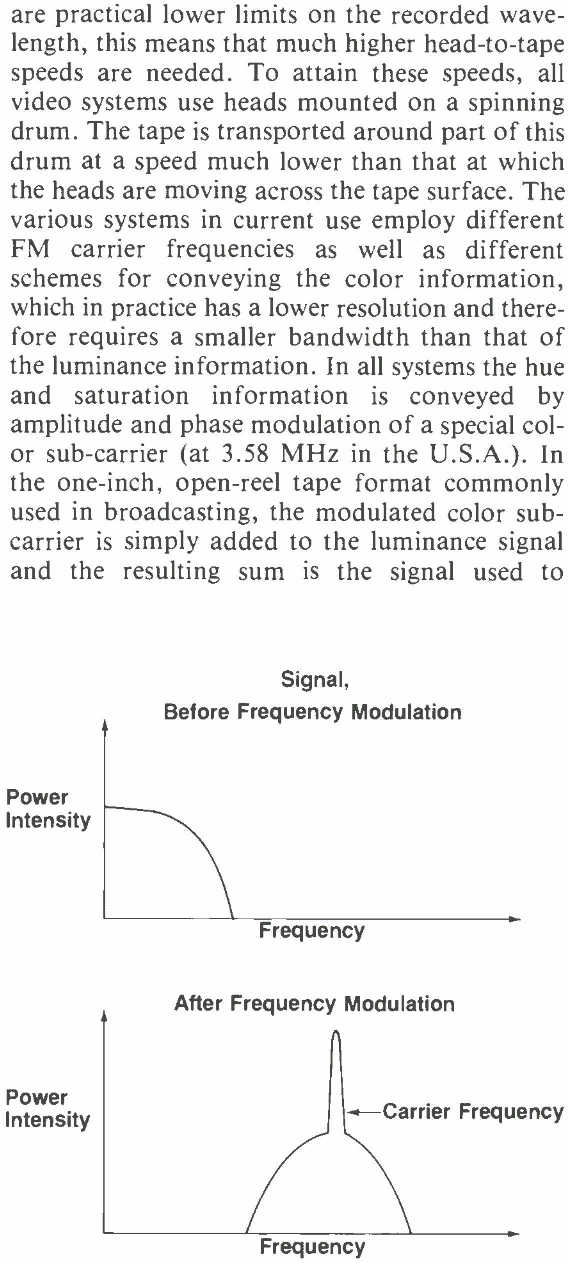


Fig. 8. In frequency-modulation (FM) the information is shifted to higher frequencies, but the ratio between highest and lowest frequencies is greatly reduced. FM schemes actually used in video recording do not use all of the sideband frequencies; see pp. 396-401 of reference 2.

MAGNETIC RECORDING MEDIA

Most magnetic recording media in use today are made by dispersing small magnetic particles in an organic binder and coating the resulting "paint" on a support material. See Fig. 9A. Each particle remains uniformly magnetized; its magnetization can change its direction but not its magnitude (this is a result of the very small size). The particles are sufficiently small that there are many of them in each magnetized region of the recorded pattern; this provides the needed resolution and signal-to-noise ratio. The particles can be of various types, according to the intended application[8,9]. In addition to the magnetic particles, the coating contains polymers for strength, lubricants to reduce friction and wear, and surfactants to aid in dispersing the particles. An additional particulate component is usually added to impart a controlled amount of abrasivity for head-cleaning purposes.

Another possible construction for a magnetic recording medium is shown in Fig. 9B. Like the dispersion-coated type, it also uses a nonmagnetic base. The magnetic substance is in the form of a thin metallic film. Thin films offer magnetization intensities far surpassing those available in the coatings of dispersed particles, which compensates for the films' much lesser thickness. The thinness of the films is in itself an advantage in that the magnetic material is all in very close proximity to the record or playback head. This intimate contact with the head is extremely important to high-density recording and is further enhanced by the great smoothness possible in the thin films. These attractive features make thin-film recording media an exciting area of research and development and possible candidates for future products. At present, their major commercial applications are in video instant-replay memories and in advanced computer disks.

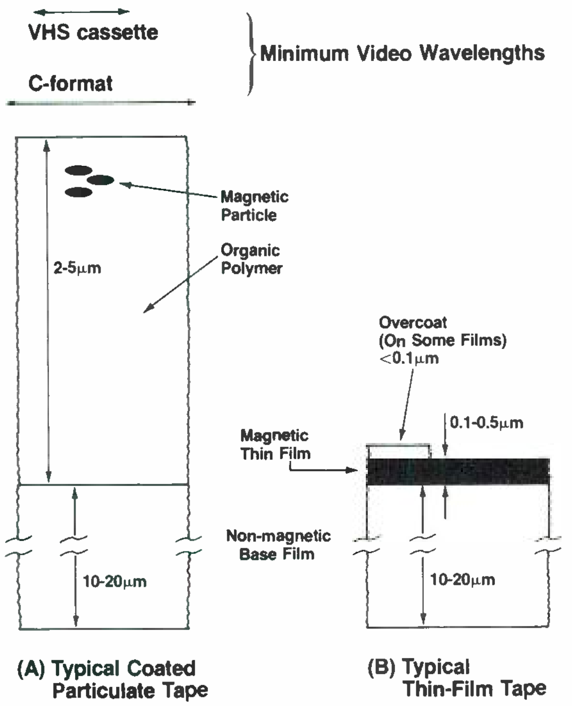


Fig. 9. Cross sections of a coated particulate recording tape and a thin-film tape. The particles are drawn approximately to scale but their spacing and uniformity have been exaggerated. Only a few have been drawn, although they actually fill the magnetic coating layer, occupying about 40% of its volume.

On the negative side, metallic thin films are subject to some concerns regarding their chemical stability, durability, and economical manufacture. Many of them are coated with a thin nonmagnetic overlayer for protection and lubrication. This layer, of course, tends to diminish the closeness of contact with the head. Metallic thin films are made by chemical plating or by some form of vacuum deposition and usually contain cobalt as a major constituent. The need for the special equipment used to carry out these processes is an additional barrier to the widespread commercial use of thin films in recording media. Futhermore, the thin-film products must compete with a particulate technology that not only benefits from more than 30 years of experience in manufacturing and application but is also evolving toward greater capability.

For the foreseeable future, economic and practical considerations dictate that coatings of dispersed particles will be predominant in audio and video tapes. The earliest comercially available audio tapes, made in the late 1940's, used iron oxide particles. The most important oxide, $\gamma-Fe_2O_3$, has been greatly improved as to particle shape and size since its introduction, and it retains an important role in audio, video and data-recording tapes today. It is also the most commonly used material for the disks, both flexible and rigid, used in digital data storage for computers. The relatively low coercivities (250-400 Oe) of pure iron oxides, however, proved to be a serious limitation as recording densities increased. An explanation for this is found in the self-demagnetization effect that exists in magnetized materials. Every magnet generates an internal field, which tends to oppose the magnetization that gives rise to it. This effect becomes stronger as the magnet is made shorter along its direction of magnetization (Fig. 10), and will be enhanced by the presence of nearby magnets if similar poles are placed together. The result of these phenomena is that magnetic recording, conventionally visualized as having the magnetization parallel or antiparallel to the direction of motion of the head with respect to the tape or disk (Fig. 1), suffers from a demagnetization effect that becomes more severe as the recording density becomes higher (that is, as the recorded wavelength becomes shorter). The ability of a magnetic material to resist demagnetization is expressed by its coercivity.

The requirement for higher coercivities was first met in the mid-1960's with the introduction of chromium dioxide (CrO_2). Like iron oxide, this material derives its coercivity and squareness properties, and therefore its ability to retain magnetization, from the needle-like shape of the particles. Chromium dioxide was first introduced for digital computer tapes. It was soon applied in audio cassettes and is today used in some video cassettes.

At the present time, the most widely used materials for applications requiring coercivities in excess of 400 Oe are iron oxide particles modified by the addition of cobalt. The interaction of the cobalt ions with the iron oxide structure provides additional resistance to the switching of the magnetization direction, and thus increases the coercivity. Values in the range of 500-600 Oe for audio cassette tapes, 600-700 Oe

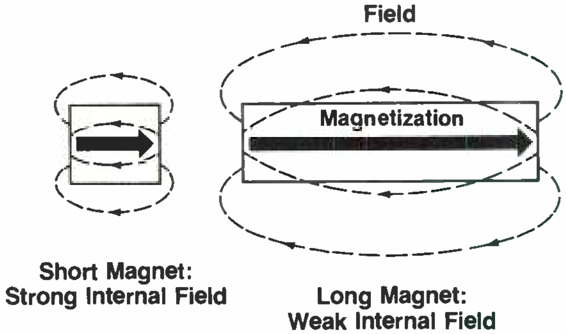


Fig. 10. Internal demagnetization fields for magnets of different shapes.

for video tapes, and 800 Oe or above for future applications are readily achieved. Many processes exist for adding the cobalt to the particles. Most in use today leave the cobalt segregated near the surface of the particles. Such particles, called "cobalt-surface-doped", "cobalt-adsorbed", "cobalt-epitaxial", or "cobalt-ferrite epitaxial" are now the predominant materials used in video cassettes and open-reel video tapes and are finding increased use in high-density data diskettes. Like undoped oxides and chromium dioxide, they retain magnetization along the particle axis. The manufacturing processes of tapes are usually designed to orient the particles along the tape length in order to increase the maximum remanent magnetization in this direction.

Tapes made with particles of metal (iron or iron alloys) have much higher remanent magnetization values than are possible in tapes that use oxide particles, and accordingly give much higher signal levels. This property is especially valuable in very compact formats, and metal particles are used in some audio cassette tapes and also in 8-mm consumer video tapes. These products have generally had coercivity values in excess of 1000 Oe, requiring much higher head fields for recording and erasure than those needed for oxide tapes. Recently, however, metal particles have become available with coercivities near 700 Oe, similar to those of cobalt-modified oxides.

Another particulate material currently being studied for recording applications is barium ferrite. The particles are formed in the shape of small hexagonal plates, each of which has its easiest axis of magnetization perpendicular to its flat faces. If these plates are incorporated into a tape coating so as to lie roughly in the plane of the tape surface, recording can be done primarily with perpendicular magnetization.

Fig. 11 shows the properties of various particulate magnetic recording materials.

Whatever the composition of the particles in a recording tape or disk, their size is extremely important, since it determines the number of particles contained in each magnetized region of the recorded signal. The greater this number, the greater will be the maximum signal-to-noise ratio that the recording medium can provide, since each particle contributes a pulse of magnetization to the playback head[10]. A great many small pulses clearly will combine to make a more accurate, less noisy signal than a smaller number of large pulses (even though the total density of magnetic material present may be the same), and the effect of random fluctuations in particle size or placement will be less.

One of the most important and difficult arts in the manufacture of magnetic recording media is the dispersion of particles in the coating. If ap-

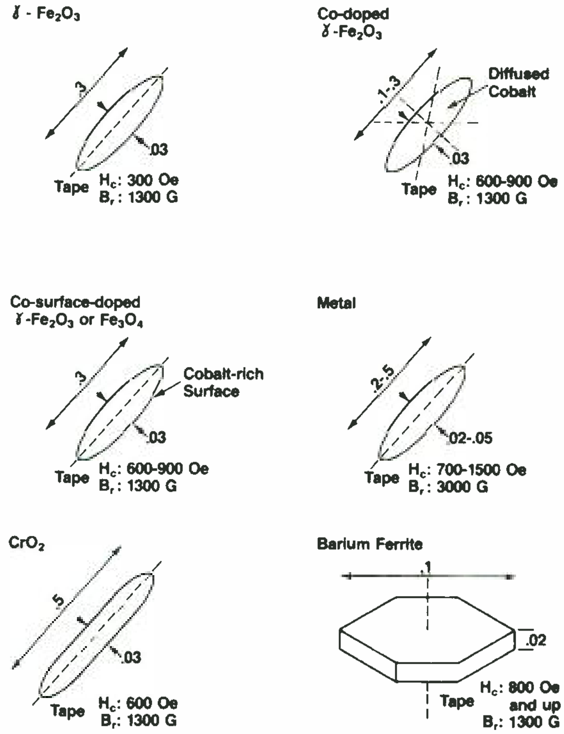


Fig. 11. Various particles used in magnetic recording. The preferred axes of magnetization are shown by broken lines. Some typical, approximate dimensions (in μm) and magnetic parameters are indicated.

preciable amounts of the particles are clumped together, the recording performance will suffer. The noise properties of the signal will be dominated to some extent by the clusters of particles, which are of course much larger than the individual particles themselves. A great deal of attention must therefore be paid to the formulation and milling of the dispersion, and to its coating and drying.

In the search for magnetic materials that give higher performance, care must be taken to avoid various undesirable features. For example, the trend to reduce particle size, and thus enhance signal-to-noise ratios, must not be carried to the point where the particles are so small that they begin to show the effects of thermal instability. In particular, the presence of excessively small particles can lead to the unwanted acquiring of a signal in one layer of tape on a reel as a result of the fields due to a strong signal on an adjacent layer. This phenomenon is often referred to as "print-through" or "pre- and post-echo", and is important only in (ac-bias) audio recording. In some applications, especially audio, thorough erasure is an important concern and one that places an upper limit on the coercivity of the recording medium. Efficient recording and erasure also require that the material's switching-field distribution (Fig. 4) be relatively narrow. This means that the fields required to reverse the magnetization of the individual particles should be tightly clustered around the coercivity (which is essentially the median switching field).

Although this discussion has focussed on magnetic properties, one must not underestimate the importance of the non-magnetic components of recording media. The base film, the binder polymer, and the various lubricants and other additives in the coating are crucial to strength, durability, and freedom from undesirable levels of friction. The manufacture of recording media requires at least as much capability in organic chemistry as in magnetic materials. The ability to reliably produce coatings with smooth surfaces, substantially free of defects, is also an absolute necessity.

PROGRESS TOWARD HIGHER RECORDING DENSITIES

The state of the art in magnetic recording has clearly advanced in recent years. Fig. 12 shows

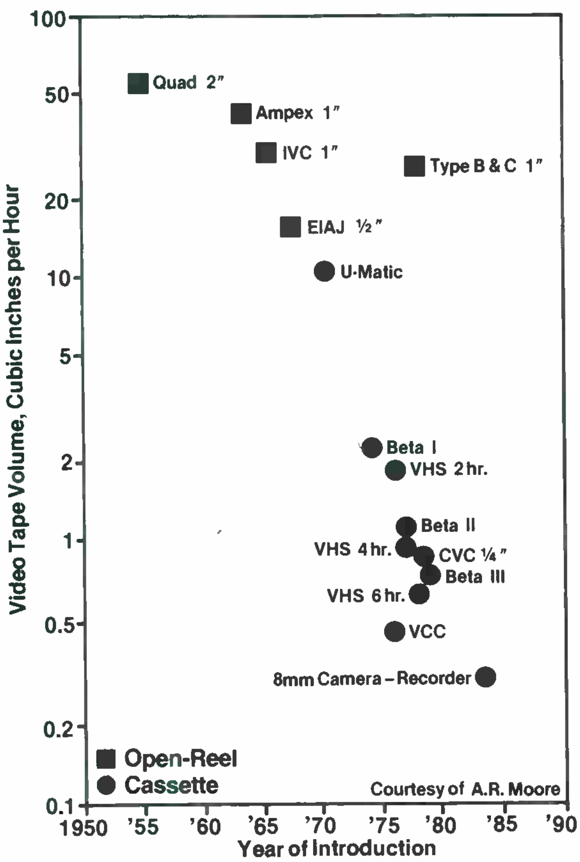


Fig. 12. Tape volume required per hour of playback, plotted against year of introduction, for various video formats. This plot was supplied by A.R. Moore, of 3M.

the great gains in practical recording density that have occurred in the video area. Intensive research and development is today aimed at continuing this trend. Two major driving forces are the desire for more compact consumer video products, especially cameras, and the trend toward digital recording of both audio and video signals. A brief survey of current developments will be given here, with emphasis on digital video.

As was mentioned earlier, the use of digital techniques requires a frequency bandwidth that is much increased over that needed for analog representation. Consider, for example, a broadcast-quality video signal that requires a bandwidth of roughly 10 MHz for adequate analog (FM) transmission. The analog recording system must handle 2×10^7 magnetic transitions per second. A digital system designed to transmit this same signal adequately would require nearly 3×10^8 bits per second[5]. Assuming that one bit of information corresponds to one magnetic transition (although this ratio varies somewhat with the digital code used), the digital recording system must handle nearly 3×10^8 magnetic transitions per second, a fifteen-fold increase over the analog rate. Clearly, if the use of digital rather than analog technology is not to entail a large increase in tape consumption, the digital recorder must use a higher density of magnetic transitions per square inch of tape. This density is the product of track density and the linear transition density along a track. Table I gives the values of these parameters in some practical and hypothetical video systems. The projected increases of recording density require the achievement of adequate signal strength and signal-to-noise ratios despite the use of much smaller magnetized regions in the recording medium. The difficulty presented by self-demagnetization at high densities was described earlier, as an explanation for the need to increase coercivities. However, a limit exists as to how much high-density performance can be bought by simply increasing the coercivity, since available heads must still be able to record, and also to erase or over-write, the material. Another route to avoiding or relaxing the density restrictions imposed by demagnetization is to avoid the head-to-head arrangement of magnetized regions. This can be done by using, to a greater or lesser extent, the perpendicular (sometimes called vertical) component of magnetization[11,12]. See Fig. 13. The relative merits of the various recording patterns, and of various head designs for creating and reading them, are currently under intensive study in many academic and industrial laboratories. It is likely that recording at relatively high densities has always used some degree of perpendicular magnetization, but new materials, including barium ferrite particles and some metallic thin films, are designed to accentuate this com-

TABLE I

System	Current C-format, one-inch open-reel video	Current six-hour VHS video	Projected digital video[a]
Magnetic Transitions per second (maximum)	2×10^7	10^7	3×10^8
Transitions per inch along track (maximum)	2×10^4	4×10^4	10^5
Tracks per inch	1.4×10^2	1.3×10^3	10^3
Transitions per square inch (maximum)	3×10^6	5×10^7	10^8
Tape consumption, in square inches per second	10[b]	0.2[b]	3

[a]This system embodies recording densities achieved in the laboratory. Digital systems closer to practical realization use somewhat smaller densities and achieve a tape usage closer to that of the analog C-format.[5]

[b]Includes tape area not devoted to video information.

ponent. The multiaxial cobalt-doped oxide particles (see Fig. 11) can be used to implement mixed-mode recording (Fig. 13) in a more or less isotropic medium.

Output losses at high densities can result from a number of causes other than demagnetization. Some are related to head design, and it is very clear that development of heads and recording media must proceed together in the search for higher-density recording. Fig. 14 shows some head designs under current consideration. The critical dimension of the head used for playback, the gap width of a conventional ring head or the pole width of a perpendicular-recording head, must be smaller than the shortest wavelength to be reproduced. The head used to record the waveform can have larger dimensions; indeed, this may be advantageous.

Whatever the head design and the recording medium composition, the closeness of the contact between the head and the surface of the medium is a factor of major importance in determining the achievable recording density. This

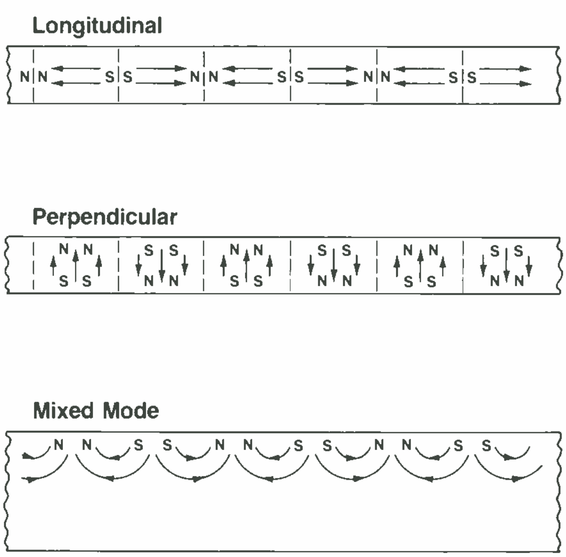


Fig. 13. Possible magnetization patterns in recording media. For the perpendicular component, increasing the recording density may actually decrease the demagnetization fields; see Fig. 10.

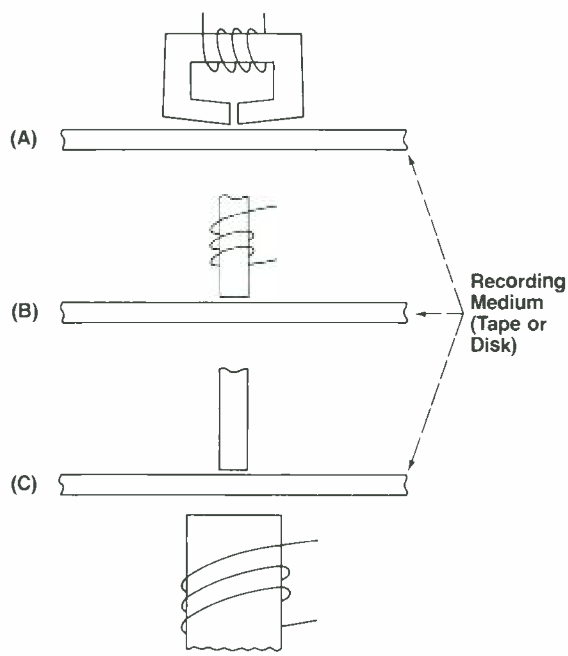


Fig. 14. (A) Conventional ring-type head, useful for both longitudinal and perpendicular recording. (B) Single-pole head, for perpendicular recording. (C) Single-pole head, driven by auxiliary pole, for perpendicular recording[11]. All heads can, if made with suitable dimensions, be used for reading as well as for writing. (B) and (C) are largely in the research and development stages at present.

follows from the fact that the resolution with which the head can record and read back deteriorates rapidly as spacing increases. The effect on the recording process has been estimated as a loss of 44 dB per wavelength of spacing, for a sinusoidal signal[13], and that on the playback process as a loss of 55 dB per wavelength[14]. This total of nearly 100 dB per wavelength means, for instance, that for sinusoidal recording at 50,000 transitions per inch (corresponding to a wavelength of 40 microinches or about one micrometer) every microinch of spacing will cause about 2.5 dB of overall loss in the reproduced signal. Obviously, variations in the spacing will amplitude-modulate the signal with the same sensitivity, and therefore add noise components to the signal. Thus, media surface smoothness may well be the ultimate limiting factor of magnetic recording density, although further development of heads and media are needed in order to reach this limit[15]. In addition to smoothness, the freedom from flaws and contaminants is crucial, if information "drop-outs", or momentary losses of signal, are to be minimized. As mentioned earlier, techniques of error detection and correction (or concealment) allow satisfactory performance even if imperfections exist in the recording medium. Even a highly sophisticated error-correction system, however, can be overwhelmed by defects that are excessive in number or size. As the recording density increases, the size of the flaw needed to cause a given system failure (visible video defect, audible audio defect) decreases. Thus, the development of advanced magnetic recording media entails the search for the means of producing highly perfect surfaces, as well as ideal magnetic properties.

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5.8

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ATTRIBUTES OF FILM

Color motion picture film has three photosensitive layers coated on a plastic base. Exposure and processing produce dye images in these layers which, when projected on a large screen or scanned in a telecine, produce the color pictures.

Film in the 16mm width is the most common format for distribution of television programs. The 35mm width has been in general use for many years for large-screen picture projection in theatres, and for the production of prime-time television programs on film telecast by the major networks. Now this format is being used extensively in television postproduction houses, and for the production of television commercials. The newer generation of television film scanners has interchangeable optical blocks enabling either format to be reproduced in telecine equipment.

Professional motion pictures, most prime-time television programs and television commercials are made on color negative film. From these originals, prints can be made for distribution on either 35mm or 16mm materials. The newer generation telecines are capable of reproducing from either color positives or color negatives.

To show motion pictures on large screens, the projector advances the film one frame at a time, and a rotating shutter cuts off the light while the film is being moved. A claw or Geneva mechanism advances the film by engaging perforations along the edges. Film 35mm wide has perforations on both edges, but the 16mm material used to make prints has perforations on one edge only; the space on the other edge is taken up by the sound track.

In camera-type telecines the film is advanced intermittently, one frame at a time, by means of the perforations. But in the newer film scanners film movement is continuous, and the perforations are used only to drive a sprocket that generates framing pulses.

The film sound track may be either optical or magnetic. Telecine projectors and film scanners usually have interchangeable optical/magnetic sound playback heads. In the playback of an optical sound track, the lens imaging the exciter lamp filament at the film must be sharply focused on the emulsion side. Usually an adjustment is provided for this purpose, as the emulsion side in 16mm prints may be either towards the lens (preferred position) or towards the light source.

The standard motion picture frame rate around the world is 24 frames/second for both 35 and 16mm formats. The 35 format has 16 frames per foot; so at 24 frames/second the rate of film movement in a projector or scanner is 90 feet per minute (fpm). With 40 frames per foot in 16 mm the film passes through a projector or scanner at 36 fpm. The size of a 35mm film frame is 0.825 inch wide and 0.600 inch high, compared with a 16 mm frame, 0.380 inch wide and 0.284 inch high.

Fig. 1. EASTMAN Color Print Film 5384 (film can label) (courtesy Eastman Kodak Company)

With proper care film should have a long useful life, but the film surfaces can be scratched easily in faulty equipment or by careless handling. Also, the dyes used in producing the color images have been greatly improved over the years, not only in colorimetric characteristics, but in dye stability, or resistance to fading, as well. Useful life has been increased as much as several times over that of previously available print films. And, since fading of the three dye layers usually occurs at different rates, and causes changes in color balance that are difficult—if not impossible—to correct in telecine (with available electronic adjustments), a dye-stable print film should help to reduce the amount of dye fading so often seen in old movies. (See Fig. 1.)

ADVANTAGES OF LOW CONTRAST PRINT FILMS

When color television broadcasting started in the mid 1950s, the motion picture industry was the only source of pre-recorded program materials. It was soon recognized that significant picture quality losses occurred in telecasting films made for large-screen projection in theatres because of the inability of the television system to transmit the range of gray scale values needed to create acceptable pictures on motion picture screens. Committees of the Society of Motion Picture and Television Engineers (SMPTE) tackled this problem and published a report recommending that the density range in films for television should be limited by adjustments in staging and lighting rather than in the making of the color prints. In this way prints could be produced with a range of density values that the television system could reproduce.

For various reasons it has not always been possible to adhere to these recommendations. For example, on location, lighting ratios cannot be held to 2:1 as recommended, nor can the reflectivity of the lightest scene areas be held below 60 percent. An even more compelling reason is that market requirements make necessary the production of motion pictures for both purposes—theatre projection as well as television display.

At the start of color television broadcasting in Europe, engineers from the British Broadcasting Corporation, members of a working group in the European Broadcasting Union, defined the preferred luminance characteristics of films for television presentations as a density range of approximately 0.20 to 2.50. C. B. B. Wood of the BBC, writing in the Journal of the Royal Photographic Society pointed out that the contrast handling capacity of a color television system in ordinary home viewing conditions is likely to be somewhere between 20:1 and 30:1, whereas the contrast in typical theatre projection is about 160:1. Because of ambient light on the television screen, it seems better to compress the lighter rather than the dark tones. The ideal arrangement would be a single color negative printed either on conventional print film or on a special print film for television.

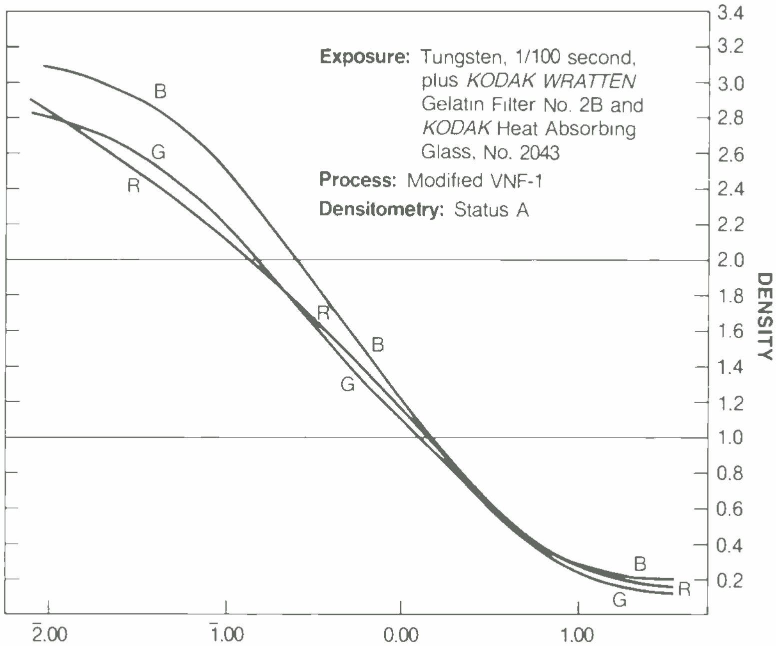
In response to these initiatives, Kodak Ltd. in the United Kingdom introduced a new Eastman color print film Type 5744. This film was designed with reduced upper scale contrast compared with print films for projection on theatre screens. This original attempt to conform with television density requirements has resulted, after numerous modifications, in a reduced contrast product in the Eastman color LC print Film 5380/7380.

The transfer characteristics of these new low contrast print films is a compromise among several considerations. The contrast has to be low as possible for good telecine performance, yet high enough to provide acceptable screen images in direct projection in review rooms, and the film must have enough color saturation to keep chroma gain at a fairly low level.

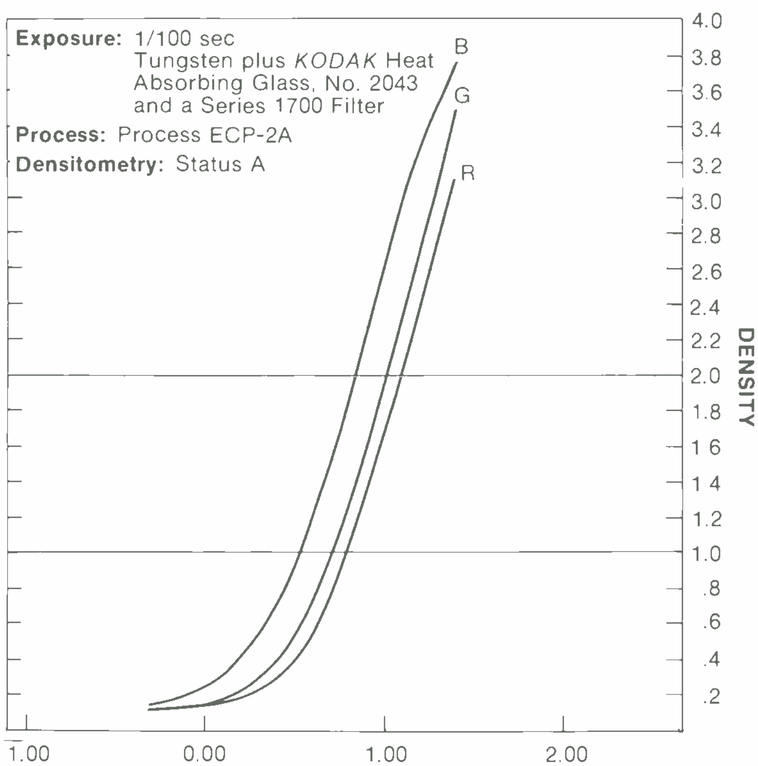
When a film with a contrast range of 160:1 is being squeezed into the television system, the shadow areas in the pictures are compressed and much of the shadow detail is lost. Some improvement can be achieved through the use of black stretch circuitry in the telecine, but little can be done with films made deliberately dark in shadow areas to make the story on the screen in theatres more convincing. No doubt television reproduction of these old motion pictures could be improved considerably by printing on low contrast print film.

FILM DENSITY TO ELECTRONIC OUTPUT

The term "density" as used in photography and motion pictures refers to the light-absorbing properties of film images. Density is defined as



(Positive print film)



(Negative print film)

Fig. 2. Characteristic curves for typical negative and positive print films.

the logarithm of the reciprocal of the transmittance, and transmittance is the ratio of the transmitted to incident light. The response of a film material is usually illustrated graphically in the form of a characteristic curve, produced by plotting density against the logarithm of the exposure.

In the negative-positive motion picture process it is customary to use a relatively low contrast negative film in the camera, and then print the negative on a relatively high contrast positive film. Fig. 2 shows the characteristic curves plotted on graph paper. Photographic gamma is the slope of the straight line portion of the characteristic curve.

The range of light values reflected from different parts of a scene can be placed anywhere on the negative characteristic curve by adjusting exposure level, but good practice is to place the darkest area of the scene at about 0.10 above base and fog level. This is accomplished by making use of a photoelectric exposure meter set for the exposure index of the negative film.

When a print is being made from the negative, printer exposure is adjusted to place the range of scene brightnesses on the positive characteristic curve in such a way that the lightest scene area is close to clear base so as to retain bright highlights on the screen. Surveys have shown that good quality motion picture prints are likely to have a minimum density of 0.25 to 0.40, with maximum densities from 2.30 to 2.70.

SMPTE Recommended Practice RP46-1972 specifies that the density corresponding to white level for television reproduction should be 0.30 to 0.40, while dark areas may have densities as high as 2.50.

However, image details and color in these areas may be lost or distorted. The Kodak gray scale test slide for setting up camera telecines has minimum and maximum densities of 0.30 and 2.35, and in a properly adjusted telecine these densities should produce video waveform levels of 100 and 7.5 IRE units.

From the foregoing brief description of the correlation between the motion picture and television imaging systems, it can be seen that in the production of a program on film, important scene elements can be placed at any desired levels on the television waveform monitor and in the brightness range of a television picture by adjusting the reflectances of these scene elements with the aid of supplementary lighting.

However, with a conventional high contrast print material, there is almost always a problem with preserving detail in the darker areas of the television pictures.

For anyone interested in how the coordination between film and television is achieved, a paper by DeMarsh "Optimum Telecine Transfer Characteristics" in the October 1972 of SMPTE Journal has much useful information.

OPTIMUM TELEVISION PICTURES FROM FILM

Two distinctly different methods of operating telecine equipment have been in general use for many years. One method is to allow the telecine to run unattended, signal level being maintained within permissible limits by automatic signal level control. This is the normal method of operation in television stations around the country. The other method, employed mainly in film and video postproduction is manual control, a skilled video operator being employed to adjust the telecine camera controls as necessary to compensate electronically for color, hue and saturation variations in the films being telecast.

More Than Just Acceptable

Automatic signal level control is an economically efficient operating method, and for the most part acceptable television pictures can be produced in this manner. Of course, films made to conform with published recommended practices for television should give equally good pictures whether the automatic signal level control is turned on or off. But in practice, the densities and colors in some films vary to a much greater extent than the recommended practices specify. Designers of television equipment have always claimed that automatic signal level control reacts much more quickly to scene-to-scene density shifts than even the most skillful video operator. However, the automatic control circuitry cannot distinguish between a good or a poor picture, something that an operator can immediately detect and then make appropriate adjustments in the settings of the camera controls.

In the large network centers where costly program materials are being assembled and reproduced, personal attention to signal levels, color balance and general picture quality is essential, especially in dealing with film commercials. For the sponsors of these products, color and texture must be preserved in the television pictures, and perhaps even enhanced. Even with the old vidicon color telecines, the changes that can be made in the appearance of the television pictures are usually quite surprising.

Scanners

Much more striking is the performance of the flying-spot scanner in reproducing color films. Its potential was recognized early. For example, as soon as the Rank Cintel scanner became available in North America, television postproduction companies were the first to install these telecines. By 1984 over 130 were in use in these centers.

The capabilities of the flying-spot scanner have been greatly extended with Digiscan, a digital storage and processing unit; Amigo, a telecine programmer, allowing speed control scene-by-scene, and many other functions including frame-by-frame color correction. After making appropriate adjustments in gain, gamma, pedestal, enhancement and so on, to produce the best possible television pictures as seen on the telecine picture monitor, the operator presses a data entry button to enter the corrections plus the scene timecode into a memory. With a secondary color correction unit, hue and saturation of any of the three primary and three secondary colors can be altered in any desired manner. These changes and corrections can be recalled from memory automatically when the film is being transferred to videotape.

Color Correction

Corporate Communications Consultants of Holbrook, New York, is a pioneer in the development of highly sophisticated color correction equipment for use in the transfer of films to video tape. Among several models of color correction equipment available from the company, The System 60XL is perhaps the most versatile. The digitized analog variable control panel contains 40 or more high resolution digital shaft encoders, one for each variable, organized in three major subsections—color or balance (pedestal, gamma and gain for red, green, and blue); color derivatives (hue and saturation for red, green, and blue); and a luminance section. The digital entry panel and floppy disk system form the main command console. Interfaces are available for the Rank Cintel Mark IIIC flying-spot scanner and the Bosch and Marconi CCD scanners. These interfaces, connected to a color correction system, can be switched by actuating a push button.

To create a color cross fade or dissolve, for example, the computer requires only that the first frame be displayed on the color monitor and the film jogged to the closing frame. The computer automatically adjusts the values for the frames in between and produces an aesthetically pleasing picture with a gradual fade that is smooth to the eye.

An important advantage of using this color correction equipment with the flying-spot and CCD scanners is that the film drive uses tension, not sprockets; therefore, valuable film originals can be run forwards or backwards with little danger of damage.

When The System 60XL is used with CCD scanners, the differential or RGB signal outputs of the digital store produce equivalent signals with the color corrections incorporated. All the critical color parameters can be corrected while the scanner is in the freeze mode. In most cases three frames have to be scanned before the optimum result can be achieved.

American Cinematographer Magazine has published a booklet, "Electronic Production Techniques," in which one chapter, "Electronic Color Control," describes the operation of The System 60XL, with four pages of color photographs showing the range of color modifications that can be achieved with this equipment, starting with a color negative.

Film/Video Colorist

Video operators working on electronic color correction equipment are known sometimes as *colorists.* These highly skilled technicians can quickly assess a film scene on the telecine picture monitor and decide on the nature and extent of the corrections needed to produce the best possible or most attractive television pictures, or to portray commercial products more accurately. To perform these operations efficiently, the telecine and its picture monitor must be carefully set up with a suitable test film and a monitor calibrator.

Video colorists need to have a good understanding of colorimetry and the way in which color images are formed in motion picture film. This knowledge enables them to predict with confidence the effect of a particular correction on the color picture. For example, if the picture on the color monitor has a reddish tinge, one might want to adjust the setting of the red control. But the color unbalance actually may be in the magenta direction, requiring a green adjustment to correct the reddish color. The video colorist has the advantage that the color monitor can be switched back and forth between the corrected and uncorrected pictures to determine whether or not the predicted change in the picture has been achieved. But with a lot of people waiting, the colorist who can make the right adjustment every time is soon in great demand.

ADVANTAGES OF CHOICES

Only a few years ago, broadcasters in the United States and Canada were limited to the use of camera-type telecines for reproducing films. Now there are several choices of systems and makes of equipment with different characteristics, as well as many new opportunities for using film more effectively as a source of television programs.

Camera-Type Telecines

Camera-type telecines equipped with automatic signal level control offer the advantages of easy frame rate conversion, economical unattended operation, simplified fault-free switching, optical multiplexing of films and slides for continuous on-air operation, and rugged, long-life, relatively trouble-free equipment installations.

The storage-type vidicon tubes used in most of these telecines give surprisingly good television pictures from film, considering the way in which the signals are generated. However, these camera tubes suffer from lag, which adversely affects resolution. Another major problem with these telecines is the difficulty of maintaining registration of the separate elements of the color pictures. The multiplexed telecine has a complicated optical path for the projected film, with numerous glass-to-air surfaces, requiring frequent cleaning to reduce flare.

Projection of the film into the camera requires an intermittent mechanism, advancing the film frame-by-frame at the rate of 24 per second. Some telecine projectors can run either backwards or forwards, but there is always the risk of damage to the film surfaces and the perforations. A major problem in all intermittent projectors is maintaining acceptable image steadiness as the picture frames are registered successively at a rapid rate in the projector gate.

Flying-Spot Scanner

In the flying-spot scanner, film motion is continuous and the light source is the rapidly moving spot of light in the raster on the face of a cathode ray tube. This type of scanner has been in use in Europe since the start of television broadcasting. The essential 1:1 relationship between film and television frame rates was easy to achieve in the 625-line, 50-field television system by speeding up the film frame rate slightly to 25 per second. This method could not be used in North America, however, because of the much greater difference in frame rates—24:30 for the 525-line, 60-field system.

In 1980, Rank Cintel, an English company, demonstrated the Mark IIIC flying-spot scanner designed for the North American television service. In this scanner, frame rate conversion was accomplished by sequentially scanning the continuously moving film, writing the scanned lines into a store and then reading out the lines into complete, interlaced television frames.

This development has had a profound effect on the use of film in television in the United States and Canada. Pictures from color film reproduced in the flying-spot scanner have a sharper appearance, with brighter and more saturated colors. There is complete freedom from registration errors in the scanner since color separation takes place after the film has been scanned. Interchangeable plug-in gate options are available for 35mm and 16mm films as well as 2 × 2 inch slides, with a single film transport.

The objectionable effects of storage and lag associated with camera-type telecines are eliminated in the flying-spot scanner. The scanner also produces a very clean black level, and flare is reduced to a minimum.

These features make the flying-spot scanner particularly attractive in the post-production of television programs where best possible picture quality and flexible, versatile user control over the transfer process are essential.

For the television station concerned mainly with on-air telecasting of 16 mm films, the flying-spot scanner with its extensive array of sophisticated features may not be an economically viable alternative to the continued use of existing camera-type telecines. Instead, a much simpler and more cost- effective telecine design—the CCD scanner, now available from several equipment makers—is likely to be a better choice.

The CCD Line Array Film Scanner

The main advantage of the CCD scanner is its simplicity. As the film is advanced continuously, an optical system focuses a narrow slit of light across the surface of the film. Light passing through the film, after color separation, falls on single line arrays of image sensing elements (CCDs) where the signals are generated. The signals emerge from the sensors in sequential scans of the film frames and are then converted into interlaced television frames by a 3, 2, 3 utilization of stored fields.

Another advantage of CCD film scanning is operating economy. Barring accidents, the line sensors have a virtually unlimited life, compared with costly, expendable flying-spot CRTs or camera tubes. The scanner requires no electrical registration—its performance is predictable and drift-free. No routine lineup procedures are needed.

CCD scanners are limited to a fixed form of scanning related to the CCD array so that many of the parameters of film-to-video transfer are predetermined, adjustable only by subsequent electronic processing. It is worth noting that the lens used in the scanner only has to form a narrow slit of light across the film, eliminating many of the problems encountered in camera-type telecines.

Basically the CCD scanner is a stand-alone, self-contained apparatus incorporating the film transport, image sensing and color separation mechanisms, and an electronics module with monitoring and control facilities. The old camera-type multiplexer has been abandoned in favor of what is now known as "electronic multiplexing." Several film transports, each with its own servo and a switchable 35/16mm optical block, can be operated from a central film deck control panel. panel.

The first CCD telecine designed specifically for broadcast use is the Rank Cintel ADS 1 digital scanner. A vertical transport orientation brings the film transport within easy reach of the operator. Another feature is a built-in automatic sequencer allowing direct entry of start, stop and on cues for three transports. An even more important innovation is automatic electronic cancellation of dirt and scratches on the film. This is accomplished by taking advantage of the sensitivity of the CCDs to infrared radiation. Color film dyes are transparent to infrared while dirt shows up distinctly. By utilizing a fourth CCD array, the telecine can be made to detect defects automatically; then image processing techniques are employed to cancel the defects.

The Marconi B3410 line array scanner is claimed to be the world's first telecine with full digital processing, a major move towards the goal of eliminating time-consuming adjustments. This scanner has full logarithmic masking and true power law gamma correction, as well as aperture correction. The telecine can be operated by automatic control or with a preprogramming system called PREFIX, controlled from a dedicated panel or a keyboard, with operating data recorded on a floppy disk. This system can be used also as a search-and-find method to give quick turnaround in running film segments.

The Bosch FLD 60 scanner has been designed for multipurpose telecine use—that is, for direct on-air broadcasting of film as well as for the much more demanding requirements of program post-production. An optional A/B interface allows two scanners to be operated from one film deck control panel. A wet gate has been developed for the scanner to hide scratches on the film surfaces, similar to the equipment used in motion picture printing machines. By means of a secondary color processor, hue, saturation and luminance can be varied in six sectors independently. A film grain reducer compares film frames for uncorrelated content and grain is cancelled out electronically.

A film reproduction programmer is also available for use with this telecine. The programmer is a microcomputer-controlled color correction system in which the controlled values for color correction can be recorded on a floppy disk for later use.

Color operation with the CCD film scanner was first introduced in the early 1980s, for use in the European television system. Subsequently, digital picture storage enabled store conversion of sequentially scanned signals from the arrays into 625-line, 50-field or 525-line, 60-field.

Further development of solid-state scanning technology can be confidently expected, taking into account the simplicity of the signal-generating system in comparison with other methods employed in the past. No doubt existing camera-type telecines will continue to be used for some years to come, and it will be difficult to displace the flying-spot scanner in the most demanding applications, but the solid-state scanner in improved and expanded versions may eventually replace all other methods for making television pictures from film.

SETUP
(PROCEDURES AND ALIGNMENT)

In the earlier days of television broadcasting there was a great deal of controversy over the reasons why the television pictures from film compared unfavorably with the pictures from live studio cameras. Then a gray scale on film, mounted in a 2 × 2 inch slide, was produced and projected into the cameras of multiplexed telecines. In this way it was found that the video waveforms varied considerably from one telecine to another, and everyone involved in the tests agreed that a standard telecine setup procedure would eliminate a major cause of unsatisfactory film performance.

A test slide with cross-step gray scales, 3¼ x 4 inches in size, for use at the field lens position in multiplex telecines, was produced by Eastman Kodak Company and made available to television stations. This slide was made by depositing metal on glass; the gray scales had seven steps increasing in density from 0.30 to 2.35, in the center of a uniform gray background. The densities of the intermediate steps in the gray scales were so arranged that in a properly adjusted telecine a linear stair-step display would be obtained on the waveform monitor, with the lightest step set at 100 IRE units and the steps with highest density at black level (normally 7.5 units).

Subsequently, the Society of Motion Picture and Television Engineers submitted to the Standards Committee a Recommended Practice RP27.7— 1972, "Specifications for Gray Scale Operational Alignment Test Pattern for Telecine Cameras," which was adopted. The Kodak test slide currently available conforms with this recommended practice.

An appendix to the recommended practice describes the use of the test slide in setting up telecines. Gamma correction circuitry can be adjusted with the aid of the slide by setting the steps at predetermined signal levels on the waveform monitor. In a color telecine the transfer characteristics of the three color channels can be compared and adjusted for neutral color balance. The test slide also enables correction circuits to be adjusted for minimum shading.

Television stations were slow to adopt the standard telecine setup procedure using the test slide and the accompanying recommended practice mainly because of the variability in light-transmitting characteristics of motion picture films supplied to the stations for broadcasting.

Designers of telecine equipment proposed that video signal levels could be maintained automatically by electronic controls but surveys showed that the control devices would have to provide a range of about 100:1 in peak signal level. It would be much easier for everyone, they suggested, if films could be made with greater uniformity in minimum and maximum densities.

The concept that films being produced for television should be made deliberately with highlight and shadow densities matching the steps of the cross-step gray scale tests slide started a trend that resulted in an entirely new approach to film making and telecasting. By making sure that each scene had some light areas to generate peak white signal levels in telecine, film makers found that it was much easier to achieve the desired artistic effects, and these effects would then be preserved in the television pictures.

The change-over to color broadcasting greatly accelerated this trend. At the same time, the telecine setup procedure had to be extended to include the color picture monitor. This proved to be much more difficult. Many different methods and types of measuring equipment have been tried, but it seems that none of these has given entirely satisfactory results. A survey by the Eastman Kodak Company showed an evident lack of consistency in the primarily subjective methods used to adjust picture monitors.

In the drive to improve film performance in television, a group of television engineers at the Canadian Broadcasting Corporation in Montreal proposed that the television pictures obtained from film in telecine should match the directly projected pictures as seen in a specially constructed television viewing room, simulating typical television viewing conditions. Practical demonstrations of this concept used two identical copies of the SMPTE Subjective Reference Test Film, one in a projector with a high-intensity light source and the other in a telecine with appropriately modified colorimetry, set up with a gray scale test slide. Comparison showed that a good match could be obtained between the optical and electronic displays.

The European Broadcasting Union (EBU) has published recommendations in a document, Tech 3096, giving specification for viewing conditions for the appraisal by means of optical projection of color films intended for television presentations. Para. 2 of this document states that "broadcasting authorities should aim to provide a standard of telecine performance such that any film which appears to be of good technical quality when evaluated under the special optical viewing conditions can also be expected to be of good technical quality when transmitted by color television."

In a statement by the vice-president for television affairs of SMPTE, in the same issue of the Journal, the position of telecine color balance was that American practice relied upon a standardized neutral balance adjustment using a neutral gray scale in the telecine camera field lens and placed the prime responsibility for color uniformity upon the film producer and the processing laboratory. This meant, in effect, that telecines were expected to operate unattended, with automatic signal level control, after having been set up with the standard cross-step gray scale test slide.

Most of the older models of multiplexed vidicon telecines still in use have a means for inserting the gray scale slide in a holder at the field lens position. To set up the telecine, it is necessary only to move the slide into the light path from one of the projectors and then make appropriate adjustments of camera controls to obtain the desired waveform display. The automatic signal level controls can then be turned on for normal operations.

In these conditions the pictures from old, faded motion pictures may not be entirely satisfactory, but if station operating conditions permit, manual adjustment of camera controls usually will result in significantly better television pictures. Later on, during a short break, the telecine neutral balance can be restored by reinserting the test slide in the projector light beam and resetting the camera controls to their original positions.

New RCA Camera-Type Telecines

New models of telecine cameras available from RCA—TK29 B and C, and TK 290—include automatic features for control of white level, black level, color balance, gamma balance, and flare corrections. In addition the TK 290 electronics can be set up automatically by a touch of a button, utilizing the same setup terminal as the TK 47 studio camera. All of these new telecine cameras have sealed non-telecentric optical systems with a field lens in two sections, and a holder which allows the test slide to be inserted into the air gap.

This arrangement allows the optical systems in these new camera type telecines to be set up with the cross-step gray scale test slide in accordance with the SMPTE recommended practice for neutral color balance in preparation for unattended operation.

CCD Line Array Telecines

The CCD line array film scanners available from Rank Cintel (ADS 1); RCA (TKS 100); Marconi (B3401), and Robert Bosch (FDL60), are far simpler in design and operation than the older multiplexed camera-type telecines. A central electronics module can be used to control the operation of two or three film scanners, while continuous video output is maintained by electronic multiplexing. Interchangeable optical blocks allow either 16 or 35mm film to be reproduced in the scanners. A condenser lens forms a narrow slit of light across the width of the film, and sequential scanning of film frames takes place as the film moves continuously through the gate.

The standard 3¼ x 4 inch cross-step gray scale test slide cannot be reproduced in the optical system of a CCD scanner, nor is there any way for handling 2 x 2 inch slides, even if the gray scale pattern could be made available in this format.

All of the CCD scanners are designed for unattended operation with automatic control of video levels, automatic color correction and automatic correction of optical shading. Matrixing, film masking and gamma correction are usually performed digitally, with the goal of eliminating time-consuming adjustments.

SURVEY OF NEW GENERATION TELECINES

Eastman Kodak Company has made a survey of 20 production facilities to evaluate the picture quality from color film through the new generation CCD and flying-spot scanners. In preparation of the survey a telecine analysis film (TAF) was developed, including an eight-bar color test pattern, a neutral density scale and a constant density surround. The gray scale was intended to be used to adjust subcarrier balance and black-and-white levels, while the color bars help to evaluate system colorimetry and other factors affecting hue and saturation, using a vectorscope. Along with the TAF, Kodak supplied subjective picture test films consisting of scenes on Eastman color negative film and a color print from the negative.

Several different methods were being used to adjust telecines in these production facilities, including open gate, "China Girl," film D-min, black-and-white test film and test materials fabricated in-house. Acceptable picture quality was being obtained from the telecines, but considerable differences in gray scale rendition and color quality were noted. The biggest differences were in the shadow areas of scenes. The TAF color bars as displayed on a vectorscope showed that the red bar was reproduced in a fairly consistent manner, but striking differences were observed in the phase angles of the green and blue bars.

This survey showed that there is a need for a telecine analysis film; the TAF film should fill that need. Very evident in the survey was a lack of consistency in setting up color picture monitors. Most of the methods being used were subjective.

CORLEY TELECINE TEST FILM (TP2)

A new 16mm test film for evaluating telecine performance is available from DSC Laboratories in Rexdale, Ontario. This test film was produced in cooperation with the Development Department of the Canadian Broadcasting Corporation in Montreal. The methods used to make the film are described in a paper in SMPTE Journal, April 1982, page 361.

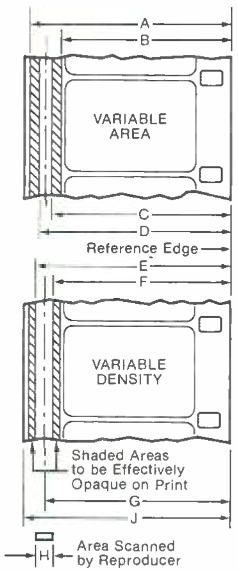
This test film has a series of patterns in two categories. The first includes colorimetry tests—white levels, gamma and black levels, and color tests of the six vector colors. The second category consists of optical and dimensional tests for shading, flare, lag, framing, focus, resolution and automatic black-and-white level tests.

The test film was designed to place the six colors—yellow, red, magenta, blue, cyan and green—in the appropriate boxes in the vectorscope display when the test film is being reproduced in a properly adjusted telecine. It should be noted that if the color-reproducing conditions, color analysis, masking, and matrixing are different than those for which the test film was developed, the test colors will not fall in the vectorscope boxes.

COLOR BARS FOR SETTING UP TELECINES

In a paper with this title in the Feb. 1978 issue of SMPTE Journal, Dr. R. W. G. Hunt of Kodak Ltd., proposed that carefully exposed strips of film arranged in sequence to produce the familiar color bar pattern on picture monitors should be used to determine optimum values of electronic masking. This proposal was based on the premise that the telecine should be considered as a complete transfer unit with the settings being done against the only relevant standard, the human eye.

In the research project by Canadian Broadcasting Corporation reported in the SMPTE Journal, March 1969, "An Engineering Approach to Color Telecine," it was proposed that the reproduction of color film in telecine should match visually the directly projected pictures as seen in a review room simulating television view-



Dimensions	Inches	Millimeters
A	0.611 max	15.52 max
B	0.513 ref	13.03 ref
C	0.540 ± 0.002	13.72 ± 0.05
D	0.600 ± 0.002	15.24 ± 0.05
E	0.610 ± 0.002	15.49 ± 0.05
F	0.530 ± 0.002	13.46 ± 0.05
G	0.570 ref	14.48 ref
H	0.071 ref	1.80 ref
J	0.628 ref	15.95 ref

Fig. 3. Standard location and dimensions of 16mm sound tracks.

ing conditions. The colorimetry of the vidicon camera telecine used for this project was modified to enable a near-perfect match to be obtained.

Another paper in the March 1984 issue of the Journal of the British Kinematograph, Sound and Television Society (BKSTS) by A. M. Harcourt of Kodak Ltd., "Film and Electronics Interface," shows (in Fig. 2 of the article) typical spectral response curves for flying-spot and CCD telecine sensors. When the spectral response curves for the typical CCD scanner are superimposed on the curves for the vidicon telecine used in the CCD research project (March 1969 SMPTE Journal), it can be seen that the red and green curves match fairly well, but the blue curve for the CCD scanner is displaced towards the green.

Research by BBC on color operation of line array CCD telecines (see SMPTE Journal, Feb. 1980, page 100) showed that chromaticity shifts in the reproduction of a series of nine test colors could be almost completely corrected by suitable matrixing.

As one writer on this fascinating subject of color reproduction in the television system has said, "The aim should be for trouble-free transference of images from one excellent medium—film—to another image maker—television—in such a way that the eye brain 'sees' it as real."

FILM SOUND

By far the most common film format in television stations is 16mm. Many of the television programs coming into the stations from distributors are in the form of 16mm prints with optical sound tracks along one edge.

An optical sound track usually consists of a kind of recorded waveform with a high density envelope, varying in area with the amplitude and frequency of the sound. Sometimes a film may have a sound track made up of transverse striations, known as variable density track.

Generally speaking, the quality of the sound from optical tracks on 16mm film is quite good. But sound quality can be adversely affected by a variety of faults in playback. The standard location and dimensions of the sound track on 16mm film is shown in Fig. 3.

Sound Track Playback in Camera Type Telecines

In the film projectors used in camera-type telecines, the sound reproducer is located just below the picture gate. An optical system focuses a slit of light from the exciter lamp on the film. Usually this lens is adjustable to accommodate films with the emulsion side—and the sound track—either towards the light source of the projector or towards the lens.

A system of damping rollers smoothes out the movement of the film over the sound scanning drum, following the intermittent frame-by-frame movement of the film in the picture gate. The sound in the film track is located 26 frames ahead (in the direction of film travel) of the corresponding picture frame.

Variations in the light-transmitting properties of the sound track modulate the light beam as it passes through the film, and the light falls on a photocell generating an output which, after suitable amplification, becomes the audio signal.

The optical system in the sound reproducer must be sharply focused (Fig. 4) on the emulsion side of the film where the sound track is located for best high-frequency response. This is a critical adjustment because the focal length of the lens is quite short; a change of only one or two thousandths of an inch can cause a noticeable high-frequency loss. Usually the lens has only two positions—either for the back or the front of the film—but the focus should be checked routinely by maintenance.

If the intensity of the scanning beam across the width of the sound track is not uniform, the

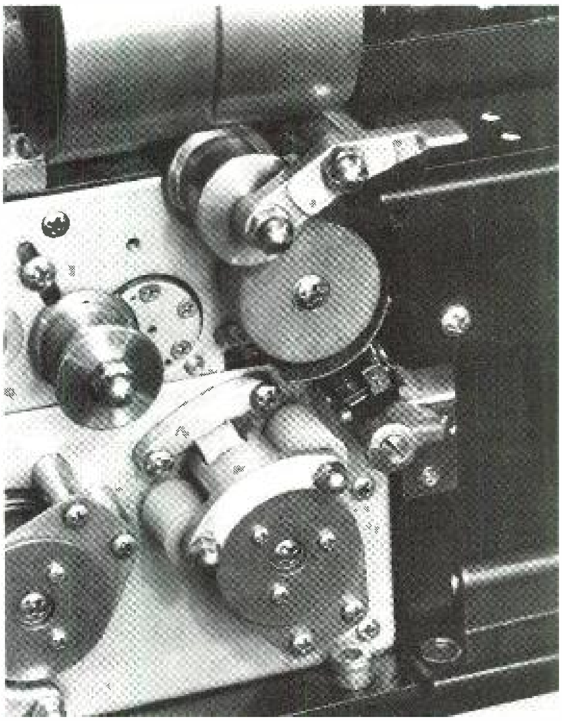


Fig. 4. A 16mm sound track scanner assembly.

reproduced sound may be distorted or low in level. Lack of uniformity is usually caused by dirt accumulating in the slit of the optical system. Incorrect adjustment of the exciter lamp filament will also adversely affect sound quality.

Many 16mm films in circulation have magnetic sound tracks. The stripe of iron oxide is coated on the side of the film facing the projector lamp. The playback head in the projector is located 28 frames ahead of the corresponding picture frame. Projectors for television service usually have selectable optical and magnetic sound playback.

The azimuth of the magnetic playback head must be properly aligned. Ideally the gap in the head should be positioned at exactly 90° to the direction of film movement. An incorrect azimuth setting results in loss of high frequencies. Head-to-film contact is vitally important in maintaining high frequency response. Periodic cleaning is essential to remove buildups of oxide or film emulsions that prevent good contact.

Sound Playback in Film Scanners

Continuous film motion in flying-spot and CCD scanners makes sound reproduction much easier, since there is no need for a system of damping rollers at the sound drum as in intermittent projectors used in camera type telecines. Fig. 5 shows the optical/magnetic sound reproducing module in the Bosch FDL60 CCD telecine.

The Bosch FDL telecine scans optical and magnetic sound tracks directly at the capstan. A

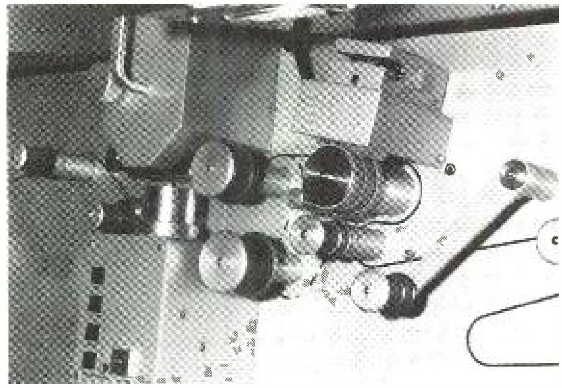


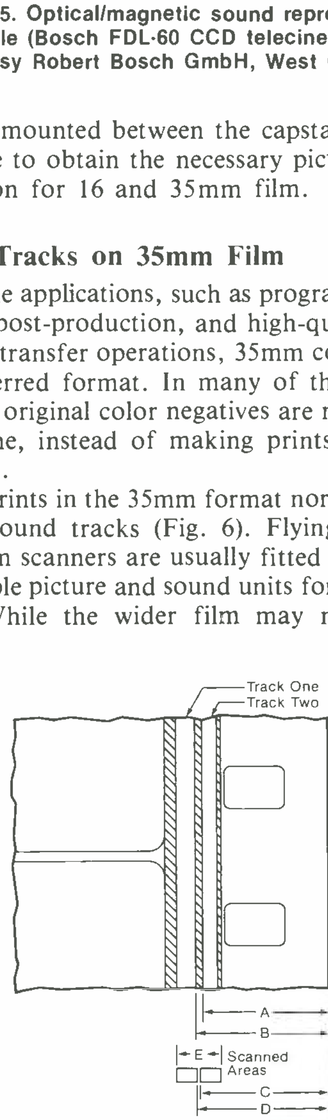
Fig. 5. Optical/magnetic sound reproducing module (Bosch FDL-60 CCD telecine). (Photo courtesy Robert Bosch GmbH, West Germany)

roller is mounted between the capstan and picture gate to obtain the necessary picture/sound separation for 16 and 35mm film.

Sound Tracks on 35mm Film

In some applications, such as program packaging and post-production, and high-quality film-to-video transfer operations, 35mm color film is the preferred format. In many of these operations the original color negatives are reproduced in telecine, instead of making prints from the negatives.

Film prints in the 35mm format normally have optical sound tracks (Fig. 6). Flying-spot and CCD film scanners are usually fitted with interchangeable picture and sound units for 35/16mm films. While the wider film may not be en-



Dimensions	Inches	Millimeters
A	0.238 ± 0.002	6.05 ± 0.05
B	0.248 ± 0.002	6.30 ± 0.05
C	0.242 ± 0.001	6.15 ± 0.03
D	0.244 ± 0.001	6.20 ± 0.03
E	0.084 ref	2.13 ref

Fig. 6. Sound track location in 35mm print.

countered to any great extent in television stations, the new generation telecines are used increasingly in operations where these two film formats may have to be intermixed.

In postproduction operations 35mm picture negatives give the best television picture quality. Normally the negatives do not have sound tracks; if sound accompanies the pictures, it will be on a separate magnetic film.

Special equipment is needed to interlock sound playback from a magnetic film with the corresponding negative or positive picture film in telecine.

Optical and Magnetic Test Films for 35mm and 16mm Sound Reproducers

The Society of Motion Picture and Television Engineers can supply a considerable number of test films for checking and adjusting telecine sound reproducing equipment. Among these are:

16 and 35mm Optical Tracks

Buzz Track—300 and 1000 Hz tracks on opposite sides of a central strip for checking lateral positioning of the scanning beam.

Flutter—A sound recording which should reproduce at 3150 ±25 Hz.

Scanning Beam Uniformity—A 1000 Hz recording moving laterally across the track.

Sound Focus—A 5000 Hz recording for 16 mm and 7000 Hz for 35mm for focusing scanning beams.

16mm and 35mm Magnetic Tracks

Azimuth—A sound recording at 7000 Hz for 16 mm and 8000 Hz for 35 mm for checking and adjusting azimuth setting of playback head.

Flutter—A sound recording which should reproduce at 3150 Hz ± 25 Hz.

Multifrequency (16mm only)—Recordings at 15 different frequencies from 15,000 to 50 Hz, with a 400 Hz reference section.

SMPTE Test Materials for Motion Pictures and Television, a catalog listing these and many other test films is available without charge by writing to: Society of Motion Picture and Television Engineers, 862 Scarsdale Avenue, Scarsdale, New York 10583

STILLS

Broadcasters have always made extensive use of 2 x 2 inch (35mm) slides in television programming. Multiplexed camera type telecines usually include a slide projector. One popular type of projector has a dual-channel optical system and two slide drums, by which a transition from one slide to the next takes place in the form of a very quick dissolve, by remote control from the telecine operating console. With this projector, lettering can be superimposed over a television picture, and slides can be used as a simple, inexpensive method for commercial advertising, with a series of slides being shown, one after the other, at any desired rate, supplemented by audio and video, if necessary, from another source.

Making slides on color film is the easiest and least costly method for obtaining color television pictures. For television service, registering slide mounts ensure uniform positioning of lettering in the television pictures.

Storage and Retrieval

In addition to 2 x 2 inch slides, television stations make extensive use of graphics and still pictures reproduced by live television cameras. Storing, searching, and retrieving slides and graphics are time-consuming, tedious operations; yet some stations stock thousands of slides and graphics.

Methods have been devised for the electronic recording, storage, retrieval, and broadcast of slides and graphics (stills, as these picture sources are known). For this purpose, digital recording techniques are used. The digital store may be a multi-surface disk pack drive and interchangeable disk packs.

At the same time, methods have been developed for generating graphics electronically, and these electronic stills can be stored in the disk packs as well.

Computer graphics have replaced much of the work being done by artists at the drawing board, but the creativity and dexterity of the artist are irreplaceable elements in producing illustrative materials. Similarly, there is no really practical alternative to the use of the easy-to-operate, inexpensive 35mm single lens reflex film camera for obtaining sharp, colorful still pictures. These pictures can be in the form of 2 x 2 inch slides, used as is, enlarged to produce color prints, or combined with other illustrative materials and transferred into an electronic still store.

Reproducing 2 x 2 inch Slides

By far, the best way to generate electronic pictures from 2 x 2 inch slides is in a telecine. The slide projectors in the old multiplexed camera-type telecines are so versatile and easy to use that this is still the preferred transfer method in many stations.

The Rank Cintel flying-spot telecine has a programmable slide-scanning facility in which the slides are held stationary in the interchangeable slide gate. Although not quite as easy to use as

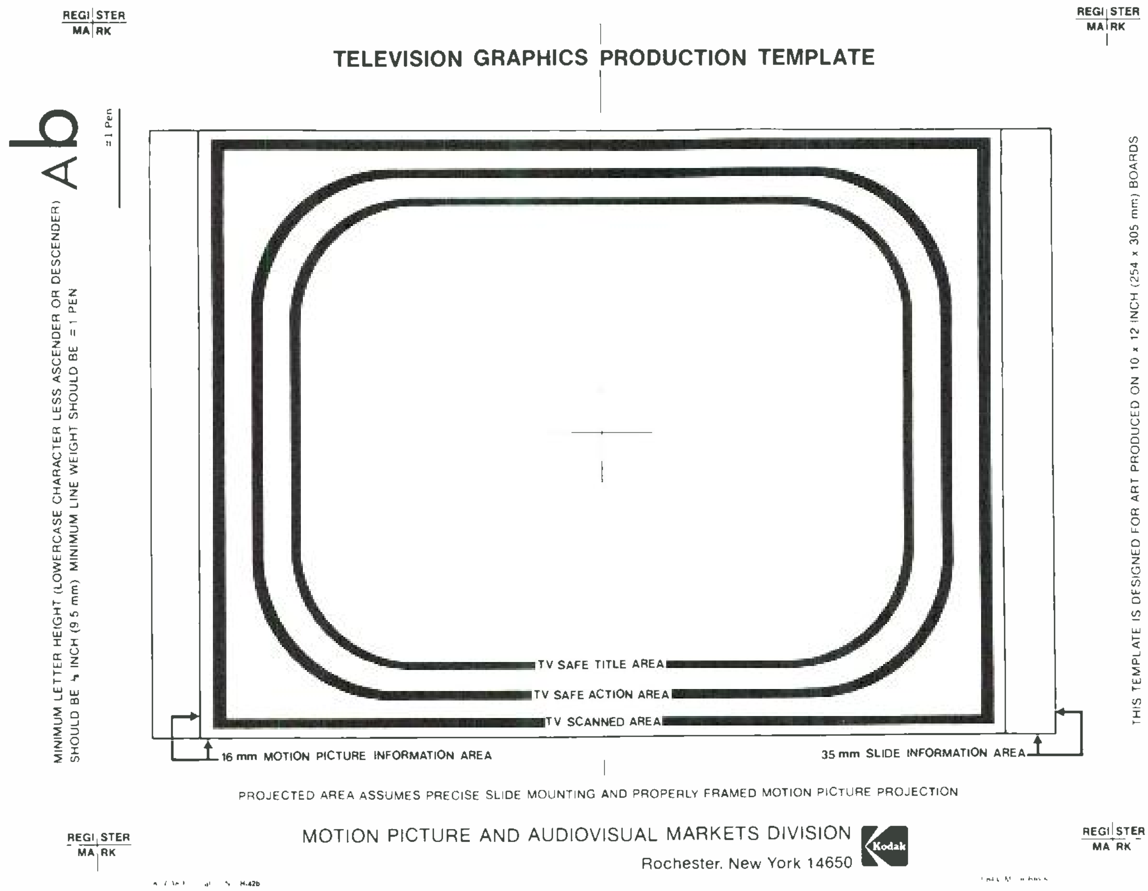


Fig. 7. TV Safe Title Area.

a camera-type telecine projector, it is capable of generating greatly enhanced color television pictures. The company also has developed an electronic slide file that can be programmed with their Amigo scene-by-scene telecine control system.

So far as is known, none of the currently available CCD line array telecines has the capability for reproducing still pictures. In these telecines single-line scanning is utilized and continuous movement of the film is needed to trace out complete television picture frames. To reproduce a 2 x 2 inch slide, the slide would have to be moved in the picture gate at the same rate as normal film movement while scanning is taking place.

Safe Title Area for Television

In the production of 2 x 2 inch slides for television, important picture information such as lettering and commercial messages should be held within the area of the television picture that will be reproduced by the majority of home receivers. Fig. 7 shows the safe title area specified in SMPTE Recommended Practice RP9-1977.

This diagram shows the areas of the slide frame cut off at each stage between the original exposure of the film in the camera to the reproduction of the picture on the average home receiver. Unless these area losses are taken into account in making the original slides, parts of objects, letters, titles or numbers may be lost beyond the edges of the mask in home receivers.

RECOMMENDED PRACTICES FOR TELEVISION FILMS AND SLIDES

Many papers have appeared in the *SMPTE Journal* with proposals for improving film reproduction in the television system. Of particular concern have been some means for ensuring uniform film reproducing conditions in telecines. This work was started long before the introduction of the new generation of film scanners, in response to complaints by motion picture makers that they were being asked to make films with different light-transmitting characteristics to conform with different telecine reproducing characteristics.

Surveys by Eastman Kodak Company indicated there was a need for an operational gray scale slide with known transmission values for use at

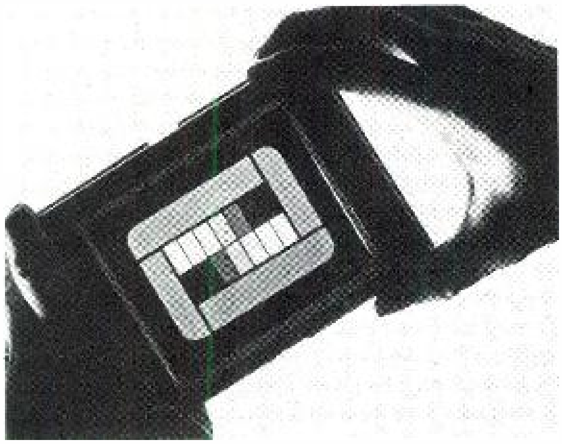


Fig. 8. Cross-Step Gray-Scale Slide (Photo courtesy of Eastman Kodak Company.)

the field lens position in multiplexed telecines, to enable balancing and shading adjustments to be made in each of the projector inputs. It should be possible also to make use of the slide for adjusting tracking and gamma.

In response to these requests, a test slide was produced (Fig. 8) and made available to television broadcasters using a metal alloy evaporated on glass. It is stable, achromatic, and non-scattering. This slide has seven steps in two rows, increasing in density in opposite directions, in the center of a uniform density surround.

In a properly adjusted vidicon telecine this slide should produce a cross-step waveform display as shown in Fig. 9.

With the aid of this slide the gain control of the camera can be set to display the minimum density step at 100 IRE units on the waveform monitor, while the maximum density step is placed at blanking level, normally 7.5 units, by adjusting the black level control.

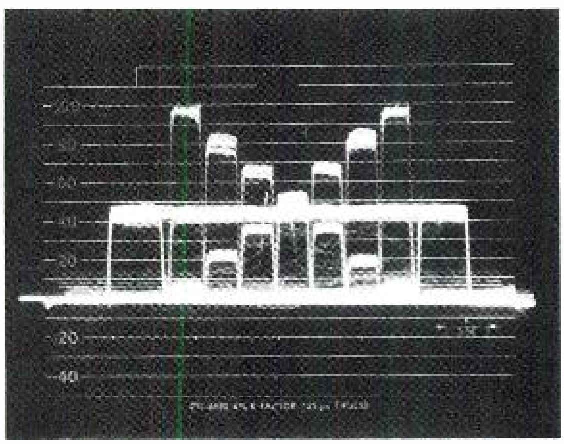


Fig. 9. Video waveform showing correctly set up cross-step gray-scale slide.

Faults in the setup and operation of the telecine show up in the waveform display from the slide as departures from the optimum waveform, as shown in the above illustration. For example, a nonlinear diffuse display of the uniform density surround indicates there is a shading problem, while blurred waveform steps are the result of poor tracking in the three color channels. In telecines where the waveforms from the three color channels can be displayed simultaneously side-by-side, tracking and gamma adjustments can be made easily.

This slide, designed to produce a linear waveform display, is an invaluable aid in setting up telecines in a precisely reproducible manner day after day. When the settings of the telecine camera controls have to be changed, as in making manual corrections of reproducing conditions for films that do not conform with density and color specifications, it is quite easy to restore the telecine to its normalized operating condition by inserting the test slide in the projector light path and re-adjusting the camera controls to obtain the normal waveform display, as shown in the illustration above.

One of the most interesting and useful side effects of using the gray scale slide for telecine alignment is that, if a normal waveform display cannot be obtained by varying the camera controls, this would indicate the existence of a serious maintenance problem. The slide can also be utilized very effectively to evaluate the effects of maintenance adjustments.

SMPTE Recommended Practice RP27.7-1972

As a direct result of this work by Eastman Kodak Company, the Society of Motion Picture and Television Engineers published Recommended Practice RP27.7-1972—"Specifications for Gray Scale Operational Alignment Test Pattern for Telecine Cameras." This recommended practice states that the purpose of the test pattern is to check:

(a) Light signal transfer characteristics of a television camera,

(b) Signal compression or clipping in the video signal channel,

(c) Operation of gamma correction circuitry,

(d) Operational setup and balance of gain and black level controls,

(e) Amplitude tracking among the video signal channels of a color television camera.

Broadcasters too became involved in devising methods for telecine alignment that would ensure uniformity of film reproducing conditions. The outcome of one of these undertakings was a telecine test slide produced by D & S Corley Ltd.

on color film material, also for use at the field lens position in multiplexed telecines. This slide has cross-step patterns similar to the Kodak slide, but with only five steps in each direction. The objective in making this slide on color film was to produce a test object with characteristics similar to the film materials being used to produce television programs.

Recently D & S Corley has produced a telecine test film TP2, designed to verify the accuracy of telecine alignment and the uniformity of film reproduction through the television system. This test film is produced on 16mm color print stock and contains several test sections.

Densities of Color Films and Slides For Television

As a guide for the makers of films for use in television, SMPTE published another Recommended Practice RP46-1972, "Density of Color Films and Slides for Television." This recommended practice specifies that the density of television white level should be 0.30 to 0.40, corresponding to fully lighted objects of a scene with a reflectance of about 60 percent. This will result in the reproduction of fully lighted human faces with reflectances of 35 to 15 percent at densities of 0.20 to 0.50 greater than television white.

At the other end of the scale, dark or black areas in which detail is not essential may have densities in the order of 2.50.

Subjective Reference Test Film

SMPTE has also produced a color television subjective reference test film, available in both 35 and 16mm formats. This film is intended for two separate purposes. First, it should serve as a subjective reference print in the motion picture laboratory for the subjective evaluation of color balance and density of prints being made for color television transmission, enabling the color timer to correct the overall balance of prints for a good subjective match with the test film. This use of the test film should result in more consistent balance of color prints for color television. Second, the test film should be used by broadcasters for the subjective evaluation of the performance of a color television film transmission system after the system has been properly set up.

The test film has a running time of about 4 minutes and contains a series of interior scenes including samples of day and night lighting.

Television Film Review Room

Another particularly interesting development has been the design of a review room for color television film evaluation, complementing the steps taken by motion picture makers and television broadcasters to improve film performance as a source of television pictures. A Recommended Practice published by SMPTE, RP41-1974, "Evaluation of Color Films Intended for Television," represents a major advance in this direction.

Recommended Practice RP41 specifies conditions that provide a standard correlate of the ambient field of view of the television screen which affects apparent picture contrast, and a constant adaptation field against which the color balance of the projected picture can be referred when making color balance judgments.

Setup of Color Picture Monitors

The setup of color monitors used for evaluation of color television pictures was another area in which improvements in uniformity were urgently needed. SMPTE Recommended Practice RP71-1977, "Setting Chromaticity and Luminance of White for Color Television Monitors using Shadow Mask Picture Tubes," specifies adjustment procedures using either a visual comparator or a light measuring instrument.

Alignment of Flying-Spot and CCD Line Array Scanners

The introduction into North American television of flying-spot and CCD scanners has raised once again the questions of how these telecines should be set up and operated to produce the best possible television pictures from film. A survey by Eastman Kodak Company has shown that good quality pictures are being produced, but there is a considerable amount of variability in gray scale and color reproduction. At the same time, users of these telecines indicated that there is a need for a test film for telecine analysis.

COLOR TELEVISION TEST FILMS AND SLIDES

Over the years, many different test films and slides have been used in adjusting and setting up television film reproducing equipment. Of these, two gray scale slides are still available, and two new telecine test films have been produced.

Cross-Step Gray-Scale Slide For Telecine

The cross-step gray-scale slide for telecine alignment developed by Eastman Kodak Company, as described in SMPTE Recommended Practice RP.7-1972, has seven steps increasing in transmission over a 40:1 range, following a 2.5 power law related directly to the gamma of the television picture tube. A value of 50 percent transmission ($D=0.30$) was selected for the lightest steps of the two wedges, corresponding to the recommended density for film picture highlights (see RP46-1972). Step seven, the darkest step, has a light transmission of 0.45 percent ($D=2.35$).

The progression of transmission values for steps one through six follows the 2.5 power law over a contrast range of 40:1, appearing as a smooth step-by-step progression from near black level to reference white level. The transmission of step seven was chosen to fall half an increment below step six on a 2.5 power law curve. This provides an additional contrast range relative to step one of slightly more than 100:1, but tonal gradations (picture details) between steps six and seven may not be reproduced properly due to black compression and gamma correction circuit limitations.

The slide was designed to produce a crossed staircase display on the telecine waveform monitor. This enables gamma correction to be adjusted by setting the steps at predetermined levels. The test slide can be used also to compare the transfer characteristics of the three channels in a color camera for neutral color balance, an essential condition in telecine setup. Another use for the test slide is to check black clipping on step seven as the blanking controls are adjusted. Normally step seven should be set at blanking level. Shading correction can also be set with the test slide, using the neutral density background in the slide as a reference.

Corley Telecine Alignment Slide

The original Corley telecine alignment slide was designed to transmit essentially equal energy RGB, and its neutrality was tested in a color telecine that had been set up to neutral by perforated metal attenuators to give equal red, green and blue signal levels. The objective in making the slide on color film stock was to produce a test object with characteristics similar to the film materials being reproduced in telecines.

In a properly adjusted telecine the five staircase steps of the slide should appear at 20, 40, 60, 80 and 100 IRE units on the waveform monitor in the red, green, and blue camera channels.

These slides were numbered and dated before delivery to users, with a recommendation for periodic replacement to avoid the effects of fading or deterioration.

Later on, in 1981, Corley produced a new series of test objects for the alignment and evaluation of telecine colorimetry. This series consists of Corley's TA2 gray scale slide, a CB2 color bar slide, an MT multitest, and a flare slide, size 3¼ x 4 inches, designed for use in the optical system of multiplexed telecines and a set of six 2 x 2 inch slides including all of the above patterns, plus a depth-of-modulation slide and a TC test chart.

Eastman Kodak Company's Telecine Analysis Film (TAF)

The telecine analysis film produced by Eastman Kodak Company in preparation for a survey of new generation telecines consists of an eight-bar color test pattern, a neutral density scale, and a constant density surround. To produce the TAF a special target transparency was photographed on Eastman color negative film. Sequential red, green and blue separation exposures of the tungsten-illuminated target were made through Kodak Wratten gelatin filters (R:29, G:99, B:98) using a pin-registered camera. The gray surround of the target is 0.75 in density, and overall film exposure is adjusted to obtain specified laboratory aim density values in that area of the processed negative. The target white is clear, the black is opaque, and the intermediate gray scale steps range in density from 0.30 to 1.8 in six 0.30 increments. Each color bar is produced by appropriate combinations of either 0.30 or 1.20 density silver neutral-density filters during the separation exposures. For example, the red bar is produced with a 0.30 density filter during the red exposure, and a 1.20 density filter during the green and blue exposures.

The TAF color bars are not intended to match electronically generated color bars, but may be used to evaluate system colorimetry and other factors affecting hue and saturation using the vectorscope.

To assist in obtaining optimum television picture quality from film, TAF is being made available on 16 and 35mm Eastman color negative film; prints from these negatives on Eastman color print film and Eastman Ektachrome video news film.

Corley 16mm Test Film for the Evaluation of Telecine Performance

The Corley 16mm test film, TP2, was designed to test the accuracy of film reproduction through the television system. The first section of the film includes tests of gray scale and color reproduction (colorimetry test); the second section includes all of the other optical and dimensional tests such as shading, flare, lag, framing, focus and resolution, and automatic white and black level tests.

Included in the colorimetry tests are white levels, gamma and black levels, as well as color tests of the six vector colors. To produce the test film a mathematical model of the ideal telecine was created, and the model was used to design the test colors, producing outputs that could be easily evaluated.

In the section containing optical and dimensional tests, the shading test consists of a 70 IRE uniform gray field; the flare test has a 100 IRE background with five black squares; the lag test has 10 diagonal traverses of a 100 IRE rectangle on a black background, and the framing, focus and resolution test chart has five bull's eyes, one in the center and one at each corner. The automatic white and black level test consists of five split field patterns in which the telecine's automatic adjusters should raise or lower white and black levels in a minimum time.

Television Film Review Room

The purpose of the television film review room is to create a viewing ambient where films intended for television can be evaluated by optical projection; that is, the directly projected pictures in the review room should match the appearance of the electronic pictures from a telecine and its associated picture monitor that have been set up in accordance with the specifications in applicable recommended practices.

SMPTE Recommended Practice RP41-1974 specifies that screen chromaticity and spectral distribution of the open gate screen in the review room should be approximately 5400°K, while the luminance of the screen should be about 40 foot lamberts. In these conditions a film conforming with Recommended Practice RP46-1972 should produce a white luminance of about 20 foot lamberts, corresponding approximately to peak white luminance of color television picture monitors at that time.

The viewing screen in the review room should have a uniformly illuminated surround at about 1/10 open gate screen illumination, and approximately eight times the viewing screen area.

FILM CLEANING AND HANDLING PRACTICES

In the normal operation of a television station it is not easy to ensure that all films are spotlessly clean and free from physical defects. The methods of film handling adopted by television stations and the degree of tolerance for image imperfections that broadcasters allow sometimes aggravate these conditions.

Film Cleaning

When film and slides are being reproduced in the television system, any particles of dust and dirt as well as scratches and abrasions on the film surfaces produce most unpleasant effects in the television pictures. And electronic image enhancement invariably enhances the picture defects as well.

The large network centers rarely broadcast a scratched or dirty print. Few stations seem to have the time, however, to clean every film prior to telecast, and replacements for defective prints are seldom readily available.

The preparation of film programs for on-air release or transfer to videotape usually involves much handling and rewinding, often at a high rate of speed. If the film reels are bent or damaged, the plastic base and emulsion may be scraped off the edges of the film and trapped between the layers on the take-up reel.

Even when the greatest care is exercised, film has the tendency to attract dirt particles. Smoking in film handling areas should be discouraged (perhaps forbidden is a better word here) because ashes and smoke residues soon become widely dispersed on floors, work surfaces, and eventually on the films.

Cleaning film to remove dust and dirt particles is not a difficult or time-consuming task. All that is necessary is a pair of film rewinds, already available in just about every station, along with a plush pad and a bottle of cleaning fluid (Fig. 10). The roll of film to be cleaned is placed on the left-hand rewind, and an empty take-up reel on the other. Put a few drops of cleaning fluid (Kodak movie film cleaner with lubricant or one of several other commercial products) on the plush pad, fold the pad over the film strand, and then by turning the crank for the take-up reel, pull the film slowly through the pad, exerting only light pressure on the film. Winding should be stopped several times through a 1200-ft. (½ hr.)

Fig. 10. Cleaning a film using a plush pad and a pair of rewinds.

roll, to shake out the collected dirt from the pad, and to add a little more cleaning solution. Don't forget to go back a few feet when cleaning is restarted, to remove any dirt buildup at the point where the film was stopped.

Avoid putting too much cleaning fluid on the plush pad, for if the surfaces of the film are left in a wet condition as the roll is being wound up, drying marks may appear on the film surfaces later on. Cleaning pads can be made up from pieces of plush velvet 12 x 16 inches in size, folded over once and sewn around the sides with the edges turned inwards.

The amount of time needed to clean a ½ hr. roll of 16mm film, winding the film slowly and changing the pad a few times, should take no more than about 5 minutes. Machines are available for large scale film cleaning operations. (For complete information on various aspects of manual or machine film cleaning, see "The Book of Film Care," KODAK Publication No. H-23, especially pages 78-83.)

Film Handling

Motion picture film will stand a considerable amount of abuse, but sooner or later a valuable color print may begin to show signs of damage. The most common forms of damage are scratches and abrasions on one or both film surfaces. Film can be scratched in any piece of equipment where it is drawn over metal surfaces. A tiny nick in the gate of a telecine projector can put a severe scratch on the film from one end to the other. Constant attention is needed to make sure that scratches are not being caused by faulty equipment.

Metal surfaces that come in contact with moving film are usually undercut in the central picture area, with slightly raised rails to support the edges of the film. Sometimes small particles of grit or dirt become lodged in these surfaces. Particles of film emulsion can also build up on these metal surfaces and cause scratches.

Abrasions are usually caused by careless handling, such as excessive rewinding speed or pulling on the end of a film to tighten up a loose roll. As a film is being rewound, one hand should be placed lightly on the edge of the feed reel to provide a little hold-back tension. Jerky or erratic winding is a common cause of abrasions. A particularly bad practice, often seen in television stations, is to start winding at high speed and then allowing the film reels to coast.

An alert telecine operator, noting scratches or abrasions on a film being telecast, can usually check quickly to determine whether the film has already been damaged, or if the damage is occurring in the station's equipment. As the film is running off the feed reel of the telecine projector, pick up the edges of the film lightly between two fingers and turn it towards a bright light. Even slight damage will show up as disturbances in the light reflected from the film surfaces.

With a little practice the probable cause of a film scratch can be identified by noting the nature of the damage. A straight, uninterrupted scratch parallel to the film edges is most likely caused in a machine where the film moves continuously. Projector scratches may show some discontinuities caused by the intermittent action of the claw and gate mechanism. Wavy scratches are usually caused by careless cleaning practices.

Another common source of damage is nicked or deformed perforations, usually caused by a faulty projector. Damaged perforations likely will give rise to unsteadiness in the television pictures. (Again, KODAK Publication No. H-23, "The Book of Film Care," is an authoritative guide on these and other aspects of film handling. See especially, pages 51-83 for tips on film handling.)

FILM SPLICING

Many different methods can be used to prepare films for on-air release or transfer to video tape. For example, a film program and all the short commercials, promos, etc., to fill a full half-hour broadcast period can be spliced together into a single, complete roll ready to run in a telecine projector or scanner.

Another method is to splice together all the commercials and promos on one reel, with short sections of leader between these items, and run this reel on one telecine projector or scanner while the main program is running on another. A complete on-air program or a video tape transfer can be produced by switching from one projector or scanner to the other.

Still another method is to transfer the film program and all the commercials, promos, etc. to video tape and package a complete program by electronic editing.

Whatever method is used in a station, some film splicing will always be needed. Each of the splices should be considered as a potential hazard in the telecasting of a program or in its transfer to tape. If a splice comes apart in the telecine, operating schedules will be disrupted, with accompanying losses of time and money.

Splicing film is not a difficult or complicated procedure. Good splices that will stand up to normal handling and repeated projections, running backwards or forwards if necessary, can be made in two different ways—either by joining with cement or by applying adhesive tape to the two sec-

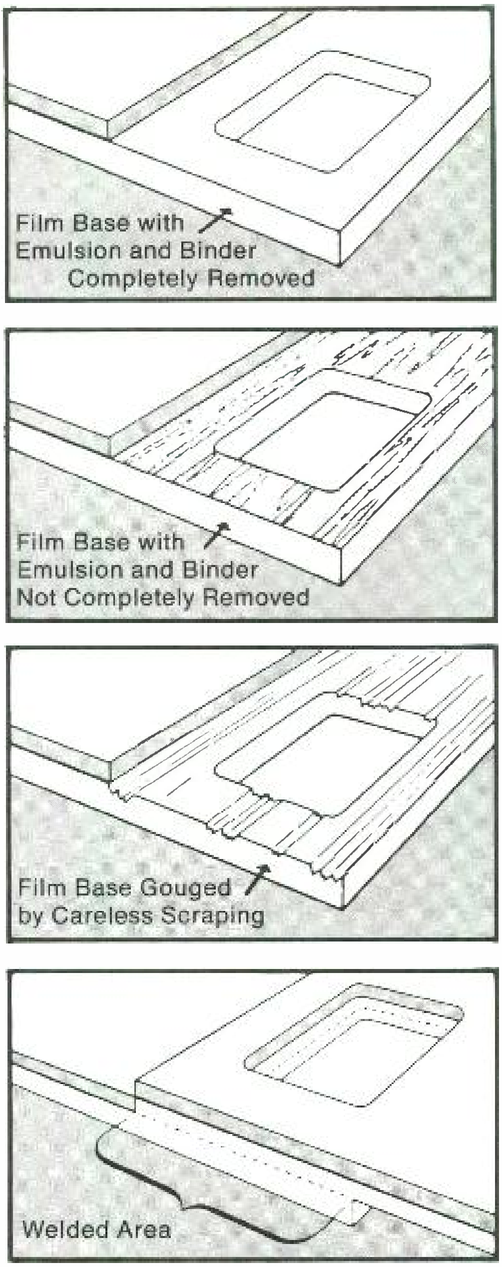


Fig. 11. Making a splice in 35mm film showing the full-hole overlap area.

tions of film at the join. (KODAK Publication H-23, "The Book of Film Care," deals with film splicing, pages 60-67.)

Cement Splices

When a cement splice is being made, the ends of the two pieces of film to be joined are overlapped (Fig. 11) in the splice area and the emulsion is removed from the lower section of film. Then a little cement is applied to the scraped area, and the base side of the other piece of film is placed over it and pressure is applied till the cement dries.

Film cement is a solvent capable of dissolving the film base.* Usually films are spliced in a small hand-operated machine, which has two hinged platens, registering pins to accurately position the two film ends by the perforations, a scraper for removing the emulsion in a narrow strip in the splice area, a knife to cut off the two ends neatly, and a hinged pressure block to hold the two ends in contact after cement has been applied. The film cement is stored in a small bottle, with a tiny brush to apply the cement sparingly in a narrow area of the film at the splice. A good splice has sufficient strength after about 20 seconds to be removed from the splicer and wound up on a reel.

Tape Splices

Tape splices are easier to make, are less likely to come apart or break during projection than cement splices, and can be made on any film base. The tape used for making splices is clear, with an adhesive backing. Many different types of film splicing tape are available.

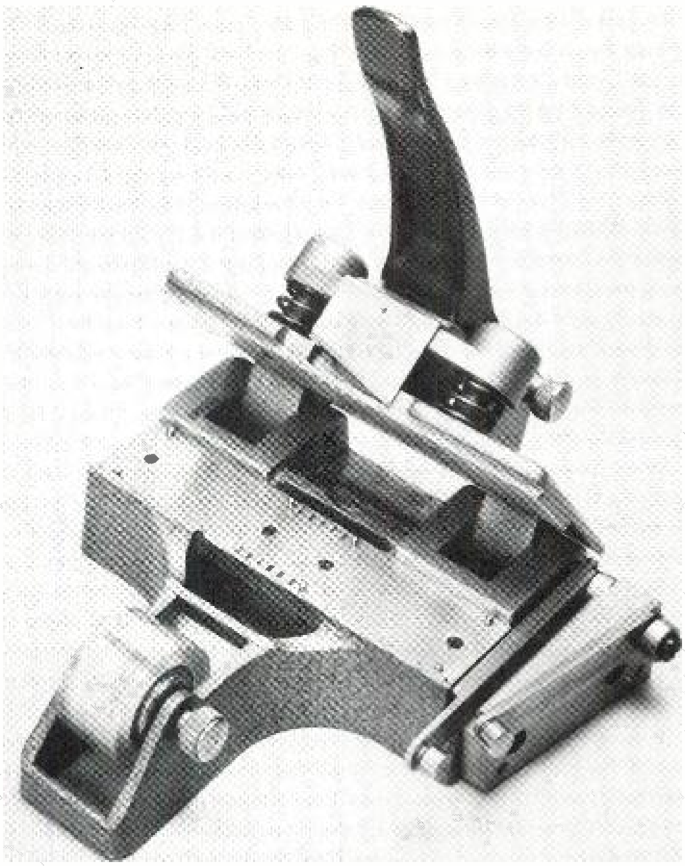
Tape splicing machines (Fig. 12), readily available, make application of the tape and finishing of the splice very easy. An important advantage of tape splices in a television station is that no frames are lost when two film are being joined together. The film ends are simply butted in the space between frames and the splicing tape applied. Another big advantage is that tape splices can be taken apart easily later on, with no damage to the film frames, by simply peeling off the tape.

STORAGE OF FILMS AND SLIDES

In a television station there is seldom any need for long-term film storage, since most of the program films coming into the station every day must be returned promptly or passed on to another station.

Film commercials, promos and the like may accumulate in a station until storage and retrieval becomes a real problem. Some stations solve this problem by transferring these short lengths of film onto video tape reels. Slides, too, require cataloging and filing unless the station has an electronic still store.

*Since no stable, commercially available solvent exists for polyester film base, cement splices cannot be made on polyester base films.



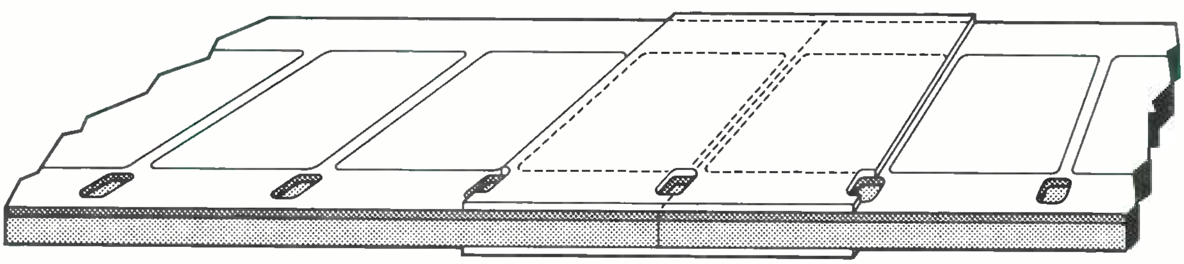


Fig. 12. Tape splicer and tape splice (perspective drawing).

Fig. 13. Film storage area in a television station. (Photo courtesy WXXI-TV, Rochester, NY.)

For those stations that must provide storage space for films and slides, a separate small room should be provided, with shelving suitably spaced for film reels, and a few drawers for slides. Metal shelving that can be assembled on the spot is easier and less expensive to install. Perhaps the best plan is to set up separate shelves or sections (Fig. 13) for 1200 ft. (½ hr.) film cans, and make use of this space for all incoming and outgoing film programs. Then, depending on the volume to be handled, a number of narrower shelves can be set up for commercials and promos. Usually these are placed in small individual 100-ft. film cans, numbered and identified for easy retrieval.

The easiest way to store slides is in cardboard or metal boxes or trays; but if a large number have to be handled, a good filing and identification system is a must to enable any one slide to be located in a hurry out of several hundred.

When Kodachrome film is sent to a central station for processing, the film can be returned uncut if desired, or the individual frames can be placed in folder-type card mounts. These mounts leave both surfaces of the film exposed and likely to pick up dust and dirt even in storage boxes. Metal slide mounts are available with the film protected on both sides by thin glass plates.

Large (1200 ft.) reels of 16mm film should be placed in cans to keep out dust and dirt and stored on the shelves upright, on the edges. Commercials and promos can be stored in any convenient manner to show the identification number. The storage area should be dry (that is, not damp) because the film emulsion absorbs moisture easily. The room should not be allowed to become too hot, as this would dry out the emulsion layer and lead to deformation. Air conditioning is needed only for long-term storage to prevent the film from becoming brittle.

The most important factor in short term storage is cleanliness of the storage area. Keeping the room clean will help to keep the film clean too and improve the station's image with the viewing public.*

*For more information on all aspects of film storage see "The Book of Film Care," Kodak publication No. H-23, especially pages 26-35.

5.9

3/4″ and 1/2″ Video Recording Systems

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3/4″ FORMAT TAPE SYSTEMS

Introduction

The U-format video tape system was developed in the late 1960's as a joint effort of the Matsushita Electric Industrial Co. (U-Vision), Victor Company of Japan (U-Tape), and Sony Corporation (U-matic). The system employs a 3/4″ cassette and records the video information using an analog component recording format known as color-under recording.

The U-format system bandwidth characteristics are less than the full-bandwidth of the NTSC system, but the U-format systems ease of use and equipment size and weight characteristics make it a desirable system for most electronic news gathering (ENG) applications and many electronic field production (EFP) applications.

Basic Definitions

The following basic definitions apply to the 3/4″ U-format helical-scan technology. They are helpful in understanding the technology employed and are adapted from SMPTE V16-861 Appendix C.

DRUM: A cylindrical column around which the tape is at least partially wrapped in such a manner as to permit rotating pole tips protruding radially from the column to form the head-to-tape interface of a video tape recording system.

SCANNER: A mechanical assembly containing a drum, rotating pole tips, and tape-guiding elements used to record and reproduce video tape recordings.

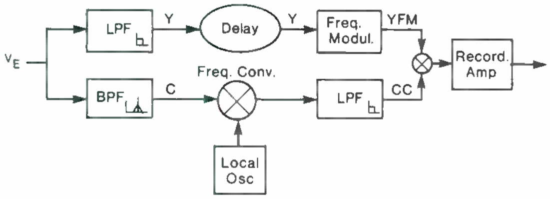
LOWER DRUM: That part of the drum in a helical-scan video tape recording system which contacts the reference edge of the tape and usually includes tape-guiding elements.

EFFECTIVE DRUM DIAMETER: A value of drum diameter which, when used in theoretical calculations, will correspond to the actual video recording produced in a helical-scan video tape recording system. The effective value is equal to or greater than the actual drum diameter.

HELIX ANGLE: The angle formed between the path of the rotating pole tips and the tape reference edge-guiding system on the scanner of a helical-scan video tape recording system.

TRACK ANGLE: The angle of a video record with respect to the reference edge of the tape in a helical-scan video tape recording system.

WRAP ANGLE: An angle that is positioned with its apex on the center of drum rotation and subtended by the line of contact between the drum and the reference edge of the tape.



V_E = encoded, composite video
Y = Luminance signal
C = Chrominance Signal
YFM = Frequency Modulated Luminance signal
CC = Converted Chrominance signal

Fig. 1. Color-under recording system.

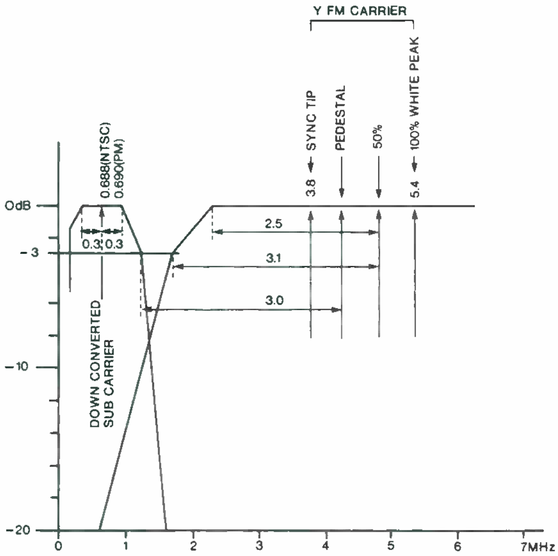


Fig. 2. Color-under frequency assignment.

Color-Under Recording

In the color-under record system, the composite color video signal is first separated into its *luminance* (Y) and *chrominance* (C) components.

The luminance (Y) component is frequency-modulated for recording.

The chrominance (C) component is heterodyned with a fixed frequency continuous signal and down-converted to a region of 600 to 800 kHz depending upon the system used. The frequency selected for the down-conversion is chosen to minimize the color-dot-crawl effects. The down-converted subcarrier (f_s) is related to the line rate (f_h) in the following manner:

$$f_s = \tfrac{1}{4}\,(2n + 1)\,f_h.$$

For U-format systems,

$$f_s = 688.374 \text{ kHz},\ n = 87.$$

After down conversion, the chrominance signal is superimposed on the Y-FM signal with the Y-FM carrier acting as a bias (Figs. 1 and 2). The Y-FM signal is recorded at an optimum level and the C signal is recorded at a level designed to obtain maximum linearity. This is generally 10 to 15 dB below the level of the FM carrier.

High frequency non-linear pre-emphasis is applied in the Y-channel to increase detail information prior to recording. A de-emphasis noise reducing circuit is applied during playback.

During playback the chrominance signal is extracted via a low pass filter and the Y-FM signal is demodulated. The chrominance subcarrier signal is up-converted to 3.58 MHz by a process that compensates for time-base error; it is then mixed with the luminance signal to produce an NTSC composite video signal.

Principles And Characteristics

A "helical-scan" architecture is used in the 3/4" format machines, in that tape unspools from the supply reel and is presented to the drum in such a manner that the heads slant across the tape to form part of a helix.

Two sets of video heads are placed 180 degrees apart on the scanner and the tape is wrapped around the scanner for slightly more than 180 degrees (Fig. 3). Each set of video heads consists of two heads, one for the channel record/playback and one for the channel erase. The scanner rotates at exactly 29.97 revolutions per second (approx. 1800 rpm) recording one TV frame per revolution and one TV field per 180 degree scan.

The 3/4" format provides, in addition to the two video channels, four longitudinal tracks, two for audio, one for control track and one for the address track (Fig. 4). The video tracks do not interfere with the address track and can be independently edited.

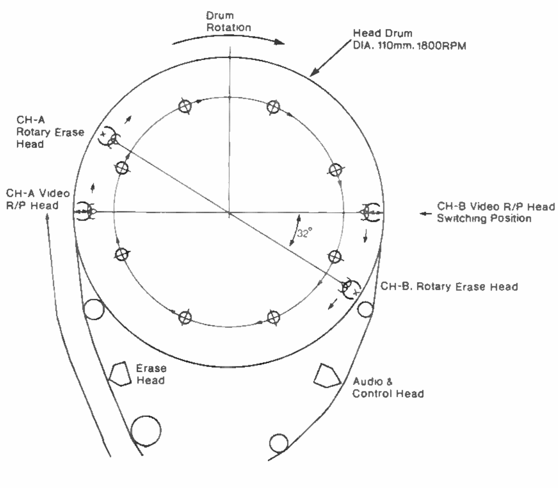


Fig. 3. U-format.

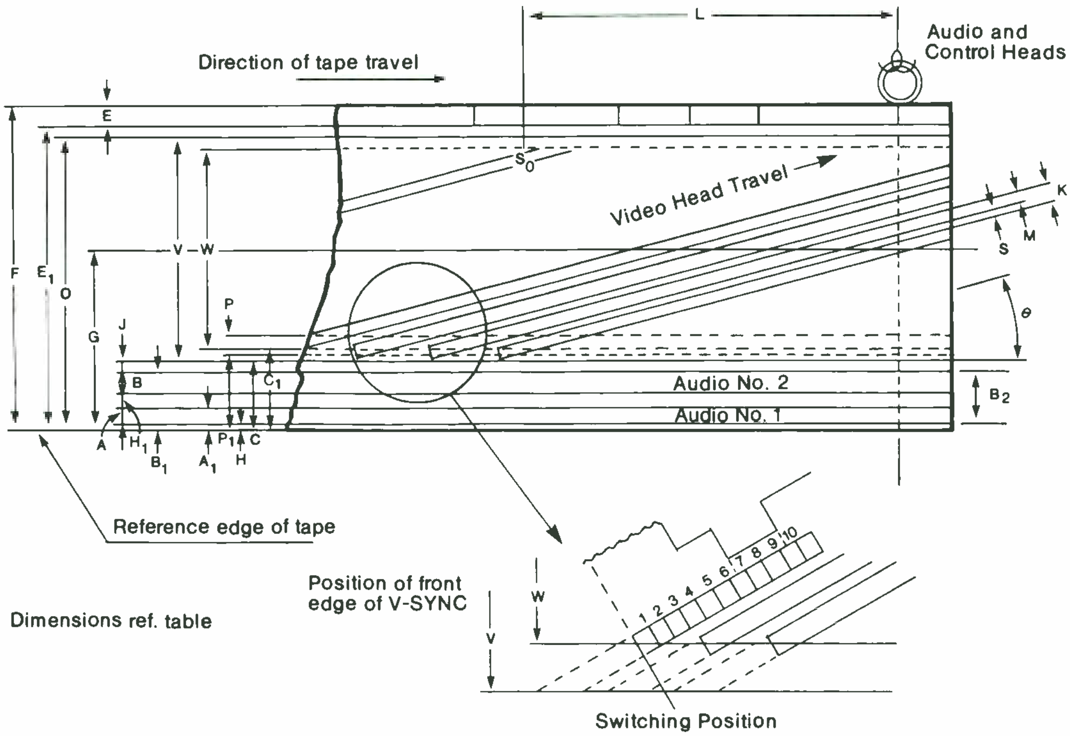


Fig. 4. Track configuration and dimensions from magneto-sensitive side.

	Dimensions	Millimeters		Inches	
A	Audio No. 1 width	0.80	± 0.05	0.0315	±0.0020
A_1	Audio No. 1 reference	1.00	nom	0.0394	nom
B	Audio No. 2 width	0.80	± 0.05	0.0315	±0.0020
B_1	Audio No. 2 reference	2.50	nom	0.0984	nom
B_2	Audio track total width	2.30	± 0.08	0.0906	±0.0031
C	Video area lower limit	2.70	min	0.1063	min
C_1	Video effective area lower limit	3.05	min	0.1201	min
D	Video area upper limit	18.20	max	0.7165	max
E	Control track width	0.60	nom	0.0236	nom
E_1	Control track reference	18.40	+ 0.28 — 0.18	0.7244	+ 0.0110 — 0.0071
F	Tape width	19.00	± 0.03	0.7480	±0.0012
G	Video track center from reference edge	10.45	± 0.05	0.4114	±0.0020
H	Audio guard band to tape edge	0.2	± 0.1	0.008	±0.004
H_1	Audio-to-audio guard band	0.7	nom	0.028	nom
J	Audio-to-video guard band	0.2	nom	0.008	nom
K	Video track pitch (calculated)	0.137	nom	0.00539	nom
L	Audio and control head position from end of 180° scan	74.0	± 0.5	2.913	±0.020
M	Video track width	0.085	± 0.007	0.00335	±0.00028
P*	Address track width	0.50	± 0.05	0.0197	±0.0020
P_1	Address track lower limit	2.90	± 0.15	0.1142	±0.0059
S	Video guard band width	0.052	nom	0.00205	nom
V	Video width	15.5	nom	0.610	nom
W	Video effective width	14.80	± 0.01	0.5827	±0.0004
θ	Video track angle, moving tape	4° 57' 33.2"			
	stationary tape	4° 54' 49.1"			

TABLE 1. Recorded Magnetic tape records.

The U-format system provides an overlap period during which the same information is recorded by the head preparing to leave the tape and the head which is initiating its pass. This overlap also exists in playback. Electronic switching is used to control the changeover to produce a continuous TV signal.

The switching position takes place in the vertical interval before the leading edge of vertical sync. The RF output extends past the switching point by approximately three horizontal lines to provide three horizontal lines of overlap.

Typical Performance Specifications

Video

Input Signal: NTSC Composite, 1.0 V p-p
Dubbing Input/Output:
  Luminance: 1.7 V p-p
  Chrominance: 0.9 V p-p
Signal-to-Noise Ratio:
  47 dB (color mode)
  48 dB (monochrome mode)
Horizontal Resolution:
  260 lines (color mode)
    (approx. 3.2 MHz BW)
  340 lines (monochrome mode)
    (approx. 4.2 MHZ BW)

Audio

Frequency Response: 50 Hz to 15 kHz
Signal-to-Noise Ratio: 48 dB (ref. 3% distortion level)
Distortion: 2.0% (1 kHz at reference level)

System Features

Use of the U-format system since its introduction has caused evolutionary changes in system capability. Various features and functions are available although many may be options depending upon system application and manufacturer and model. These features include:

Shuttle and Jog Search Functions
The search functions enable quick location of edit points. In the shuttle mode a recognizable picture is obtained at speeds from 1/30 to 10 times normal speed in both forward and reverse direction. In the jog mode the picture moves in one field increments according to the motion search control dial.

Dynamic Tracking
Playback head position is servo-controlled to maintain optimum tracking in still frame and other nonstandard-speed modes.

Editing
Equipment which provides edit mode capabilities, in general, permits preroll, assemble, insert mixed audio channel editing, preview, entry in/out, and trim field-by-field capabilities.

Dubbing
Dub mode provides the capability of directly recording the separated luminance and chrominance channels without the need to up-convert and reassemble and then decode and down-convert.

Time-Code Generator: internal automatic time-code generator and reading per SMPTE Time Code

Time-Base Correctors: normal color-under processing corrects time-base error sufficiently for viewing video playback directly on a color monitor or receiver. However, to meet broadcast specifications, additional time-base correction is required. Time-base correction is also required to provide proper signal timing in the stop motion or enhanced motion modes.

Drop-out Compensation (DOC): A playback carrier drop-out detector and compensator replaces the missing RF information with a delayed signal.

Summary

The 3/4″ U-format system was conceived originally as a non-broadcast system due to its low cost and limited bandwidth. The heterodyned signal was sufficiently stable to lock a monitor but was not broadcastable.

With the advent of low cost TBCs and portable recorders, 3/4″ became used in broadcast news operations because of its obvious advantages over film with respect to turn around time, program length and general adaptability to news gathering. It thus became the defacto ENG standard.

As technology has improved, 3/4″ continues to be improved to its limits. But there are definite constraints of the format itself. The color-under technique allowed slower tape and drum speeds by mixing the color information to a lower frequency, and bandlimiting the luminance information. The restricted luminance bandwidth limits detail and can cause ringing on abrupt transitions. Because the chroma is recorded as an AM signal, it is susceptible to off-tape amplitude variations (unlike FM). Because the color signal is separately filtered and processed, chrominance-to-luminance delay can be difficult to maintain, and the resulting bandlimiting can cause ringing on large color transitions. However, recent developments have overcome some of these limitations.

The 3/4″ U-format continues to fulfill a need. Full-bandwidth recording provided by Quad or

Type C was not cost efficient. Low cost TBCs allow heterodyned U-format video to be broadcast and 3/4" has found a comfortable niche in ENG and EFP recording despite its limitations.

MAINTENANCE

Introduction

The key to maintaining any piece of equipment is a systematic approach for both attacking an unknown problem and preventing recurring problems. A systematic preventative approach maintains performance and reduces out-of-service time.

One can break down the maintenance of U-format machines into two areas. One area is the routine procedures for prevention and upkeep. Troubleshooting and problem analysis can be divided into functional subgroups to aid in problem determination.

There are a few items discussed herein that pertain to the Sony BVU-800, but for the most part, all items are generally applicable to all 3/4 inch equipment.

Routine Activity

The following should be performed on a regular basis, starting with that which should be done most frequently.

1. Operational Check

An operational check should be performed before each extended operational session. This is merely a quick verification of all functions of the machine, from tape threading to editing.

All button lamps should be checked because they burn out regularly. The lamps which illuminate the cassette area should also be checked.

2. Cleaning

Cleaning of video heads

Cleaning of the video heads should be performed with a non-residue producing evaporative solvent such as freon, and a non-lint producing cloth wipe. (There are commercial products available for this application.) When the drum is cleaned, one must take care in gently cleaning heads in a direction parallel to head travel only, otherwise the heads can be damaged. The best technique is to turn the drum slowly while holding the cloth against it.

Cleaning of the tape path

All surfaces that are in contact with the tape should be cleaned, including the fixed heads (time code, audio/control track, erase) and all tape guides.

Cleaning and inspection of pinch roller

It is very important for the pinch roller to stay clean. It easily collects oxide residue, and loses its ability to grab the tape. The roller should be inspected for aging or deformation, and changed if necessary. The roller can dry out and not grab effectively.

Head Degaussing

Both the rotary heads and the fixed heads should be degaussed. The degausser is placed as close to the head as possible, turned on, and drawn away before it is turned off. This removes residual magnetization from the heads. (This procedure does not have to be done as often as cleaning but is easily done at the same time.)

Cleaning of head drum slip rings

This need only be done when necessary but is included for completeness. The slip rings should be cleaned with a soft brush to remove built-up dirt. If necessary, freon can be used.

3. Tension Detector Adjustment

The BVU-800 has two tension detectors which control tape tension by regulating power supplies to the take-up and supply reels (Fig. 5). The detector is an electromechanical device which consists of an LED which shines through a slit onto a special photoresistive cell. The slit is on a plate which is in contact with the tape and moves according to how much tension is present. When the slit moves, a varying control voltage results. This control voltage is used to vary the reel motor torque.

The tension detectors are replaceable and should be checked and adjusted as needed, on a regular basis (manufacturer recommends 500 hrs.). Both physical alignment and calibration should be checked.

4. Head Replacement and Optimization

The rotary video heads are not replaced individually, the whole upper drum assembly is replaced when any one head fails. The upper drum is easily changed, but it must be carefully aligned. The manufacturer supplies an eccentricity gauge which is used to align the drum. The gauge assembly is attached to the chassis adjacent to the drum so that the tip of the gauge is in contact with the rim of the drum. If the drum is not centered on the flange the gauge indicates its deviation. The position of the drum on the flange is adjusted until the gauge indicates a deviation of less than 5 microns during a revolution.

After replacement of the head assembly, adjustments should be to optimize performance for the characteristics of the new head. The following adjustments, should be made (check with manufacturer's manual).

- Equalization
- Playback RF Levels and Balance
- Record Currents (Y, Chroma, Erase)

It is also desirable to perform these adjustments several times throughout the life of a set of heads, because their characteristics change as they wear. Manufacturer's recommendations are for replacement after 1000 hours and optimization at 200, 500 and 750 hours.

Functional Divisions for Troubleshooting

1. Threading

The cassette loading process involves sensors, solenoids and motors, which detect and control the loading process. If a sensor becomes misaligned, the process will halt. If a solenoid or motor cycle is out of specification a mechanical jam could occur. The manufacturer's manual should be consulted for the mechanical adjustments.

A good understanding of the control logic associated with the loading cycle will ease in troubleshooting. Often the manufacturer outlines a logic timing diagram which displays when each event is supposed to occur. Also, the control logic, is often designed to put the machine into a safe state (one that will not result in tape or machine damage) when it detects that certain cycles have not occurred. Understanding the possible conditions that result in the machine entering this state, will help.

2. Control System

The control system interprets all input from the buttons and sensors, and for actuating events in the sequence required. Most problems with the control system "hanging up" are due to a cycle that was not completed. This is unlikely to be a control system problem but a problem with a signal that the control system is supposed to see. Therefore it is important for all the sensors to operate properly.

3. Tension and Tape Run

If the cassette successfully loads, but jams still occur, or if portions of the picture appear consistently noisier—implying a nonlinear RF envelope, the tension and tape run should be checked.

Tension

As previously mentioned, the tension detectors feedback a control voltage which in turn regulates the take-up and supply reel tension (Fig. 5). Other adjustments that affect tension are calibration of the brake and motor sensitivity.

Tape Run

After the tension adjustments have been made, the tape guides should be checked. The manufacturer's manual should be consulted for this. But the objective is to be able to play an alignment tape, and observe a uniform RF envelope, implying that the video heads are successfully staying within the recorded tracks on the alignment tape. If the entrance or exit guides are not properly aligned, the beginning or the end of the RF envelope will have a lower amount of detected RF. The many other tape guides are responsible for aligning the tape properly as it passes by the stationary heads, and maintaining a uniform edge to edge tension as the tape turns and changes height. These all must be checked and adjusted per manufacturers specifications.

Notes on use of Alignment Tape

For both audio and video systems alignment, procedures regarding the use of alignment tape should be understood. The alignment tape should be handled very carefully, and stored under good conditions. One should make sure the machine is running with correct tension, and that the tape path is clean and degaussed, before the alignment tape is played on the machine. It is best to run the tape all the way through without stopping because starting and stopping may stretch the tape and destroy its accuracy.

4. Servo System

Elements of a Servo System

There are 4 elements to a basic servo system:

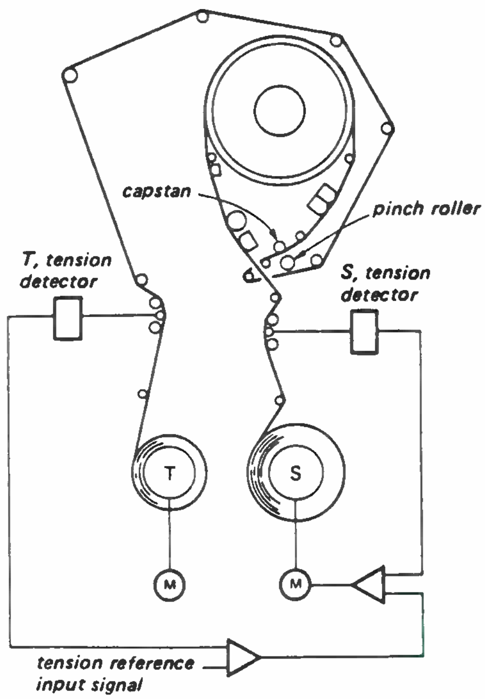


Fig. 5. Tension controlled by regulating power supply.

1. Input Reference
2. Motor or some other device to provide movement
3. Feedback detector of output position
4. Comparator to determine difference between desired and achieved output

Adjustments to the many servo systems in a machine (such as that in Fig. 5) include calibration of references and calibration of open loop responses. The manufacturer's manual should be consulted for these adjustments.

5. Audio System

Alignment of the audio system is similar to an audio recorder, except that there is a combined record/play head. Physical alignment should be checked, then an alignment tape played back to verify playback bias, equalization and levels should be adjusted to result in comparable playback through the set up playback electronics. The erase current also should be checked.

6. Video System

Alignment of the video system should be performed in the same manner as that described for audio, in that playback electronics are aligned first according to alignment tape, followed by record adjustments to optimize playback through the previously adjusted playback electronics.

Functional Divisions

1. Playback amplifier optimization (as previously described)
2. Demodulator adjustments (both luminance and chroma)
   ACC adjustment
   DOC adjustment
   Playback color oscillator
3. Modulator adjustments
   Color separation filter
   Local oscillator for chroma modulator
   Record equalization

The manufacture should provide procedures to perform the above adjustments.

7. Power Supply

The power supplies can be checked and adjusted if they are suspected of causing a problem. Things to look for:

1. Purity of dc
2. Regulation under varying loads
3. Regulation over the specified range of input voltages.

Summary

This section, in conjunction with the theory section, is intended to provide an understanding and an organized approach towards maintenance of 3/4 inch equipment. The two most important items being an effective preventative maintenance schedule and a thorough functional understanding of the machine to enable quick isolation.

1/2" FORMAT VIDEO TAPE SYSTEMS

Principles And Current State Of The Art

Introduction

The introduction of the 1/2" VHS and Beta cassette tape formats was directed at the consumer market and featured long playing times at the sacrifice of professional performance levels. Subsequent improvements in tape technology offered an opportunity to provide 1/2" tape formats meeting various levels of performance for specific professional applications.

Three different 1/2" formats were introduced in the 1982-1985 time period. The three systems, in the order of introduction, are the M-Format, Betacam format and MII-Format.

All three are based on a helical-scan video tape recording system using 1/2" tape cassettes. All three employ component recording in that they record the luminance (Y) and baseband chrominance (C) signals for one television field on separate tracks with each pass of the recording head. Assigning separate tracks for the luminance and chrominance-video permits the system to have a wider baseband recording capability, but with a sacrifice in tape usage.

Among other factors, differences in the record locations and dimensions, tape speed, and the method of encoding the chrominance channel produce various levels of performance among the three systems.

These differences in implementation and the resulting differences in performance are described in following sections.

Component recording offers the following advantages over composite recording:

- editing without being constrained by color framing considerations,
- no color misregistration from envelope delay problems inherent in composite systems,
- lack of sub-carrier caused moire.

The disadvantages are:

- the need to maintain signals in component format throughout the editing process, or subject video to degradation due to continual encoding and decoding

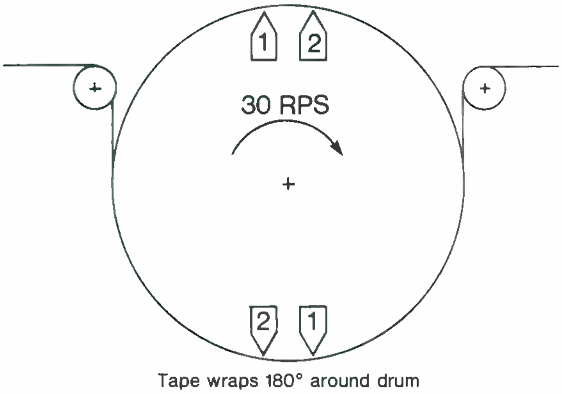


Fig. 6. Half-inch tape format.

- chrominance/luminance delay variations
- less efficient use of tape

Basic Definitions

The basic terms, such as *drum* or *wrap angle*, are essentially the same for 3/4″ and 1/2″ recorders. Please see page 5.9-159 of this chapter for a brief glossary.

Principles And Characteristics

A "helical-scan" architecture is used in all three 1/2″ format machines, in that tape unspools from the supply reel and is presented to the drum in such a manner that the heads slant across the tape to form part of a helix.

Two sets of video heads are placed 180 degrees apart on the scanner and the tape is wrapped around the scanner for slightly more than 180 degrees (Fig. 6). Each set of video heads consists of two heads, one for the luminance (Y) channel and one for the chrominance (C) baseband channel with the two channels (Y and C) recorded simultaneously. An additional set of heads is employed for "flying erase." The scanner rotates at exactly 29.97 revolutions per second (approx. 1800 rpm) recording one TV frame per revolution and one TV field per 180 degree scan.

All three systems provide, in addition to the two video channels, four longitudinal tracks, two for audio, one for control track and one for time code (Fig. 7). The video tracks do not interfere with the time code track and can be independently edited.

The differences in performance are attributable to, among other factors, the variations in implementation such as drum diameter, frequency spectrum, chrominance coding, writing speed and tape speed, the details of which are provided in Tables 2 and 3 and Fig. 8.

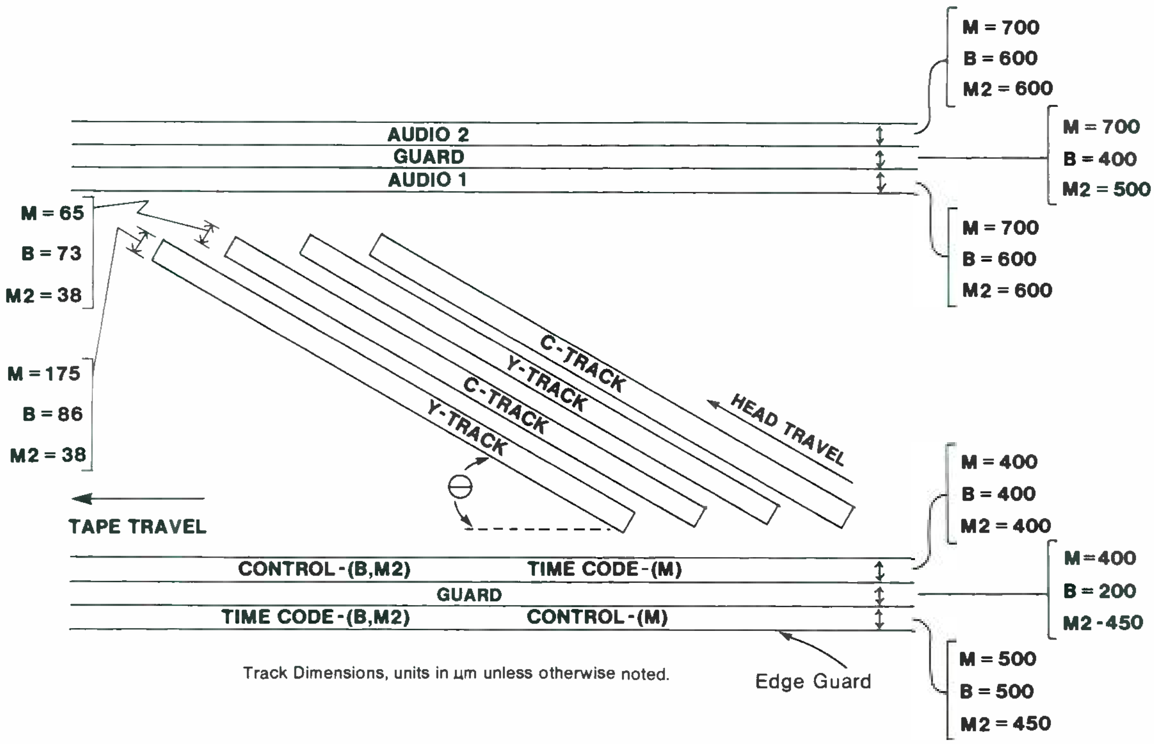


Fig. 7. Tape format comparison.

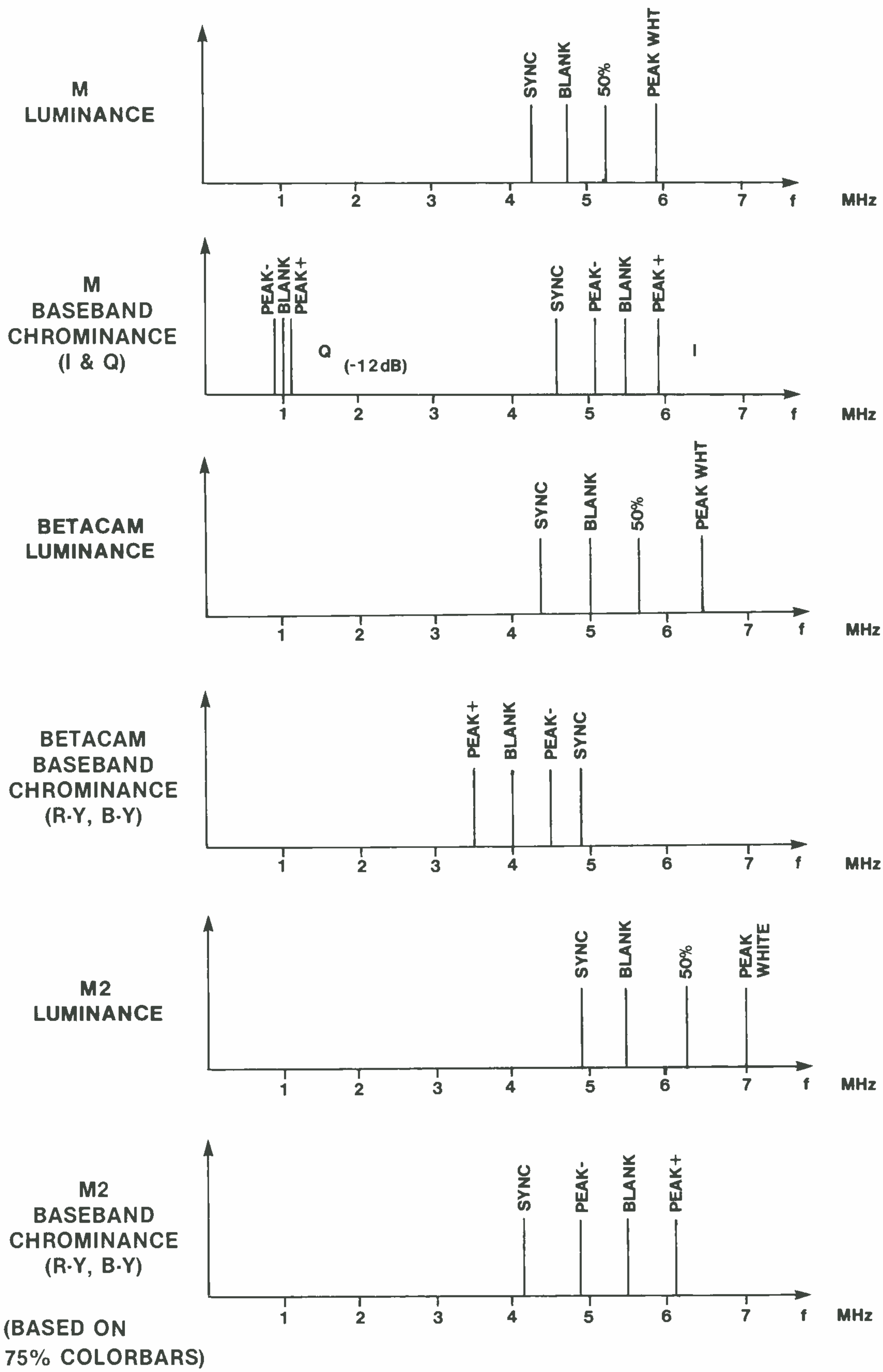


Fig. 8. Frequency spectrum of formats.

TABLE 2. Format implementation comparison.

	M	BETACAM	MII
DRUM DIAMETER	62.0 mm	74.487 mm	76.0 mm
R.P.M.	1800	1800	1800
WRITING SPEED	5.63 m/s	6.89 m/s	7.09 m/s
LINEAR TAPE SPEED	204.5 mm/s	118.52 mm/s	67.65 mm/s
CASSETTE LENGTH (standard)	246 m	150 m	391 m
PLAY LENGTH (standard)	20.0 min.	21.1 min.	96.3 min.

In all three systems, the luminance (Y) and baseband chrominance (C) channels are pre-emphasized by two methods before modulation.

Luminance And Chrominance

In all three systems the luminance signal is converted to an FM signal and is recorded on the Y track. The three systems differ in the method of encoding chrominance and the FM deviation used as shown in Table 4.

The M system carries the color information encoded in the I and Q format defined for the NTSC system. These are converted to a 5.0 to 6.0 MHz deviation FM signal for I and a 0.75 to 1.25 MHz deviation FM signal for Q. The FM signals are then summed (frequency multiplexed) and recorded on the color track as shown in Figs. 3 and 4. The Q signal is linearly recorded with the I level increased to provide the high-frequency bias needed for linear recording.

The Betacam and MII systems carry the color information encoded as R-Y and B-Y. These are time compressed 2:1 so that each of the R-Y and B-Y components are contained in a period equivalent to ½ active horizontal picture interval (Fig. 10). An additional delay of approximately ½ H is applied to the B-Y component to place the compressed B-Y information in the last half of the line. The information is then FM modulated as described in Fig. 8.

TABLE 4.

M	
Luminance	(deviation 1.6 MHz)
Chrominance	I and Q frequency division multiplexed (Q deviation 0.36 MHz, I deviation 1.25 MHz)
BETACAM	
Luminance	(Deviation 2.0 MHz)
Chrominance	Time division multiplexing of compressed R-Y signals (deviation 1.4 MHz)
MII	
Luminance	(deviation 2.1 MHz)
Chrominance	Time division multiplexing of compressed R-Y and B-Y signals (deviation 2.0 MHz)

During playback the decompressed R-Y and B-Y information is demultiplexed, and any system processing delay differences between the three channels (Y, R-Y, B-Y) are corrected.

System Features

All three systems offer the following features, some of which may be optional on specific models:

Control track: The control track provides a constant flux level, alternating in polarity, and completing one cycle per frame. The polarity of the track-control record is such that the south pole of the magnetic domains points in the direction of tape travel when channel 1 (field 1) is recording.

TABLE 3. Manufacturers' specifications.

Technical Performance

Parameter	M – (1)	Beta(2)	MII(3)	
Luminance Bandwidth (–3 dB)	4.1	4.1(–6 dB)	4.5	MHz
Luminance SNR	50	48	49	dB
K factor (2T)	2	2	<2	%
Chroma Coding	I,Q	R-Y,B-Y	R-Y,B-Y	
Chroma Bandwidth	1.3	1.5	1.5	MHz
Chroma SNR	52	50	50	dB
Diff Gain	<3%	<3%	<2%	%
Diff Phase	<3°	<3°	<2°	degrees
Y-C Delay	<30	<20	<20	ns
Audio Bandwidth (±3 dB)	40-15k	50-15k	50-15k	Hz
Response	±2	±3	±1.5, –3	dB
Audio SNR (3% distortion)	53	50	56	dB
Audio Distortion	1.0	2.0	1.0	%
Wow & Flutter	.15	.15	.1	%
Track Width	Y175, C65	2 × 73 Y86, C73	2 ×38	um

(1) Per AU 300 Specification
(2) Per BVW 40 Specification
(3) Per AU600 Specification

One Piece Recorder/Camera: Each of the systems offers a one-piece recorder/camera featuring a recorder time of 20 to 30 minutes depending on the system. Power consumption is less than 12 watts and weight of less than 4 kg (sans lens) depending upon the model.

Studio Recorder: Each of the systems offers a studio player or recorder/player with built-in TBC, jog/shuttle functions, built-in time-code generator, dynamic tracking and/or editing functions depending upon the model.

Field (Portable) Recorder/Player: Each of the systems offers a field unit with various options with power consumptions less than 24 W and weight less than 7 kg depending upon the model.

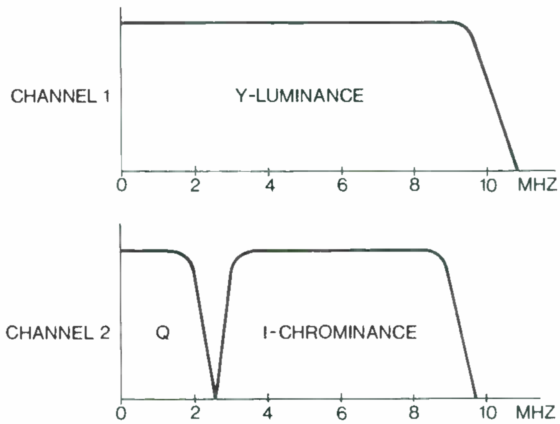


Fig. 9. M system two-channel FM multiplexed spectrum allocation.

Time-Code Generator: SMPTE/EBU Time Code is automatically recorded on the dedicated time-code track.

Audio Noise Reduction: Dolby C ™ is included on most models.

Additional audio tracks: some systems offer additional audio channels recorded in portions of the video tracks using FM or PCM techniques, offering better quality than the analog tracks.

High Speed shuttle: between 5x and 32x depending upon the manufacturer and model.

Camera/Recorder Playback: viewfinder playback with various performance indications is provided in most models.

Back-space edit: on camera/recorders, provides sequential recording without picture break-up at transition points.

Editing features: include frame-by-frame forward or reverse trim, selectable pre-roll time, auto edit in/out functions, dub interconnections, audio mixing, built-in TBC's, and most features found on professional broadcast VTRs.

Summary

Pursuit of full bandwidth recording, such as Quad or Type C, resulted in a push for higher FM carrier frequencies (Low-band to High-band

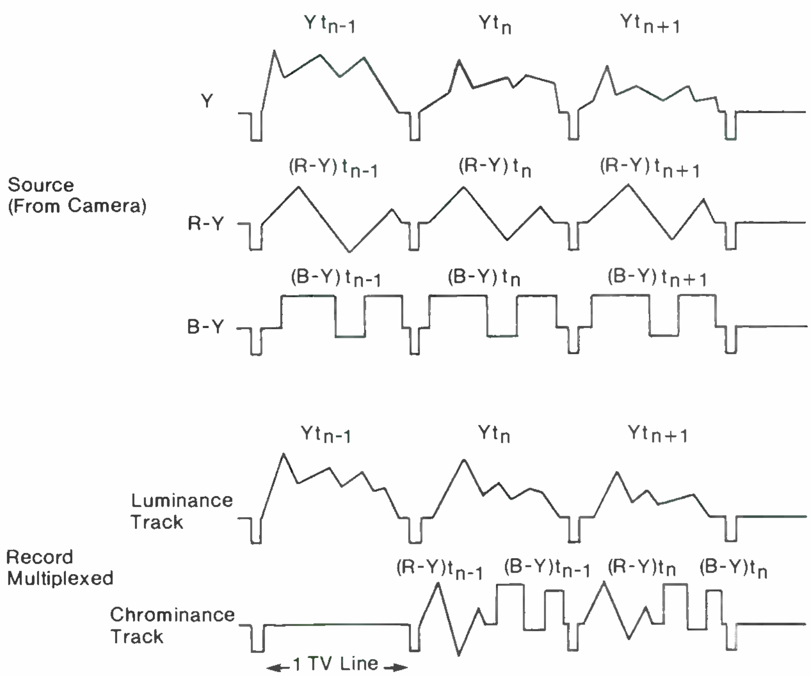


Fig. 10. Betacam, MII two-channel time domain multiplexed allocation.

etc.) This resulted in shorter recording wavelengths. The problems associated with using shorter wavelengths are a poorer carrier-to-noise ratio due to smaller, longitudinal magnetic components and an increased chance of drop-outs.

By separating luminance and chrominance information prior to recording, and recording on two tracks in one scan, one can utilize longer recording wavelengths. Also, if source material is from a camera, the information can remain in component form to the recorder, preserving chroma bandwidth.

These 1/2″ formats apply these techniques to enable a smaller tape and a smaller scanner with only a 180 degree wrap (as opposed to Type C <360 degree). Near full bandwidth can be achieved as opposed to color-under systems which sacrifice bandwidth.

MAINTENANCE

Introduction

Although at first glance, 1/2″ component format machines may appear similar to 3/4″ U-format counterparts, the similarity ends there. The video signal system is significantly different and requires additional test equipment. Also, the circuit board construction techniques applied in camera recorders require a significant change in repair procedures. Both new equipment and techniques are required for repair work to be done on the boards. In this section we will discuss maintenance of 1/2″ component equipment.

Routine Activity

The following should be performed on a regular basis.

1. Operational Check

An operational check should be performed before each extended operational session. This is merely a quick verification of all functions of the machine, from tape threading to editing.

2. Cleaning

Cleaning of video heads

The following applies to essentially all helical video tape recorders.

Cleaning of the video heads should be performed with a non-residue producing evaporative solvent such as freon, and a non-lint producing cloth wipe. (There are commercial products available for this application.) When the drum is cleaned, one must take care in gently cleaning heads in a direction parallel to head travel only, otherwise the heads can be damaged. The best technique is to turn the drum slowly while holding the cloth against it. For 1/2″ equipment there are 2 sets of opposed heads, for 1/2″ machines with Slo-mo, there are 3 sets of opposed heads.

Cleaning of the tape path

All surfaces that are in contact with the tape should be cleaned, including the fixed heads (time code, audio/control track, erase) and all tape guides.

Cleaning and inspection of pinch roller

It is very important for the pinch roller to stay clean. It easily collects oxide residue, and loses its ability to grab the tape. The roller should be inspected for aging or deformation, and changed if necessary. The roller can dry out and not grab effectively.

Head Degaussing

Both the rotary heads and the fixed heads should be degaussed. The degausser is placed as close to the head as possible, turned on, and drawn away before it is turned off. This removes residual magnetization from the heads. (This procedure does not have to be done as often as cleaning but is easily done at the same time).

Cleaning of head drum slip rings

This need only be done when necessary but is included for completeness. The slip rings should be cleaned with a soft brush to remove built up dirt. If necessary, freon can be used.

3. Tension Check

It is advisable to check tape tension in all modes of operation. Improper tape tension can cause tape or head damage, yet tension can be checked quickly.

4. Head Replacement and Optimization

As with U-format equipment, the upper drum assembly is replaced when any one video head fails. The upper drum is then aligned using an eccentricity gauge. The gauge, which must be attached to the chassis, is used to verify that the drum is centered on the flange. The drum is rotated slowly and the gauge is observed. Adjustments are then made, and the process is repeated until the drum is centered.

After replacement of the head assembly, adjustments should be made to optimize performance for the characteristics of the new head. And because these systems are component systems, there are two sets of record and playback electronics to be optimized. The following adjustments should be made for both chroma and luminance channels:

- Equalization (Playback and Record)
- Playback RF Levels
- Record Currents (also for erase heads)

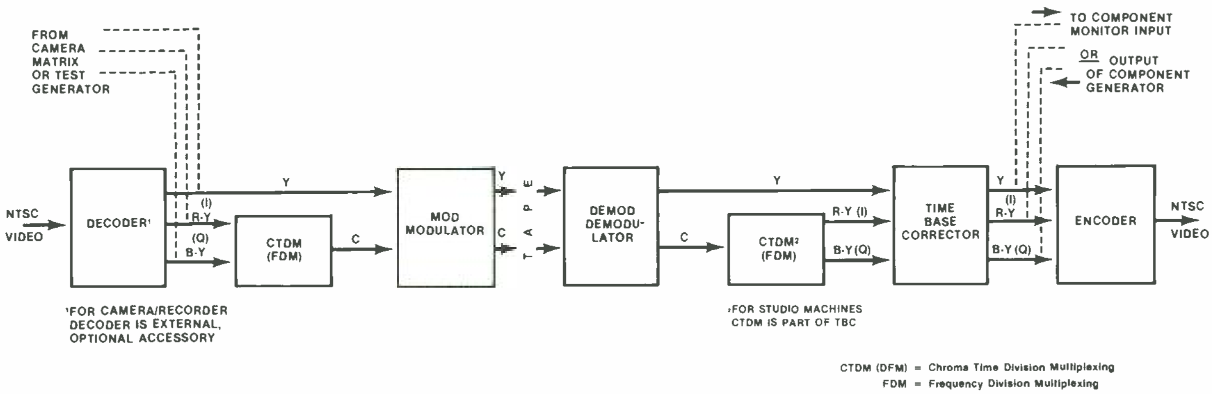


Fig. 11. Component video recorder player: functional divisions.

In addition to performing these adjustments after head replacement, it is also desirable to perform these adjustments several times throughout the life of a set of heads. This is because their characteristics change as they wear. Manufacturers recommended interval for replacement should be checked. If you maintain consistent selection of tape and operating conditions, you may be able to determine your own head life projection, and base your replacement schedule on that.

Tools and Test Equipment

1. Component Test Equipment

At this point it is appropriate to discuss the important differences in a component machine. The video signal system has two separate paths (Fig. 11).

It becomes important to have component test equipment available to facilitate isolation of problems. With component test signals available, it is easier to set up the encoder accurately. With component monitors the record/playback system performance can be scrutinized without considering encoder effects. Using both, one can concentrate on the record/play system by itself.

2. Soldering Stations

The introduction of camera recorders, brought with it a new circuit board construction that requires new techniques for component replacement. Surface mount components require special tools, such as different soldering tools. The stations are different in that they have suction and regulated heat as well as special tips that are designed for use with specific components. Use of these tools requires training. Component replacement is dramatically different from regular printed circuit board work, but given the time, one can become proficient.

Functional Divisions for Troubleshooting

1. Threading

The cassette loading process involves sensors, solenoids and motors, which detect and control the loading process. If a sensor becomes misaligned, the process will halt. If a solenoid or motor's cycle is out of specification a mechanical jam could occur. The manufacturer's manual should be consulted for the mechanical adjustments.

2. Control System

The control system interprets all input from the buttons and sensors, and for actuating events in the sequence required. Most problems with the control system "hanging" are due to a cycle that was not completed. This is unlikely to be a control system problem but a problem with a signal that the control system is supposed to see. Therefore it is important for all the sensors to operate properly.

3. Tension and Tape Run

If the cassette successfully loads, but jams still occur, or if portions of the picture appear consistently noisier—implying a non-linear RF envelope, the tension and tape run should be checked.

Tension

As previously mentioned, the tension detectors feedback a control voltage which in turn regulates the take-up and supply reel tension (Fig 11). Other adjustments that affect tension, are calibration of the brake and motor sensitivity.

Tape run

After the tension adjustments have been made, the tape guides should be checked. The manufacturer's manual should be consulted for this. But the objective is to be able to play an alignment tape, and observe a uniform RF envelope, implying that the video heads are successfully staying within the recorded tracks on the alignment tape. If the entrance or exit guides are not properly aligned, the beginning or the end of the RF envelope will have a lower mount of detected RF. The many other tape guides are responsible for aligning the tape properly as it passes by the stationary heads, and maintaining a uniform edge to edge tension as the tape turns and changes height. These all must be checked and adjusted per manufacturers specifications.

Notes on use of Alignment Tape

For both audio and video systems alignment, procedures regarding the use of alignment tape should be understood. The Alignment Tape should be handled very carefully, and stored under good conditions. One should make sure the machine is running with correct tension, and that the tape path is clean and degaussed, before the alignment tape is played on the machine. It is best to run the tape all the way through without stopping because starting and stopping may stretch the tape and destroy its accuracy.

4. Servo System

Elements of a Servo System

There are 4 elements to a basic servo system:

1. Input reference
2. Motor or some other device to provide movement
3. Feedback detector of output position
4. Comparator to determine difference between desired and achieved output

Adjustments to the many servo systems in a machine include calibration of references and calibration of open loop responses. The manufacturer's manual should be consulted for these adjustments.

5. Audio System

Alignment of the audio system is similar to an audio recorder, except that there is a combined record/play head. Physical alignment should be checked, then an alignment tape played back to verify playback bias, equalization and levels should be adjusted to result in comparable playback through the set up playback electronics. The erase current also should be checked.

6. Video Systems

Alignment of the video system is significantly different from either type C or U-format. Looking at the block diagram (Fig. 11) we see that we now have a decoder and an encoder to align (decoder for studio NTSC machines). Also there are two modulators and demodulators.

Alignment should be performed in the same manner as that described for audio, in that playback electronics are aligned first according to alignment tape, followed by record adjustments to optimize playback through the previously adjusted playback electronics.

Functional Divisions

I. Playback Electronics (done first)
   - A. Demodulator Adjustments
     - Playback Equalization (de-emphasis)
       - —Non-Linear
       - —Non-linear, incremental
   - B. Encoder Adjustments

II. Record Electronics
   - A. Decoder Alignment (for NTSC studio machines
   - B. Modulator Adjustments (for BOTH modulators)
     - Record Equalization (pre-emphasis)
       - —Non-Linear
       - —Non-linear, incremental

The manufacture should provide procedures to perform the above adjustments.

7. Power Supply

The power supplies can be checked and adjusted if they are suspected of causing a problem.

Things to look for:

1. Purity of dc
2. Regulation under varying loads
3. Regulation over the specified range of input voltages.

Summary

This section, in conjunction with the theory section, is intended to provide an understanding and an organized approach toward maintenance of 1/2" equipment. The two most important items being an effective preventative maintenance schedule and a thorough functional understanding of the machine to enable quick trouble isolation.

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Video Tape Editing

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INTRODUCTION

The advent of Electronic News Gathering (ENG) brought high volume video tape editing into the average broadcast facility for the first time. Previous to ENG, news editing was usually done with film, and production video tape editing performed on a small scale using a pair of the "air" VTRs located in the video tape machine room.

The news edit suite of today has become almost a complete "out of the box" system, requiring minimal custom engineering. Once a satisfactory design has been achieved, additional suites are built to provide sufficient editing capacity. In total, a complete news editing assembly line can exist with excellent speed and efficiency in performing this specific task.

It should be with little surprise that the engineering manager, who so efficiently replaced the cantankerous film processor with the technology of ENG, should hear the request of his general manager to eliminate the cost and expense of utilizing out-of-house post production facilities. As the general manager sees it, nothing could be simpler than combining a more sophisticated editing controller with other pieces of studio equipment that sit idle between productions. The facility probably already owns a digital effects system, character generator and numerous VTRs that, according to the advertising brochures he has seen, can be "easily" integrated into an in-house post production system.

The purpose of this chapter is to provide a combination catalog and primer designed to fill in the gaps between the information found in glossy brochures and the experience gained from ENG editing.

HISTORY

The introduction of quadruplex video tape recorders provided the broadcaster with an early means of video mass storage. The new found ability to record and immediately play back the recording appeared to be such an obvious advantage over film that, soon after introduction, the future of film was predicted to be that of obsolescence. Today, more than a quarter of a century after the introduction of video tape recording, the predictions of the total demise of film have proven to be a bit premature.

One major reason that film has not been entirely pushed out of the editing room is that a frame of film can be directly observed. Using this ability, it is possible to mark a desired frame and mechanically or optically splice it to a different frame, thus producing an edit.

Video tape recording technology cannot easily duplicate the simplicity of direct observation editing. Video tape recording makes use of magnetic media to record electronic signals representing the visual information. The storage of visual information by a method that can not be directly observed eliminates the ability to edit using direct observation. Ever since encountering this problem, advances in video tape editing technology have been aimed at giving the video tape user the same or similar ability found in film editing, and the "real time" advantages of tape.

The initial approach to editing video tape attempted to directly emulate film techniques. The tape was played back, and the edit points marked on the tape with a grease pencil. In order to physically observe the magnetic track in the vicinity of an edit point, tapes were developed with fine particles of ferrous materials in a liquid suspension. These liquids provided the ability to visually locate, with the aid of a microscope, the frame pulse and associated video track near the edit marks. The actual edit was then performed by cutting and splicing the source and record tapes. A complete cut of the tape effectively divides both video and audio channels thus preventing individual channel editing.

The finality of each edit, the questionable durability of tape splices, and many other physical limitations of the mechanical techniques, produced a demand for an improved method of editing. About the only advantage of this type of editing was that it required only the original tape generation at a time in recording technology when generation loss was severe. The mechanical editing technique was replaced by the first generation of the electronic editors. VTRs equipped with electronic editors contained the systems necessary for correct record timing and servo switching to properly transition from the playback to the record mode while the tape continuously rolled. This switch occurred between the tracks before the vertical interval, and upon playback appeared as if the transition had been accomplished by the studio switcher.

To perform an edit, the operator would mark the tape at the pre-roll point, backspace each VTR by hand to allow enough time for lock-up and manually start them, as simultaneously as possible. If both VTRs locked-up before they reached the edit point, the operator would manually initiate the edit. If the pre-roll time had slipped between the source VTR and the record VTR, the edit point would be missed. If the editing operator's reactions were off slightly causing the desired edit point to be missed, there would not be a second chance. These problems produced the need to devise a method that would reduce the dependence on operator reaction time, and allow the testing of an edit without actually recording.

The Ampex Editec, see Fig. 1, approached these problems by allowing the operator to record a cuetone to make the edit point. This tone could be used to trigger the edit record timing circuitry of the electronic editor.

An edit could be previewed by using the cuetone to switch the record VTR input to the play VTR, without turning on the record amplifier and erase systems. The play VTR signals were then routed through the Electronics to Electronics (E to E) path of the record VTR instead of the record VTR Video. This cuetone technique became the common place method, but it still relied on manual per-rolling of the transports.

A complete description of the Ampex Editec can be found in the video tape recording chapter of the previous, 6th edition of this *Handbook*.

One of the difficulties encountered in quadruplex recording is the inability to examine edit points on a frame by frame basis. This problem was addressed by utilizing magnetic disc technology, as exemplified in the Ampex HS-200, see Fig. 2. Early magnetic disc technology offered the ability to examine video on a frame by frame basis. The HS-200 integrated the programmable control of the HS-100 video disc recorder and cuetone control of quad VTRs. This allowed the slow motion and still frame capability of the disc system to be initiated by edit marks. The ability to program switcher transitions and effects between the HS-100 and VTRs under control of the edit marks was also contained in the HS-200.

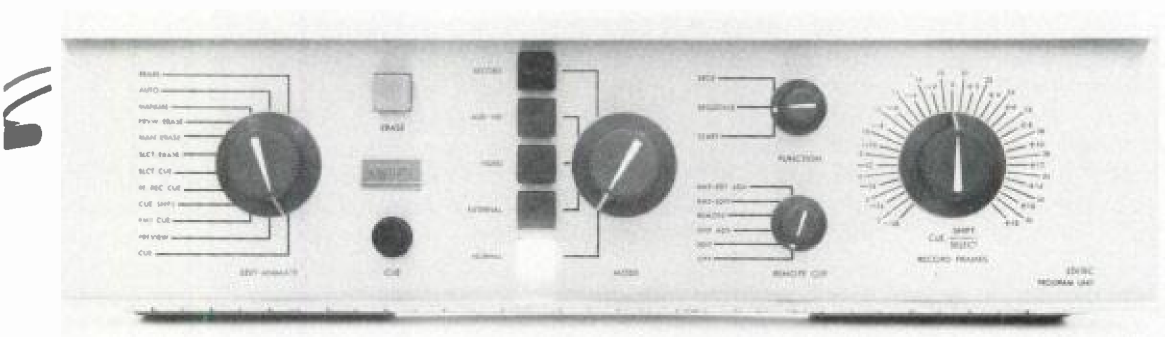


Fig. 1. Ampex Editec. (Courtesy Ampex)

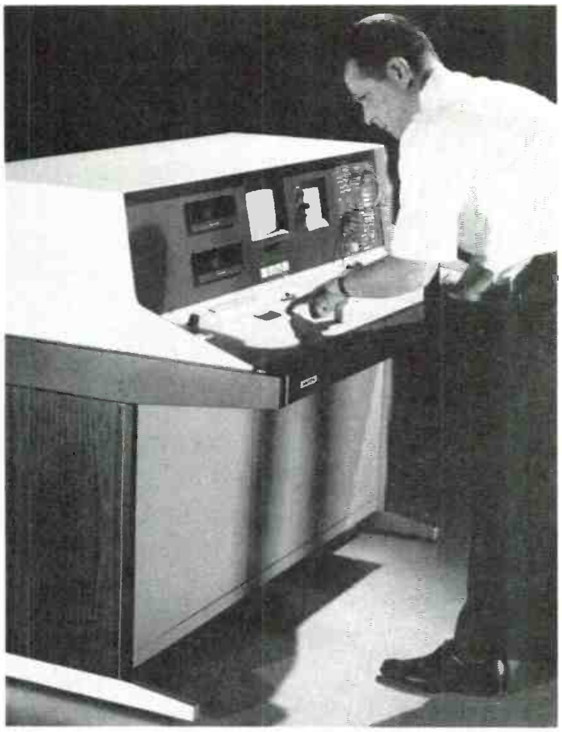


Fig. 2. Ampex HS-200 Operating Console. (Courtesy Ampex)

The problems of transport control for single and multiple VTR editing were addressed by the use of a frame code in the Ampex RA-4000, see Fig. 3. A unique address for each frame was laid down during recording on the cue track. This number stream provided references for transport control servo systems, allowing a desired frame to be searched for automatically.

The technique of multiple source VTR editing became feasible with the use of frame code. With the code as a common reference, the multiple play VTRs and the record VTR could be automatically synchronized by the RA-4000.

The use of disc technology to preview edit points, and time code to provide random access, was integrated into an on-line/off-line concept exemplified by the CMX 600 and 300 systems, see Fig. 4 and 5. In the 600 system, a total of 30 minutes storage capacity was attained when recording single monochrome fields with audio tracks encoded into the front and back porch of the video signal using six disc packs.

Off-line editing using the 600 disc system generated a list of time-code numbers defining edit decisions, that could be printed out or punched onto paper tape. The CMX 300 on-line system loads the paper tape into memory and runs a pro-

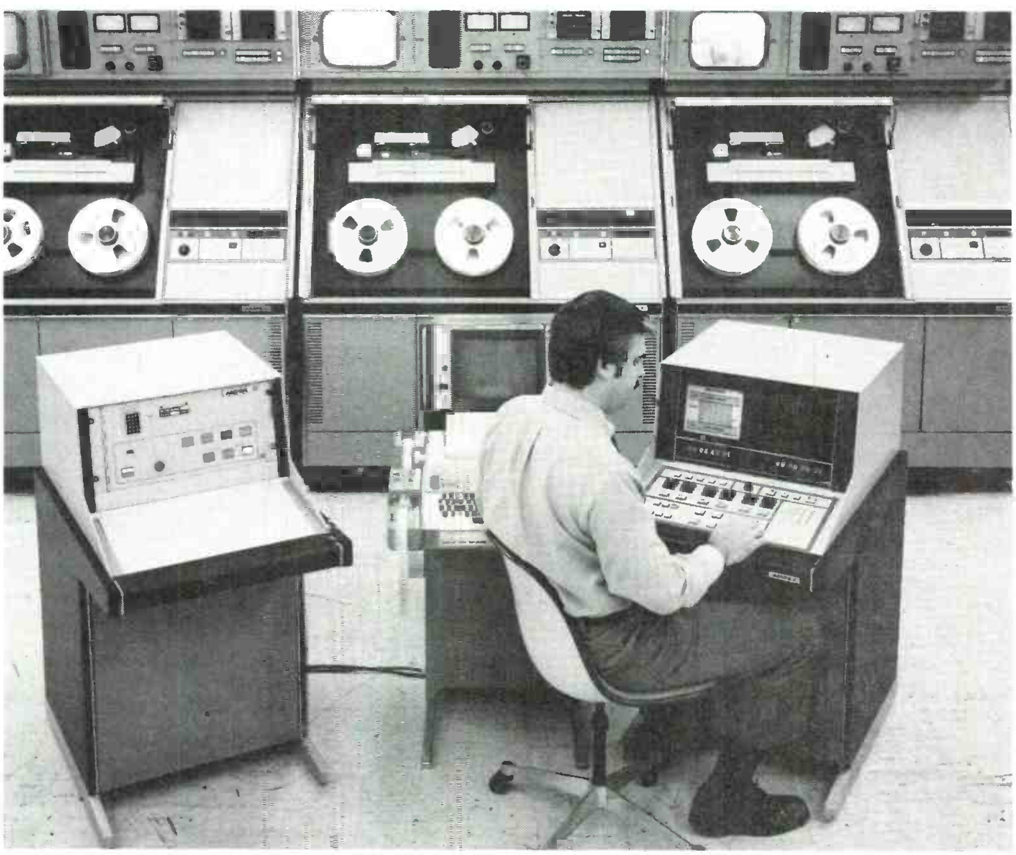


Fig. 3. Ampex RA-4000 Controlling AVR-1 VTR's. (Courtesy Ampex)

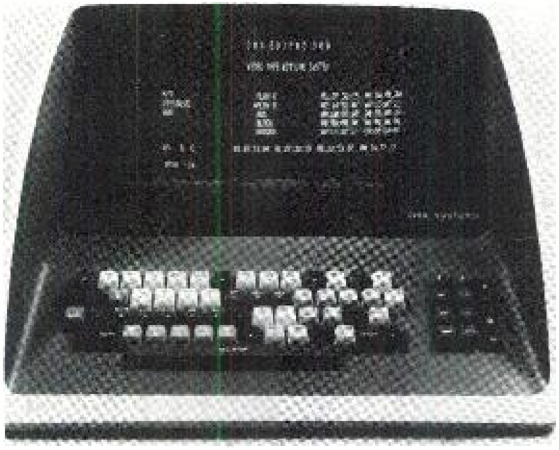


Fig. 4. CMX-300 Edipro.
(Courtesy CMX)

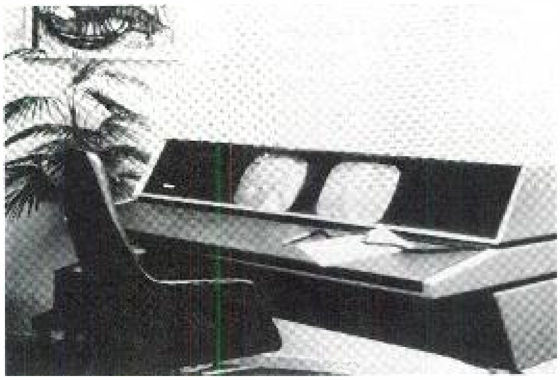


Fig. 5 CMX-600 Operating Console. (Courtesy CMX)

gram that uses the data to automatically assemble the master using quad tape.

The introduction of broadcast-quality, full field helical VTRs and their ability to view a still field produced the next major advance in editing technology. This removed the need to utilize disc technology to compensate for the quad VTRs shortcomings by providing three hours of recording time, viewable pictures while searching, and slow motion/still frame ability.

The current generation of editing controllers takes advantage of the helical scan recording technology and the increasing power of inexpensive microprocessor based computers. This combination has allowed video tape editing technology to come closer to combining the techniques of film editing with the advantages of magnetically stored video.

VTR EDIT TIMING

All external video tape editing control systems or edit controllers, rely on the record VTR to im-

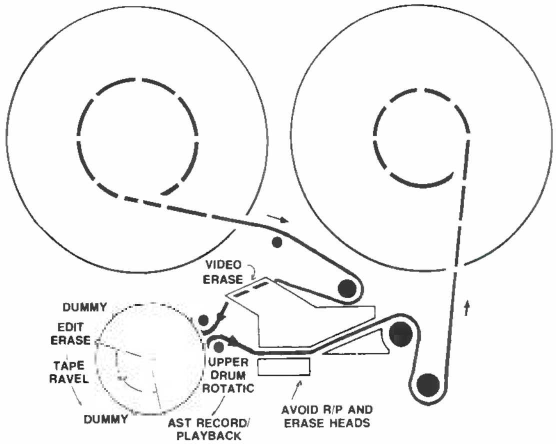


Fig. 6. Head position diagram Ampex VPR-80.

plement the edit electronic sequence. An operator command to the edit controller keyboard determines the place to perform the edit, the selection of tracks to be edited, the duration of the edit interval, and the editing mode, but the editing record VTR provides the internal systems necessary to switch servo references and sequence the erase heads and record/play electronics. These timing sequences are determined by the physical constraints of the tape format, involving the video and audio head positions and tape speed. The following example is based on the operation of the internal edit timing system of a type "C" one inch VTR.

The reception of an edit record command starts a timer that results in the operation of the flying erase head. This head is located in the video scanning assembly, see Fig. 6, and provides erasure of video tracks that have travelled past the stationary video erase head prior to the start of the edit interval. The flying erase head traces the same path as the helical record/play head and therefore can erase only the video track.

The record/play relay switches to the record position after the second timer in the sequence expires, and is followed by another timer delay, that activates the record amplifier and the edit interval recording begins. At the end of the video edit interval the sequence is run in reverse. The flying erase head shuts down first, then the record/play relay switches back to play, and the record amplifier is turned off. To perform an audio edit, the edit timing systems control the audio erase current and the record switching for any or all channels. The audio erase current is applied to the erase head in advance of the edit point. This timing advance compensates for the distance between the audio erase head and the record/play head stack. Upon arrival of the erased segment at the record/play stack, the record amplifier is ramped-up by the completion of its timer cycle, and the audio "edit-in" is performed. Conversely at the edit-out point the erase command is removed several frames before the edit-out point reaches the R/P head and the audio record amplifier is ramped down upon reaching the edit-out point. Attention should be paid to these ramping characteristics when moving a tape between recording VTRs of different manufacturers in order to avoid audible discontinuities at edit points.

The internal VTR editor can perform edits in two modes: *Insert* and *Assemble*.

Assemble mode editing is similar to non-edit recording in that the original control track is erased and a new one recorded. Upon arrival at the edit-in point, the new material is added onto the previous material in a building block fashion. The new material being recorded then becomes the "previous" material to which another segment may be added. Consequently, an edited tape is built by adding segment to segment to segment over previously blank tape. This assembly of new material onto previously recorded blank material typically occurs for all tracks. The typical VTR utilizes two full width erase heads thus precluding separate track editing.

In the *Insert* editing mode, the newly recorded material replaces previously recorded material for an edit interval duration shorter than the original recording length. The new material is sandwiched between previously recorded material at the edit-in and edit-out points. The full width erase head is disabled and erasures occur on a per track basis. The original control track pulses are maintained reducing servo instability caused by differences between the original and new recordings.

Basic Time-Code Editing Controller

The following discussion is an operational overview of a "general model" micro computer based time-code editing controller. It is used to illustrate the basic concept upon which larger systems are built, A/B roll or multisource editing. See Fig. 7.

In this example, the record VTR internal editor functions as before. Commands to the VTR internal editor are now generated by editing software, processing user commands from the editor keyboard. In the standby mode, the host computer running the operating system awaits a user command. Selecting a key on the editor keyboard sends a character to the keyboard port located in the computer system where the CPU initiates the program necessary to service the requested function. In the case of a play command for VTR-A, a play command is sent to the VTR-A

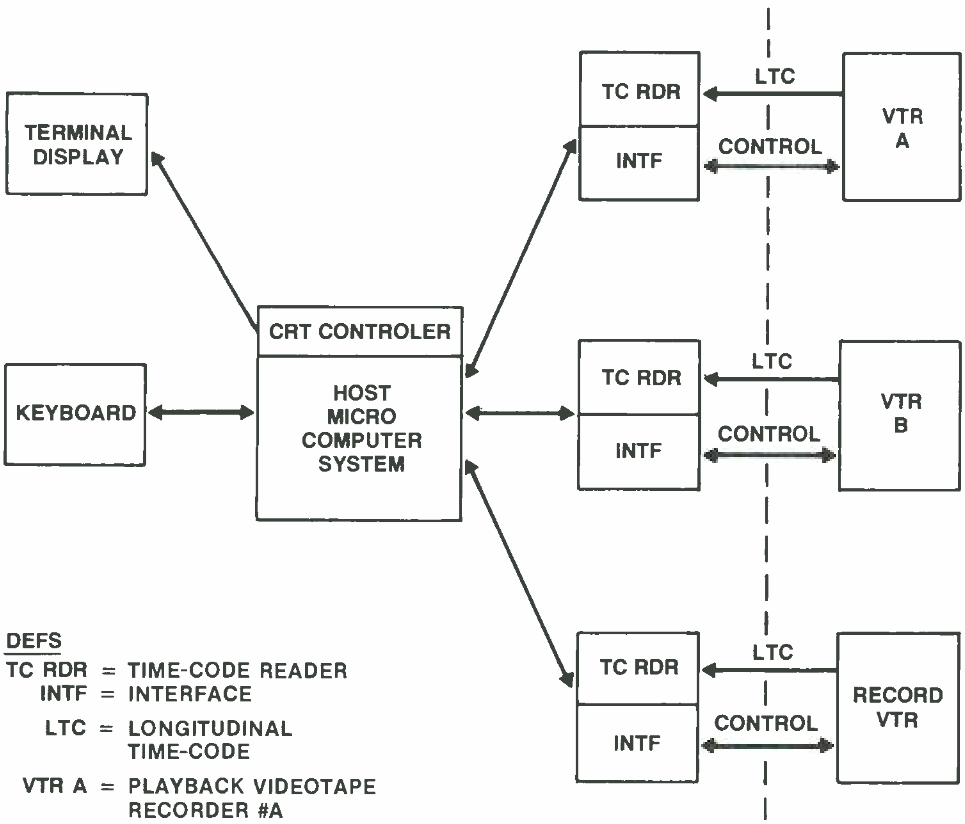


Fig. 7. Basic time code editing controller and A, B, record VTRs.

interface microcomputer subsystem. The micro-subsystem stores the command received from the host, initiates a local routine and produces a play function at the selected VTR. Since control methods vary widely between VTR manufacturers, the play command received requires format and protocol conversion. The interface runs a conversion program written to match the protocol characteristics of the dedicated VTR, and then the command signal is hardware converted to match the VTR port configuration. Confirmation or tally that the controlled VTR has received the desired command follows a reverse process back through the same chain to the keyboard, lighting a tally at the associated key. Concurrently with the initiation of the VTR mode, the corresponding time-code reader converts the SMPTE Time-Code bit stream into a suitable parallel form, making it available to the system buss where the CRT controller uses it to up-date the operator's time-code display. If the operator selects a certain location in tape as a desired edit-in point, that time-code number is transferred from the data buss to the system memory address defined as a VTR start time for the current event number. The same method applies to selecting a VTR edit-out point except for the memory address. Similarly, the record VTR is commanded into the Play or Shuttle mode until an edit point is located and matching time-code number is transferred to its particular memory address. If the operator elects to preview the edit, the host computer calculates the cuepoints by subtracting the preroll time from the edit-in point. The amount of preroll for each transport has been assigned by the operator via the keyboard or default values resident in the operating system. The interface computer runs a search program to determine the direction to the desired location and rapidly searches the VTRs, stepping down the shuttle speed as the cuepoint is approached. As this point is approached, the VTR is placed into the play mode and upon arrival, parked. As each VTR reaches the cuepoint its respective slave interface returns a cue status to the host computer. Upon receiving confirmation of cued status from all VTRs involved in the preroll, the host computer issues play commands to all machines. As the VTRs approach the edit-in point the host monitors the synchronization error between Play VTRs and the Record VTR. The difference between how far the Record VTR must go to reach the edit-in point and how far the Play VTR must go to reach the same point is termed *sync error*. The host sends this time difference sync error to the slave microcomputer where it is converted into the appropriate speed-up or slow-down command, thus enabling the Play VTRs to reach the edit-in point synchronized with the Record VTR.

If a VTR fails to arrive at the edit-in point by this synchronization loop, a sync error signal is displayed and the entire system retries the preroll. In order to insure that the VTR will arrive in synchronization during the next attempt, many systems will automatically add preroll time. If this attempt also fails, the system will try again with an even greater preroll interval. A non-sync condition after three attempts will cause a halt to the synchronization process, and operator intervention is required. If synchronization between transports is within the acceptable error prior to the edit-in point, the preview will continue. Prior to reaching the edit-in point, the host computer must also receive a servo locked response from the slave computer. The VTR interface slave monitors the horizontal and vertical servo lock signals from the VTR, and if both are satisfied before the edit-in point is reached a servo confirm tally is issued to the host and the preview continues. If failure of the VTR to achieve horizontal and vertical lock prior to entering the edit interval results in a preview abort, the system will re-try as for a sync roll failure. The Record VTR and the Play VTR time code are compared for color frame match. Color frame status is displayed and, if a match is achieved, the preview will be performed by switching from the Record VTR playback at the edit-in point, the "E to E" mode allowing the Play VTR to be displayed up to the edit-out point where switching reverts back to the Record VTR play video. If colorframing mismatch occurs, the operator is requested to offset the play VTR edit-in point one frame. This restores the four field color sequence described in EIA Proposed Standard RS-170A. See Fig. 2 on page 75, Section 5 Chapter 4 of this *Handbook*.

Edit recordings are performed essentially the same way as edit previews. The observable exception being that the Record VTR switches from playback mode to the record mode at the edit-in point.

Medium Scale Time-Code Editing Controller

The previous section described the operation of a relatively unsophisticated time-code based edit controller. It allowed the operator to remotely control the VTR functions, to select edit points using time-code numbering, and to synchronize multiple VTRs in order to preview and perform edits. The following description is based on one example of a moderately-priced microcomputer-based editing controller suitable for advanced broadcast production application, the Paltex Edit-Star ES-1, see Fig. 8. It may be considered representative of other editing controllers in this category, which operate in essentially the same

Fig. 8. Paltex Edit-Star ES-1.
(Courtesy Paltex)

manner. The first or "software" section is a condensed description of what such a system can be expected to accomplish and the parameters required. The second or "hardware" section describes the task of integrating the editing controller into real world equipment.

Upon turn-on of any sophisticated editing controller, various initial conditions must be entered. Set-up data is usually entered at the beginning of a new session and can be completely or partially up-dated as the session progresses. Typical parameters required by this initial dialogue are shown in the following list.

Preroll Time

The amount of time before the cuepoint that the system will backspace the VTR. This number is usually expressed in seconds or frames.

Postroll Time

The amount of time the VTR will continue to roll after the edit event has been completed. This allows editing continuity to be observed and is often expressed in seconds or in frames.

Reaction Time

This allows a system to compensate for the reaction time of the operator when marking edit points while tape is moving and is typically expressed in frames or seconds.

VTR Assignment

This identifies to the system the operational function to be assigned to the interfaced VTRs. Any VTR can be delegated as a player or recorder under software control in the ES-1.

Reel Number Assignments

In order to construct a useful edit decision list, the material edited must be traceable. Reel numbers are the only library key the system has.

Event Number

Provides the editing system the current event number assigned to the edit decision first completed after set-up. Edit events are numbered sequentially ascending from this base number.

Time Code

Assigns system Time Code operation to drop or non-drop frame. Automatically monitors the status of incoming time code from each VTR and will override an incorrect manual selection in order to provide proper calculations.

Record VTR Color Framing

Selects use of system or Record VTR color frame. A test edit is performed and color frame reference is selected to eliminate horizontal shifts if system color framer is used.

Time-Code User Bits

If present the system time-code reader will display the data read from the user bits in the space normally allotted to the SMPTE time-code values.

Edit Interval/Duration

Allows the selection of edit duration calculation method. Duration can be that of the Record VTR or the interval of the effective play source.

Display Out/Duration

Depending on operator preference, allows for the display in the edit decision list to record VTR out-times or the edit duration.

Once this initial information has been loaded into the operating system program, the editing operator is now ready to start an editing session using the editors VTR manual functions.

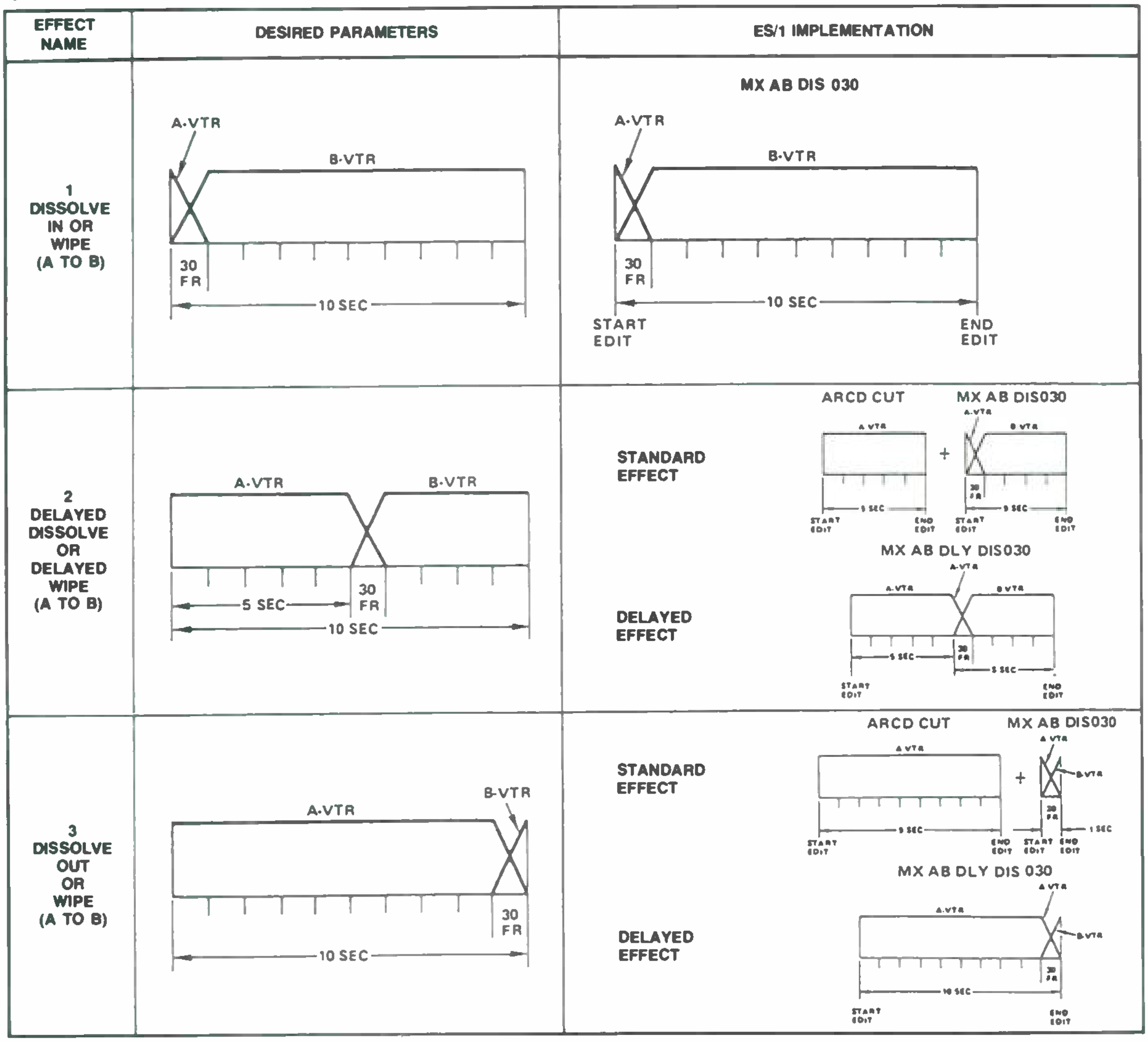
EDITOR VTR CONTROL

A manual function is a basic VTR command that has been remoted from the VTR front panel to the edit system control panel. The manual functions on an editing controller include Stop, Play, Record, Rewind, Fast Forward and Pause, along with the selection of which VTR they control. In the ES-1, controls defining each of these functions are used on a per key basis. A rotary control is used to simulate the shuttle/job knob found on helical VTRs.

An edit entrance point is selected in a variety of ways using editor manual controls. One method is entry of the time-code number via a calculator pad. In this method the operator shuttles or jogs tape to identify the correct edit point for each or all tracks, and enters the numerical data from the time code displayed by hand. This technique requires excessive operator intervention. A more efficient method for transferring this information is accomplished by the "on-the fly" or "mark" method. In this technique, edit points may be selected while tape is rolling by pressing a single, dedicated key. By selecting either the In or Out key the entire time-code number is transferred at once, instead of loading each digit via a key pad. This technique can be applied to the selection of all tracks or split tracks and it can occur quickly, while the VTR rolls, without requiring each point to be located with a still frame. Once an edit point has been selected by either method, trimming of an entry can be accomplished by either entering a new number or by shifting the existing point.

The majority of tape editing occurs in a sequential order. After the completion of the first edit, the next edit is butted adjacent to the first edit-out point. Instead of having to re-enter the out-

TRANSITION FORMATS
A. DISSOLVE IN
B. DELAYED DISSOLVE
C. DISSOLVE OUT



NOTE

Commands and Edit Decision Reports shown pertain to dissolve effects only. WIPE is substituted for DIS for wipe effects.

Fig. 9. Transition formats. (Courtesy Paltex)

time of previous edits, a dedicated key provides transfer of the previous out-time to the current edit-in time.

Once an edit-in time has been established, the edit-out time can be manually selected by using similar techniques. Often specific duration requirements will define the edit-out time. The calculation of the edit-out time can be done manually, using the electronic scratch pad, or provided automatically, in much the same way an electronics spread analysis sheet program calculates new data based on changing input variables. If the edit-out time is known and the edit-in time undefined, the duration can be entered and the edit-in time will be calculated.

Transition Selection

After the selection of the play/record segments to be used in the upcoming edit, the transition between segments must be specified. The variables to define are:

1. Type of transition (cut, wipe, dissolve, or key)
2. Duration of transition in frames.
3. Tracks affected: Video, Audio 1, Audio 2 (V,A1,A2)

The cut mode is the simplest and most direct transition between two sources. The second most common transition is the dissolve, see Fig. 9. Dissolves may be as *dissolve-in*, when the dissolve occurs at the edit-in point, or *dissolve-out*, where the dissolve occurs at the edit-out point. Delayed dissolves place the transition between the record edit-in and out points. Entry of a standard dissolve as an edit point requires selecting the dissolve start point and the dissolve duration. In the case of a delayed dissolve, the delay time must also be entered. This time must fit between the start of the edit-in point and the actual occurrence of the dissolve. The same considerations apply to wipe transitions.

The specification of a key transition involves defining how the key is to be brought in or taken out. The key transition commands, in their most general forms, are: Key-in, Key-out, Fade key-in and Fade key-out. The key-in command causes the foreground image to dissolve in over the background image at the specified rate. The fade key-in causes the foreground and background images to fade up together from black at a specified frame rate. The key-out and fade-out commands are used to terminate a key effect in progress.

Due to the tremendous growth in switcher and digital effects technology, many transitions necessary in modern post production are not addressed with specific edit decision list labels found in many EDL formats. In order to trigger these auxiliary device transitions, an event is often defined by a specific numeric code designating the type of device and function required. This special function code is stored with the time-code number designating the beginning of the transition.

Edit Preview and Recording

Once transitions have been selected, preview modes are used to rehearse an upcoming edit. An automatic preview cycle consists of cueing and rolling the designated VTRs through the pre-roll time while synchronizing in time to be locked and colorframed upon arrival at the edit-in point. After passing the edit-out time, all VTRs continue to roll for a pre-set length of time.

Edit previews may be viewed in a number of formats. A Video-Video-Video (VVV) mode presents the record machine playback video up to the edit-in point at which the new source material is switched-in by the preview switcher, and then switched-out at the edit-out time back to the record machine video. The Video-Black-Video (VBV) mode inserts black for the new source material, in effect creating a black hole. The Black-Video-Black (BVB) mode, conversely, presents only the newly inserted source material sandwiched between black segments.

Once a previewed event has been deemed acceptable, an edit record command is used to initiate sequence to the preview except that the source material is routed to the record VTR and recorded onto the tape. The event data are also stored in the edit decision list.

The edit source VTRs are recorded onto tape through the video switcher and audio mixer for the duration of the edit interval with effects transitions occurring as specified in the preview mode. At the conclusion of the edit record, the current event number will be updated to the next consecutive number. In the duration display, the show duration is updated to include the duration of the edit that has just been completed.

An alternative editing technique to A/B roll is the sync roll. Sync roll editing is the process of cutting between continuously rolling VTRs, each locked to the timetable of the original recorded event. Editing is done by switching between VTR sources much like switching between cameras during a live production. Just as in studio production, edit points can be selected on the fly, in "real event time". This technique assumes that the source material has been recorded on a separate reel for each source.

To achieve synchronization between event time related audio/video tracks with different time-code numbers requires that the operator manually synchronize the VTRs within a few frames of each other. Once the VTRs are within a few frames of each other the operator can enter the required offset to the out-of-phase VTR, and the system

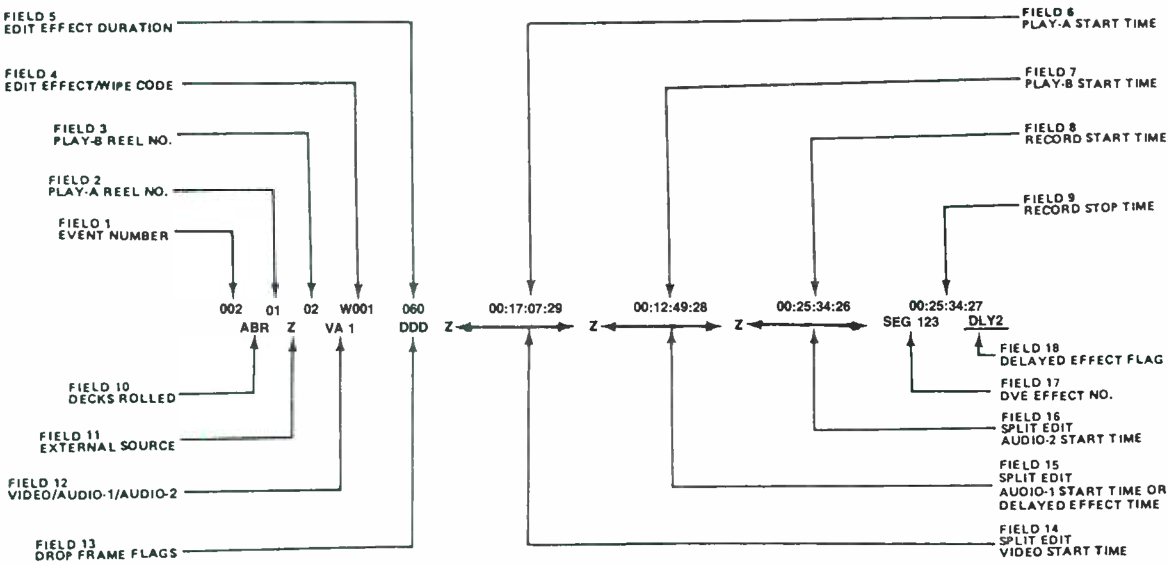


Fig. 10. Edit decision list format — Paltex ES-1. (Courtesy Paltex)

will automatically resynchronize. The offset required is then added to the edit-in time of the adjusted VTR.

The actual performance of a sync roll edit consists of a sync roll preview and then automatic assembly using the edit decision list produced in the preview mode. All VTRs re-cue and resynchronize as in the preview mode. However, source selection is now performed by the edit controller, according to the edit decision list, until the edit interval ends.

Edit Decision List

The edit decision list is a formatted data base defining a series of edit events. This data base is organized into fields by the editing equipment manufacturer. An example of an edit decision list format can be seen in Fig. 11.

These fields contain sufficient information to reconstruct a series of edit events from source material. This list may be generated by a main or on-line editing system or by a smaller scale off-line system.

An off-line system may consist of an ENG style cuts-only editor utilizing dubs of the original tapes with the "burned-in" time-code characters in the active picture area.

As approximate edit points are determined, time-code numbers can be manually logged for later entry into the on-line edit list. This approach reduces the amount of search time used in the on-line system.

Edit decision list information can also be manipulated using personal computers. A list can be loaded into a pc then edited via the keyboard and loaded back into the editing system using a floppy disc as the transfer medium. The pc can perform many of the sophisticated list management chores normally relegated to the on-line system. Systems that allow tc editors to transfer banks of edit information directly over a serial buss to a pc enable the pc to perform as an outboard data base manager and memory expansion.

Currently there are numerous EDL formats in existence, representing efforts to find a better way to preserve records of the edit events. Fortunately, with the drastic decrease in price of computing power, more editing manufacturers are providing for alternate formats in the hope of attracting a larger market through increased list interchange ability.

An effort to standardize on a common format is currently being attempted by the Video Recording and Reproduction Technology Committee of SMPTE. As of this writing the proposed format resembles the original CMX style format but provides additional inclusions of user data, virtual events and commands.

MEDIUM LEVEL TIME-CODE EDITING-CONTROLLER HARDWARE SYSTEMS

The hardware of many editing controllers can be divided into two major systems. The first contains the host microprocessor and I/O devices similar to those found in personal computer systems. The second major system category includes custom interfaces between the system buss and the video/audio signal equipment. The combination of these major systems, operating under editing application software, comprises the overall editing controller system.

Most modern editing controllers use a technique known as distributed processing. This technique divides the computing task between a host computer and a number of slave computers. The host performs the task of systems supervisor, while the slave systems are dedicated to specific tasks. In the case of the VTR interfaces, each has a dedicated microcomputer slave system running a software routine specifically written for that type of VTR. This division of tasks has a number of advantages over a single centralized processor. Failure in one subsystem need not halt the overall process. In the case of cueing multiple VTRs, one failure to cue does not inhibit the cueing of others.

System Description

In Fig. 11 the host is shown communicating with the computer support and I/O devices. These computer devices consist of: the keyboard, the CRT controller, printer interface, and disc I/O. These devices combine with the host to provide a customized computer system optimally designed for the editing application.

The system host processor runs the main editing application program from internal programmable read-only memory (PROM). Once running and communicating with the CRT controller and keyboard, the host is ready to run an editing function where, operating as the central processor, it will arbitrate the data buss and provide service to each peripheral.

The keyboard, see Fig. 8, provides the major operator control point in the computer assisted editing system. Operationally there are two basic sections, the 86 key command section, and the rotary encoded VTR transport control. The keyboard area provides selection of system commands combined with numeric entry. Extensive use is made of dedicated keys in order to reduce the number of key strokes necessary to perform common functions. Certain keys are left as "user-definable". This allows the user to create custom commands consisting of a memorized sequence of other key strokes. This feature resolves the problem of trying to anticipate all desirable keystroke functions.

To provide hard copy of edit decision lists, the computer I/O system contains an EIA RS-232 serial port that is usable with most common printers. In order to use devices such as the Teletype Corporation's ASR-33, the serial port also provides a current loop driver for printer selec-

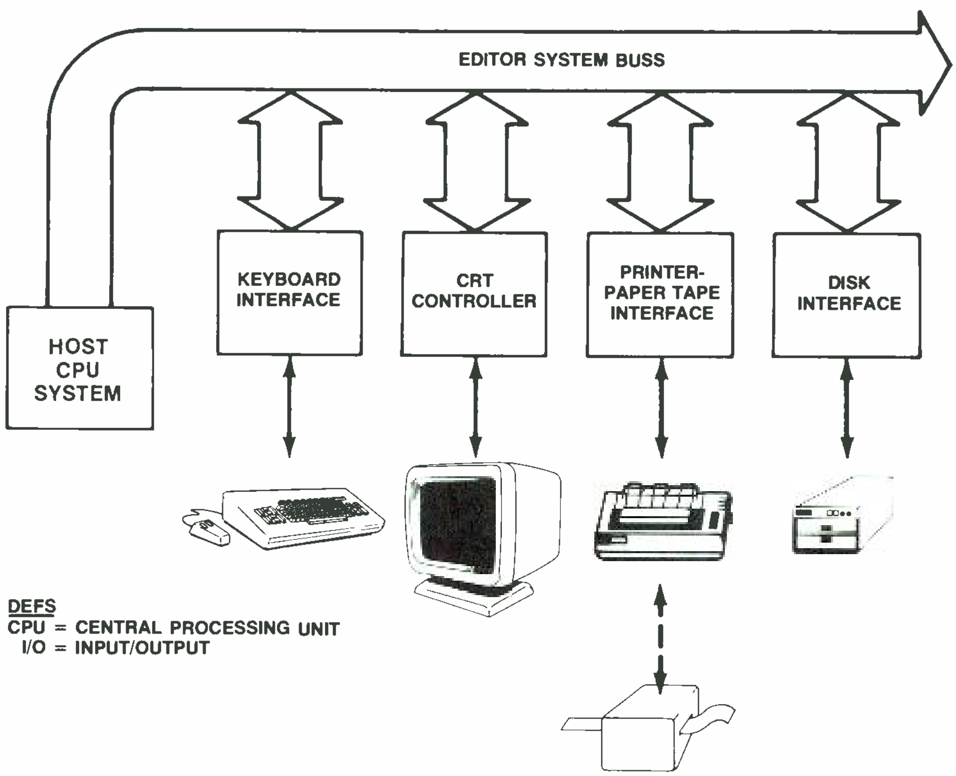
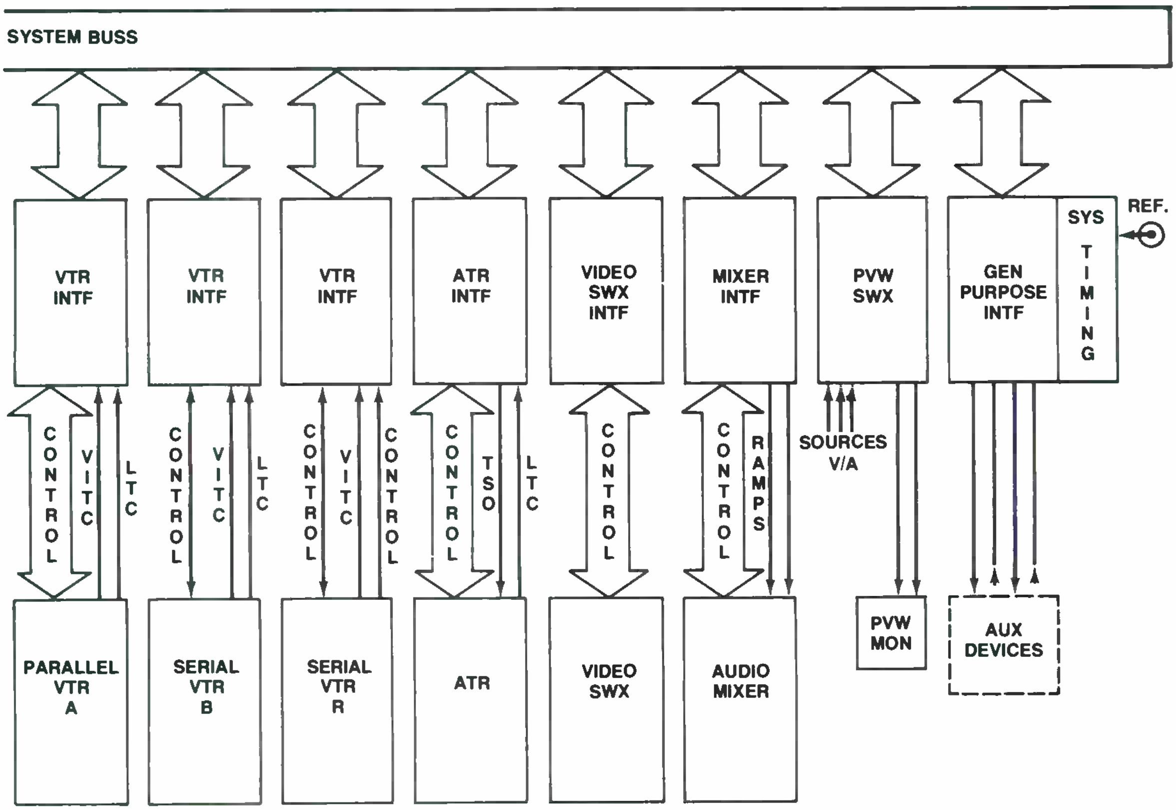


Fig. 11. Computer support and I/O devices.



DEFS
ATR = AUDIO TAPE RECORDER
PVW = PREVIEW MONITOR
AUX = AUXILIARY
SWX = SWITCHER
SYS = SYSTEM
INTF = INTERFACE

Fig. 12. Editor buss interfaced video and audio signal system devices.

tor magnets and internal paper tape reader/punch. Although paper tape is relatively slow and very noisy, there are some advantages. Paper tape is directly readable, printable, cannot be accidentally erased and is often the only medium available as removable storage on many of the smallest edit controller systems.

To provide random access storage for edit decision list information, the microcomputer I/O system also contains a serial interface to floppy disc drive unit. Floppy disc memory provides higher speed density and quieter operation than the paper tape storage technique. Edit decision lists can be loaded, changed, and rewritten easily; something that is a bit difficult to do with paper tape. An edit list can be stored by dumping the complete list to a file at the end of a session or by using the Protect mode, individual edit events are written to disc upon performing an Edit Record. In order to facilitate interchange, many manufacturers offer their own disk size and format and eight inch disk drive system conforming to SMPTE RP 132.

The second major system that the hardware can be divided into consists of the collection of interfaces used to control the video and audio signal system devices, see Fig. 12. This group of interfaces consists of: the VTR interfaces, the ATR interfaces, switcher interfaces, Audio Mixer interfaces, General Purpose interfaces, Preview switching and system timing.

VTR INTERFACES

Fig. 13 illustrates an example of a parallel VTR interface between a Type "C" VTR and an edit controller. Thirty-six command and tally lines interconnect the transport to the VTR interface on a one signal per wire basis. These signals may be divided into: transport commands, internal editor commands, tallies and control voltages. Transport commands such as Play, Stop, and Rewind are controlled by logic levels. Tallies from these commands are also returned by logic levels. Internal editor commands such as Insert or

Assemble are also selected by setting the respective control lines.

Exceptions to this simple "on/off" type of control are the tape speed override (TSO) and shuttle voltages. As their names imply, both are controlled by variable dc levels, each generated by a digital to analog convertor. Initial installation of a parallel interface requires calibration of buffer circuitry between the D to A converter output and the VTR analog inputs in order to match the required control range used by the VTR manufacturer.

The parallel interface technique is a very hardware intensive method from the viewpoint of the edit controller manufacturer. Since the editing controller is fundamentally a digital computer, all interfacing ideally occurs in the digital domain. As VTRs employ microprocessors for internal systems control, digital interfacing becomes the appropriate approach. By utilizing a serial data control link system, interconnection becomes simpler. Typically, a nine pin connector carries all the necessary lines to reproduce the functions provided by the parallel interface. Once encoded, the VTR control line can easily be switched between desired control points.

In order to apply the serial interface technique two links must be defined: the physical link and the data link. It is all too common to hear the terms "RS-232" or "RS-422" improperly used to define both the physical and data links. Either EIA standard defines a set of transmission signal levels and conditions for the physical link. The earlier RS-232 format utilizes "single-ended" or unbalanced techniques for driving the transmission line while the later RS-422 technique utilizes a differential or balanced line approach to improve noise immunity. Each standard defines a different voltage level representing data, and other line driver/receiver characteristics. Connector types and pin-outs are left undefined; baud rates and data formats are also omitted, therefore even though the two devices match the standard they may not be able to achieve communication.

An attempt to define these standards is contained in Proposed American National Standard PH 22.207 (reproduced at the end of this chapter). The signal conditions of RS-422 with the inclusion of a standard baud rate (38.4 kbd), interface connector (DB-25), pin assignment, plus definitions of data words, message paths and buss structure allow interconnection compatibility between different manufacturers.

Once the hardware is able to communicate using the same language, it is necessary to define who is going to speak to whom and when. This is defined by the SMPTE recommended practice RP113, reproduced for reference at the end of this chapter courtesy of SMPTE.

In Fig. 14 the same VTR and editing controller used in the parallel interface example have been reconfigured using the serial interface technique of PH22.207 and RP113.

Each VTR interface also contains a time-code reader as a subsystem. Its task is to convert off-tape time code into a compatible data format for use on the system buss and by the VTR interface. Two types of time code can be utilized by the reader: Longitudinal or Vertical Interval. Longitudinal Time Code (LTC) is recorded on a cue channel or, if unavailable, an audio channel. This code referred to as SMPTE Longitudinal Time Code is described in ANSI Standard V98-12 and can be found in the chapter on Video Tape

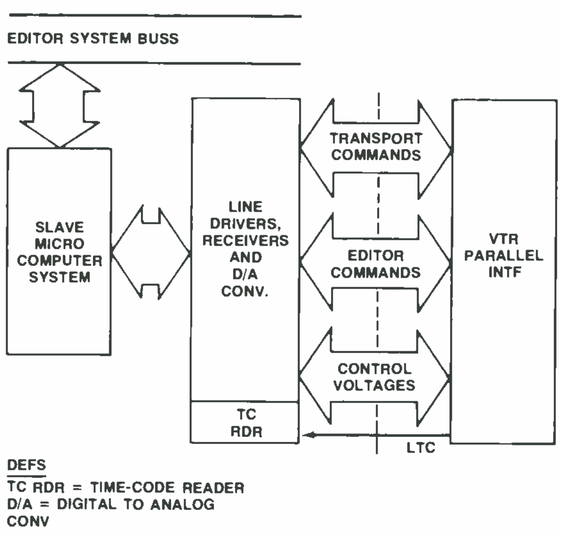


Fig. 13. Parallel port VTR and editor interface.

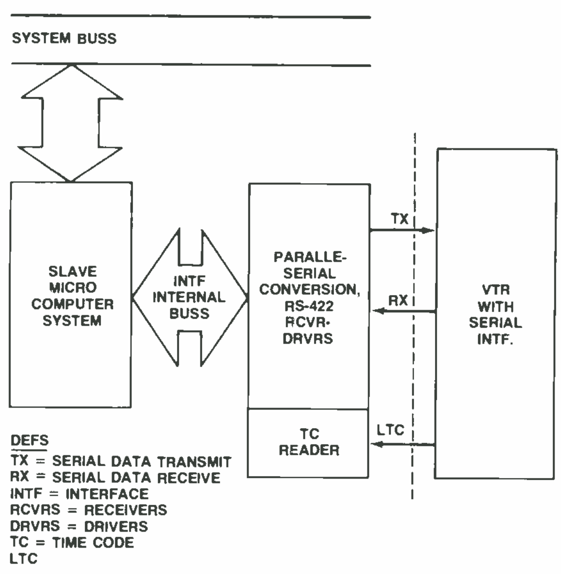


Fig. 14. Serial VTR and editor interface.

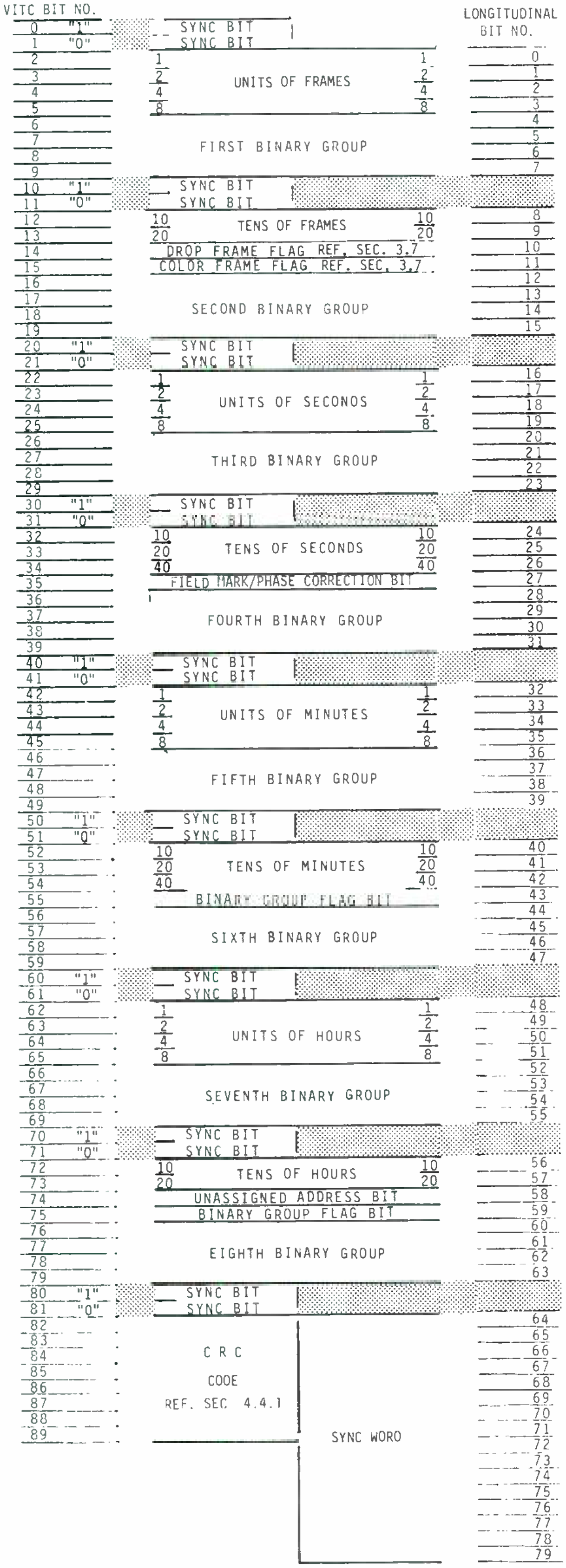


Fig. 15. Relationship of vertical interval time code to longitudinal code. (Reprinted with permission of SMPTE)

Recording, Section 5.5-105. LTC can be decoded at play or higher speeds, but as tape speed drops 1/20 of play signals reproduced by the stationary head become unreadable.

The time-code reader in this example can resolve this problem by utilizing a Vertical Internal Time-Code decoder. VITC contains the same information found LTC, but is inserted into the vertical interval of each field during a recording. During slow motion and still frame operation of helical VTRs, VITC data can be recovered from video. At high speeds where video is not recoverable the reader can revert to LTC.

VITC contains the same information found in LTC along with some additional improvements. An error check code called *cyclic redundancy check*, or CRC is included and VITC is recorded on two adjacent lines. These redundancy features help reduce code loss due to tape dropouts. Another improvement is the addition of field identification bit. In order to accommodate these additional features, VITC requires ten more bits than the 80 bits in LTC, see Fig. 15.

Audio Tape Recorder Interfaces

A multitrack audio tape recorder (ATR) designed for capstan servo and remote control may be interfaced to the edit controller buss much in the same way as a VTR, see Fig. 16.

Time code is recorded on a single track providing location reference. Off-tape time code is compared to the desired location in the interface microcomputer system which produces commands issued to the transport remote control until a near match is achieved. Frame lock is achieved by comparing the TC frame units to a reference frame pulse generated from house reference black burst. The resulting dc error voltage is applied to a voltage controller oscillator producing a tone proportional to the speed error.

The ATR accepts this speed tone as an input to the capstan servo comparator thus completing the overall servo loop.

An example of a suitable ATR for this application is shown in Fig. 17. Serial control is also available as the interfacing path utilizing similar techniques to those for serial control of VTRs.

Video Switcher Interfaces

The ability to control video switcher functions from the editing computer provides a number of advantages. Transitions such as dissolves, wipes or keys may be performed repeatedly without the inherent chance of error found in manual operation. This relieves tedium and provides continuity if edit lists are transported between similar editing suites. A number of interface configurations are

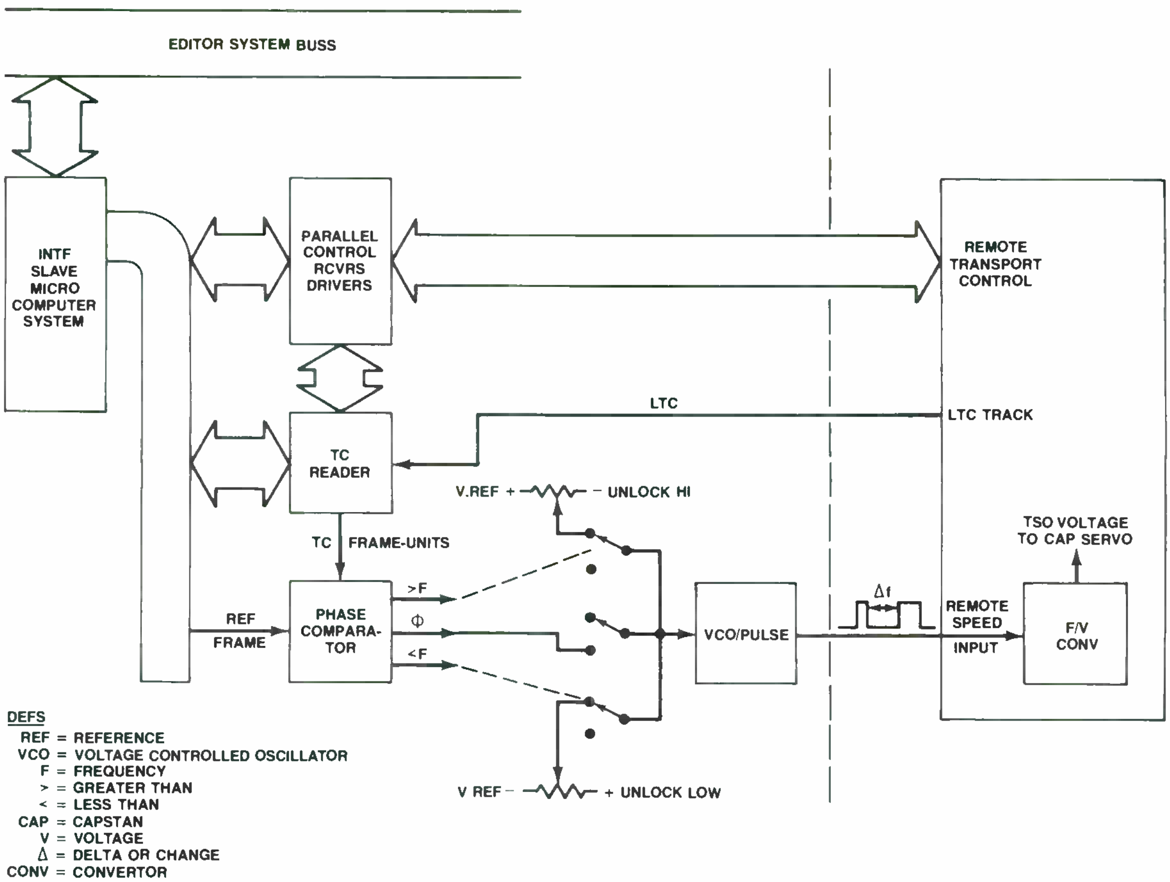


Fig. 16. Parallel port ATR and interface.

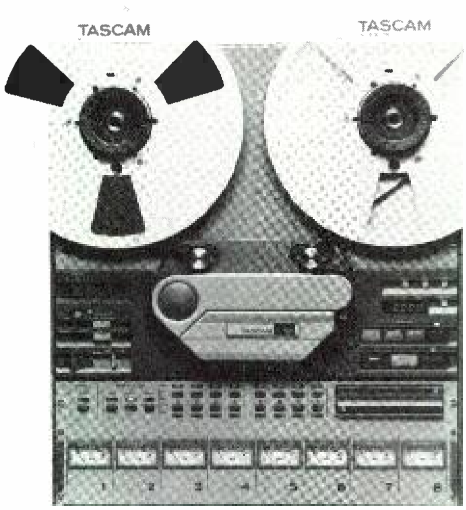


Fig. 17. Multi-track ATR suitable for editor control. (Courtesy TASCAM-TEAC)

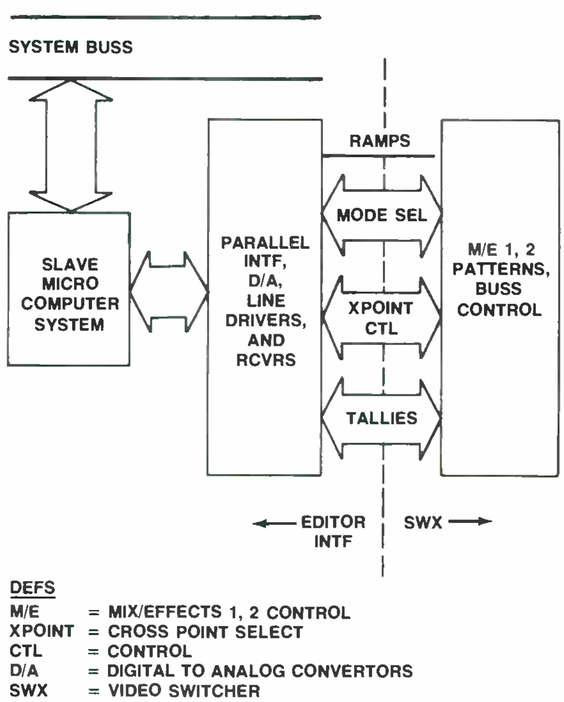


Fig. 18. Parallel port switcher and interface.

possible depending on the capability of the switcher. In Fig. 18 we see a parallel interface to a video switcher in which crosspoints, key functions, and wipe patterns are selected, and tallies received on a one wire per function basis. Fader arm and other control voltages are generated by a D/A converter and switched to the appropriate effects amplifiers to replace those produced by the manually operated transition lever arm. The wire per function method of the parallel interface technique becomes unwieldly as more functions are added to switchers. In Fig. 19 the task of automation has been divided between the switcher and the edit controller. Although crosspoints and effect types are selected by the parallel technique, the switcher contains an internal automatic transition system. Using this system, a transition may be manually rehearsed by the operator and automatically performed by the edit controller, recalling by the assertion of a single command. An example is shown in Fig. 20.

To alleviate the parallel wiring and to provide a memory system for switcher set-ups, an accessory is available for the parallel switcher, see Fig. 21.

This subsystem is inserted between the switcher control panel output and the switcher electronics frame control inputs, see Fig. 22. Manual operation of a panel control is converted from analog to digital form enabling it to be stored in memory. These data can be recalled and converted back to analog form for control of the switcher electronics frame. Instead of control panel information, data from the editor's serial interface can be converted to analog form for control of the switcher electronics.

As switcher architecture has evolved, the operator control panel can now be thought of as a specialized terminal. The various crosspoint selections, fader and knob positions are encoded into a bidirectional serial data stream. This data stream may be originated by the switcher control panel or an editor controller interface.

The edit controller's dedicated microcomputer system runs an interface program that converts data received from system buss into the com-

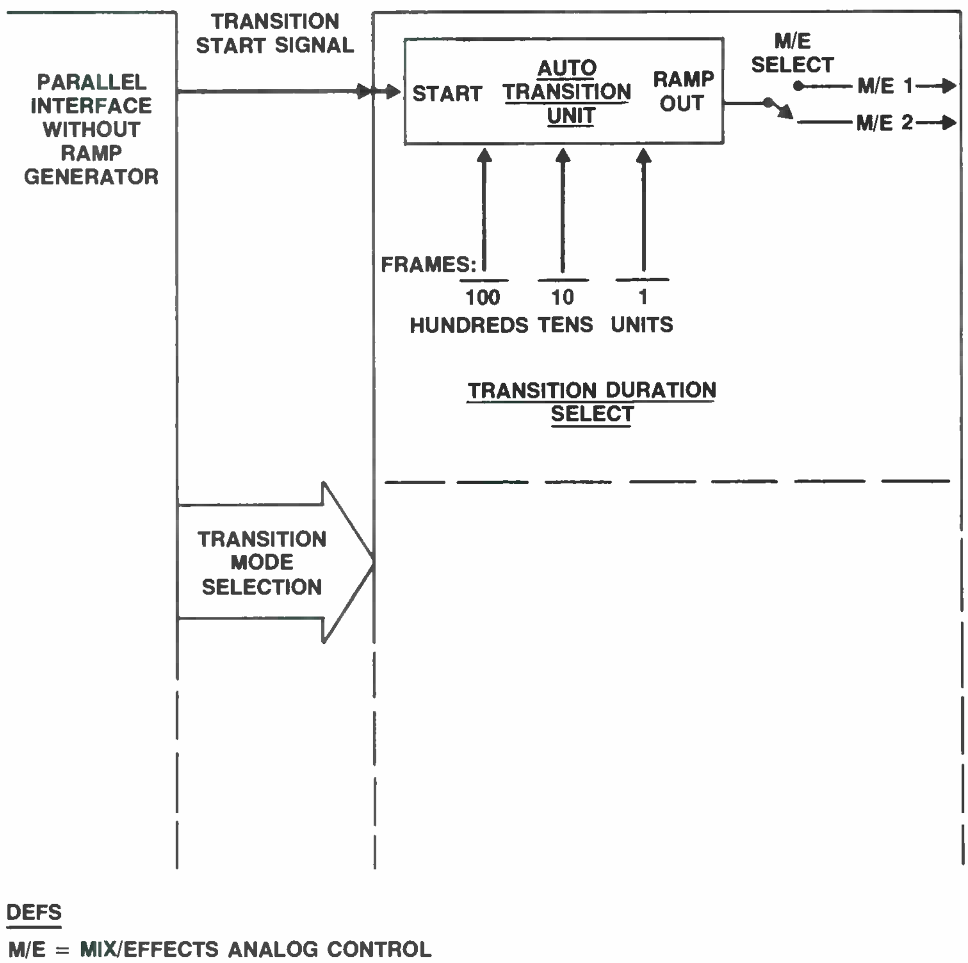


Fig. 19. Automatic transition unit added to parallel interfaced switcher.

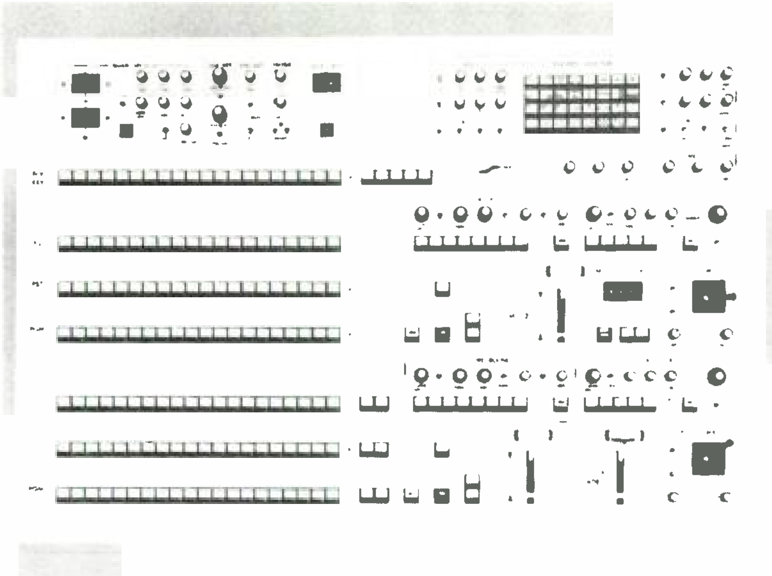


Fig. 20. Switcher containing automatic transition unit. (Courtesy Ross Video)

mand's specified by the switcher manufacturer. The switcher electronics frame decodes the data stream and implements these commands. Tally information sent in the reverse direction is decoded by the interface and placed on the system buss where it is available to the host.

The video switcher shown in Fig. 23 is an example of combining total serial control with automatic transition and set-up storage memory. A total of twelve switcher set-ups may be stored in the memory system and recalled via the serial interface. Multiple automatic transitions can be performed with complete control of the effects system.

Function selection and transition performance on the serially interfaced switcher occur by sending register select codes from the edit controller at the desired time-code trigger point (this information can be stored in the edit decision list and recalled later by using a special function cell of SFC). In this case, the SFC contains a number denoting a serial switcher and desired transition. Multiple SFC's can be activated during the edit intervals providing complete repeatable effects.

The ability to apply the SFC technique along with other powerful methods of switcher control via the edit decision list is currently hampered by the lack of interchangeable protocol and interface formats between manufacturers. In the case of the Ross 210 Switcher a decision to emulate the Grass Valley format allows one editor interface to be used with either manufacturer.

Digital Video Effects Interfaces

Interfacing to a digital video effects system (DVE) is also accomplished in this example by

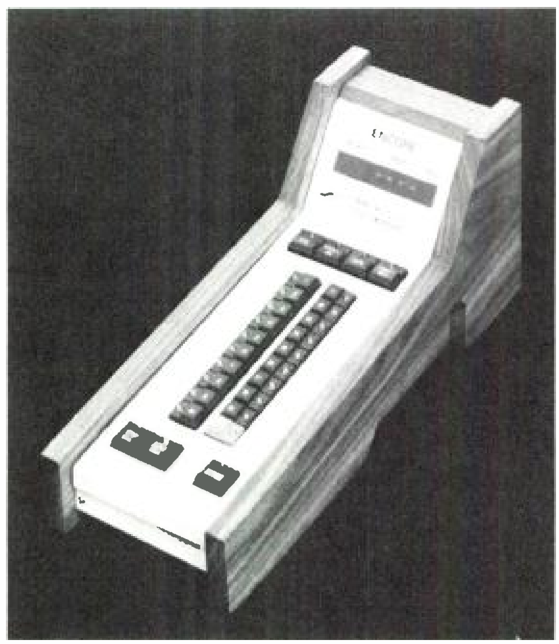


Fig. 21. Memory and serial interface for parallel port switcher. (Courtesy Ross Video)

use of the SFC system, relying on the ability of many digital effects systems to produce programmed effects on activation of a contact closure. The SFC device code for a relay closure is selected, which directs the software to the relay interface circuitry, and closes the appropriate contact pair. The edit decision list records the closure trigger time and the SFC code for a digital effect device. The manufacturer may offer a protocol

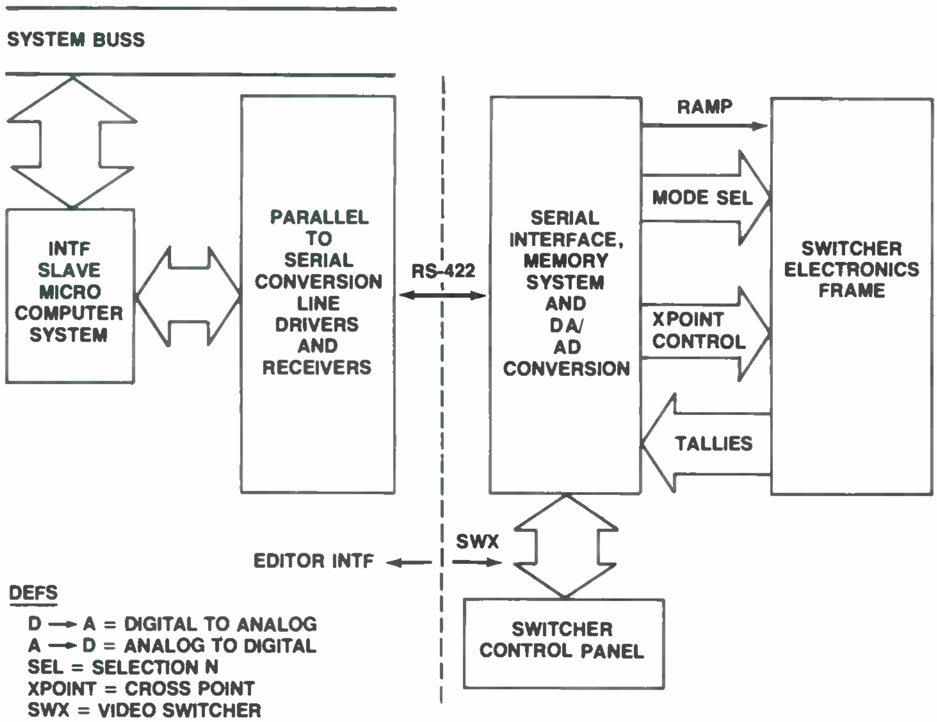


Fig. 22. Memory and serial interface system integrated into parallel port switcher.

Fig. 23. Ross RVS-210 (Courtesy Ross Video)

for an external interface designating each function. An SFC containing the required register selection can be used to select the desired effect under editor control.

Audio Mixer Interfaces

The ability to interface an audio mixer to the editing controller computer buss provides similar types of advantages to that of the edit system controlled video switcher.

In Fig. 24 we have an example of a parallel interface between the system buss and an audio console designed for editing application. Audio crosspoints are selected using a wire per function approach or by multiple data bytes transmitted sequentially across the parallel lines. These lines are optically isolated to insure minimal noise coupling between the interface and the audio console.

In order to provide remote control of mixing levels the interface provides an analog control voltage to the buffered input of a voltage controlled amplifier (VCA). This VCA can be located as a channel, submaster or master gain control cell. An example of an audio console embodying these features is shown in Fig. 25.

An alternative approach to the parallel interface method is exemplified by the audio mixer shown in Fig. 26. This approach utilizes the techniques of total serial control and automatic transitions. This device can perform as a stand alone audio mixer, under control of an editor serial interface, or slaved to a video switcher, see Fig. 27. By looping the video switcher interface serial control line through the AMS 210, commands issued to the video switcher can be followed by the audio mixer. The editing host controller running suitable software can select the desired crosspoint to follow video or provide split audio using one or two channels. Level trimming data are available to the edit controller to allow listings of trim values in the edit decision lists. Programmed transition modes available are "V"

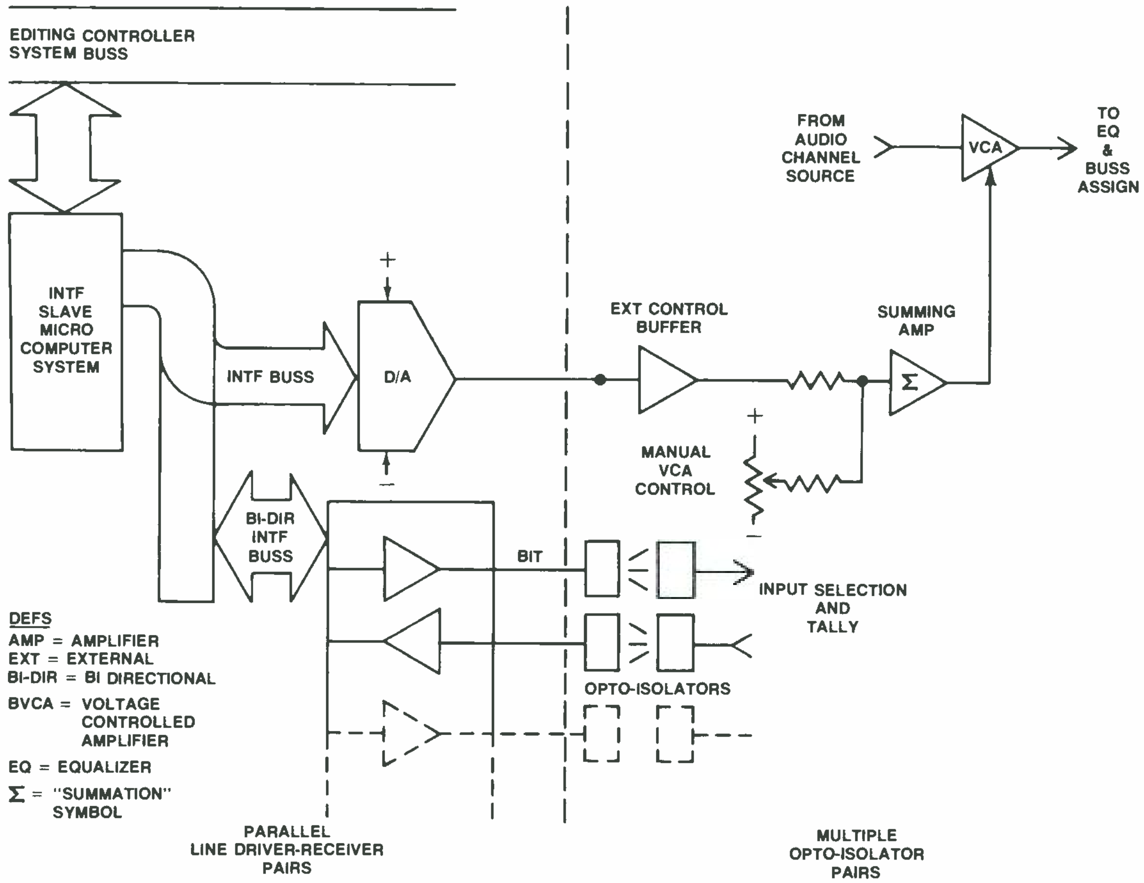


Fig. 24. Parallel port audio mixer with VCAs and interface.

Fig. 25. Post Pro 12 ADM Technology audio mixer. (Courtesy ADM Technology)

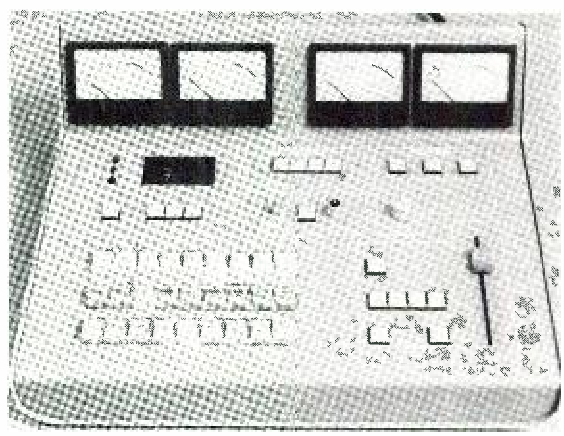


Fig. 26. McCurdy AM-210 post production audio mixer with serial control. (Courtesy McCurdy)

fades, cross fades, cut/fade and fade/cut. An "over" buss is selectable allowing common voice-over mixes to be summed into the master output.

Preview System

The Preview system provides an internal routing switcher for selecting the video and audio sources. Each VTR, Black, External Video, and Switcher output, with corresponding audio sources and audio mixer outputs is available on the switcher buss. Edit previews are automatically performed by the host CPU. Sources are switched at transition points to the preview audio and video

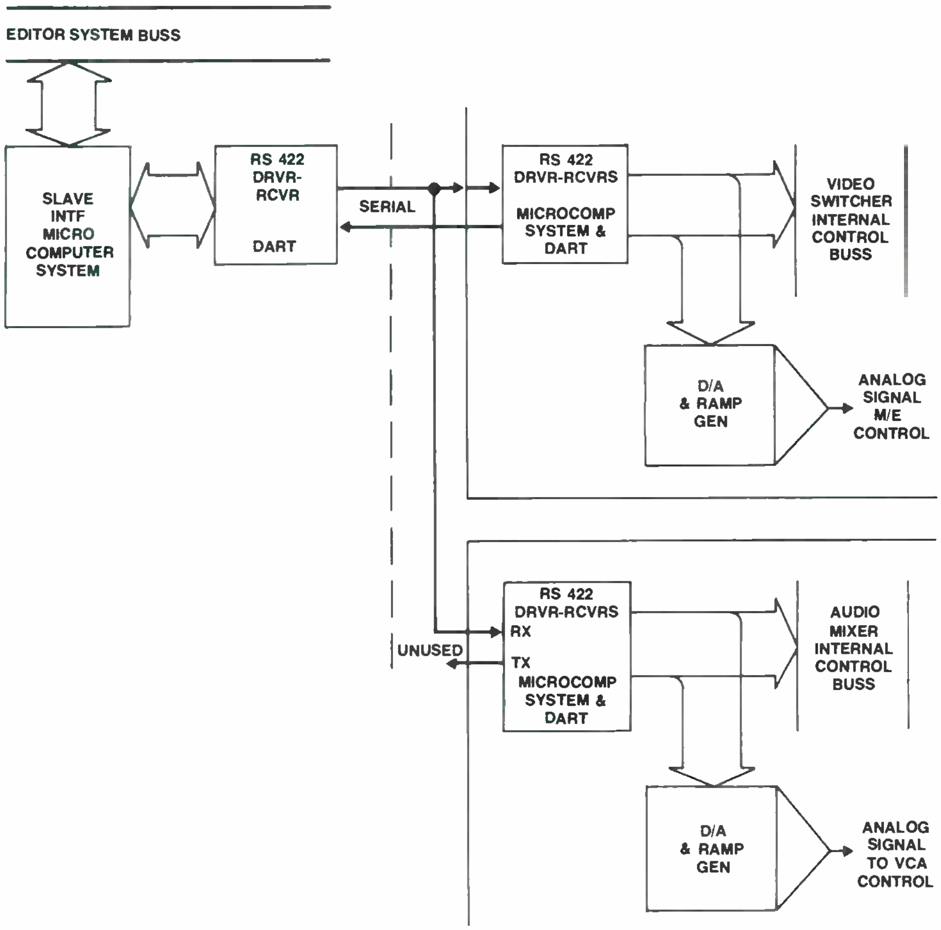


Fig. 27. Serial video switcher with audio mixer as slave.

monitors, thus simulate and edit. Inclusion of a preview switcher eliminates relying on a specific preview switching system being included in the record VTR.

General Purpose Interface

The general purpose interface system is the simplest and most versatile of the interfaces. Addressed by selecting a specific code for a special function cell, a contact closure or logic level can be asserted at a desired trigger time to activate external devices.

The triggered pulse is only a momentary command and may not be sufficient in duration for the controlled device, therefore a programmable timer allows the closure time to be lengthened in multiples of video frame duration.

Fig. 28 Editing suite utilizing external VTR machine room. (Courtesy Multivision)

EDITING SUITE DESIGN CONSIDERATIONS

Editing suite physical design comprises a sufficiently large subject so that it cannot be completely examined within the scope of this chapter. It is suggested that the broadcaster faced with designing an editing suite examine those found in commercial post production facilities. Conversations with the engineering and editorial staff will bring to light many points learned from design and construction of numerous suites over several generations of changing technology.

Although the equipment level may exceed the broadcasters needs, the pressure of the commercial post production environment quickly exposes flaws in system lay-out or functional design. This information, combined with the experience already available on broadcast facilities design, can produce a well integrated and functionally efficient suite.

The following list is an abbreviated collection of topics to be examined during suite physical design.

Suite Design Physical Considerations

Traffic Patterns

- Client Access
- Operator Access
- Maintenance Access
- Observer

Technical Environment

- Temperature Control
- Humidity Control
- Power Quality
- Radio Frequency Interference Susceptibility
- Static Discharge
- Contamination Control

Ergonomics

- Operator Console
- Storage Facilities
- Seating Positions
- Client/Producer Work Space
- View Path
- Acoustic Path

Visual Environment

- Operating Lighting
- Service Lighting

Acoustical Environment

- Frequency Response
- Noise Containment
- Stereo Imaging

Traffic Patterns

Client Access—Ability to enter and exit suite with minimal disruption to operator

Operator Access—Operator can access secondary hardware located out of immediate reach without disturbing client

Maintenance Access—In case of failure, maintenance technician can patch in auxiliary equipment without disturbing client

Routine maintenance to unassigned equipment can be performed without disturbing operators or client

Observer Access—Provides enough observation access to tours to indicate the application of the facility while avoiding disruption of sessions

Ergonomics

Operator Console—Allows for the operator to reach primary controls without excessive motion, while monitoring their effect.

Placement of keyboard, switcher control panels, audio consoles, remote controls, etc. so that typical operation doesn't place two simultaneously needed controls out of reach.

Height, width and depth of cabinetry chosen to allow comfortable, prolonged operation.

Client/Producer Workspace—Provision for a dedicated location for comfortable workspace allowing the client or producer to work with the operator without being oppressively close. Position should provide writing space and include lighting that does not interfere with program monitors.

View Path—Placement of equipment and console heights to allow as parallax-free view of the program monitors as possible for both operators and clients by minimizing the angle away from in-line view that will occur as operator moves between positions. Accomplished by placing primary controls directly in front of or adjacent to an operator without using a horseshoe approach that requires excessive neck twisting.

Acoustic Path—Provision for a direct acoustic path for the desired sources such as the monitoring system while interrupting the path of noise sources such as equipment fans. Operator and client must be able to be located in this path at their normal locations.

Technical Environment

Temperature Control—Provision for sufficient cooling of equipment without freezing clients. Operating consoles often allow too little ventilation for the current generation of high density electronics.

Humidity Control—Inclusion of humidity controls such as a humidifier to suppress the generation of static electricity by personnel and hardware or, conversely, dehumidifiers to reduce excessive type adhesion produced by high humidities.

Power Quality—Editing suite power sources should receive the same general consideration as those in a computer facility. The likelihood of losing an edit session due to an edit computer program crash caused by a power line spike can be reduced by utilizing any of the many line filters now commonly available for the larger personal computer systems.

Radio Frequency Interference Susceptibility—The equipment manufactured for an editing suite may not have been specifically designed for operation in a RF field. A typical commercial post production suite often does not have a broadcast transmitter, microwave transmitter, weather radar or numerous two-way radio systems operating nearby. Placing an editing suite in such a location may produce difficulties not normally experienced with similar broadcast equipment.

Static Discharge Control—Provision for a safe discharge path for the static charge that builds up on personnel utilizing the editing suite. Humidification, conductive floor treatments, and anti-static sprays are all good steps in reducing the likelihood of system crash due to static discharge. Combining these measures with operator static awareness training can significantly improve system reliability.

Contamination Control—In an editing suite with remotely located VTRs, the contamination precautions are the same as those in a studio control room with the added caution to avoid contaminating floppy disks used for edit list storage. Maintaining the contamination control level for a suite with on-site VTRs requires the same or stricter precautions as those used in a standard operating tape room. If dropouts are considered unacceptable so must be their human-generated causes. The multiple passes endured by the tape during computerized editing increase susceptibility to tape damage from any cause.

Visual Environment

Operations Lighting—Proper lighting conditions during an editing session will provide minimal glare on monitors and console surfaces, clear readability of controls, work light for scripts and walk away lighting for safety. Attention to color temperatures produced by lighting fixtures and monitors will minimize contamination. Room color treatment should be neutral and minimally non-reflective.

Service Lighting—Include an auxiliary set of lighting fixtures or a higher operating level for the existing operations' lights that provides sufficient illumination for equipment or general maintenance. The method of providing service lighting should not disturb the ability to quickly reproduce operational lighting conditions.

Acoustical Environment

Frequency Response—Care must be taken to minimize control room conditions that produce frequency response nonlinearities. Many edit points are selected relative to a specific aural cue. Loss or exaggeration of these cues may cause judgement errors in selecting appropriate edit points. These conditions may also interfere in performing objective quality control tests.

Stereo Imaging—Placement of monitor speakers, room obstructions and personnel must all be considered in order to provide an environment that allows the mixdown process to correctly place the stereo image. Frequency response and reverberation time have to be controlled in order to avoid distorting the monitored stereo image. It is advisable to draw upon the experience of persons who have successfully dealt with these issues in dedicated mixdown facilities.

As is often the case in broadcast facilities many commercial post production suites are built with a separate equipment room for VTR, digital effects and terminal equipment. This technique offers the advantage of keeping the noise and heat generated by such equipment away from the operating environment. In case of equipment failure, patching over to alternate equipment and service occurs with minimal alarm to the client.

Alternatives to the separate machine and control room approach become practical as VTR and terminal equipment size are reduced. In Fig. 30 the reduction in size and power provided by interformat editing allows all equipment to be placed in the suite. This centralization allows fast and efficient operation by single operator.

Systems Integration

Integration of the editing suite electronic components into the broadcast facility in order to utilize other plant facilities often involves consideration of the following systems.

External Devices Accessed By Edit Suite

Video
- Routing Switcher and Distribution System
- Digital Effects Equipment
- Character Generator
- Graphics Camera and Art System
- Still Store System
- Digital Noise Reducer
- Color Correction
- Test Signals

Audio
- Routing Switcher and Distribution
- Announce Booth
- Audio Sweetening Room
- Test Signals
- Intercom System

The integration of these systems may require provision of remote control panels, delegation switching, delay compensation, distribution equalization and level compensation. Judging when it is practical to time share one piece of equipment or to purchase a smaller dedicated device must take into account the inevitable priority conflicts that will occur between the editing group and on-air operation

This problem is rarely permanently resolved due to the continual increase in the power of technology that can be purchased per dollar.

A typical situation is that the falling price of technology allows a character generator (CG) to be purchased for dedicated use by an edit suite. This solves the conflict over (CG) time but the same advance in technology that made the (CG) more economical also brought the price of an electronic graphics and paint system down to a level where the facility can afford to purchase one. This has been the case for digital effects, still store and noise reduction systems.

SMPTE RECOMMENDED PRACTICE RP 113–1983

Supervisory Protocol for Digital Control Interface

1. *General*

1.1 Scope. This recommended practice defines the supervisory protocol used within a general purpose communication channel of an interface system which transports data and digital control signals between equipment utilized in the production, post-production, and/or transmission of visual and aural information. It is intended that the supervisory protocol described in this practice be part of an overall system, allowing interconnection of programmable and nonprogrammable equipment as required to configure an operational system with a defined function, and to allow rapid reconfiguration of a system to provide more than one defined function utilizing a given group of equipment.

1.1.1 The primary intent of this practice is to establish supervisory procedures of the communication channel for the purpose of transmitting control messages to equipment by external means. (The contents of the messages are not defined.) This practice, or sections thereof, may be applied to the interconnection of elements within an item of equipment.

1.2 Definitions. (See Fig. 1.) For the purposes of this practice, the following definitions apply:

Bus Controller: Each system contains one bus controller which supervises all tributaries in the system. Supervision is exercised through the use of this supervisory protocol.

Byte: A byte consists of eight bits of information. Bits used to effect transmission such as byte start, parity, or end are not part of the byte.

Tributary: A tributary transfers messages to and from an operational device via the interface system. The tributary is distinct from the function of the operational device and exists to transfer control messages between the communication channel and the device.

Word: A word consists of a byte and associated bits used to effect transmission such as start, parity, or end.

2. *Message Types*

Two types of messages shall be transmitted on the channel:

Supervisory Messages to supervise the channel and direct the flow of device messages.

Device Messages to control operation of equipment functions. This type of message shall be transmitted only within standard message blocks or during device defined communications modes. Details of device messages will be the subject of another document now in preparation.

3. *Tributary Addresses*

Tributary addresses shall consist of two bytes: the most significant byte, which is transmitted first, and the least significant byte. The most significant bit of both bytes shall be set to binary 1. This provides an address range starting at 8080_h. Each tributary shall be assigned two unique addresses, a SELECT address and a POLL address.

3.1 Select Address. An address in which the least significant bit of the least significant byte equals binary 0 is a SELECT address.

3.2 Poll Address. An address in which the least significant bit of the least significant byte equals binary 1 is a POLL address.

3.3 Group Addresses. Address pairs $8080\text{-}8081_h$ and $8082\text{-}8083_h$ through $81FE\text{-}81FF_h$ are reserved as GROUP SELECT addresses. The addresses in which the least significant bit of the least significant byte equals binary 1 (POLL address) shall not be used but are retained in the address numbering scheme for software considerations.

3.4 Discrete Addresses. Tributary addresses shall start at $8280\text{-}8281_h$. Precisely 8064 discrete tributary address pairs are available.

3.5 Address Allocation Table:

$8080\text{-}8081_h$	Group select—all call
$8082\text{-}80FF_h$	Group select—Groups 1-63
$8180\text{-}81FF_h$	Group select—Groups 64-127
$8280\text{-}82FF_h$	64 tributaries
$8380\text{-}83FF_h$	64 tributaries
⋮	
$FF80\text{-}FFFF_h$	64 tributaries

Approved 3 February 1983

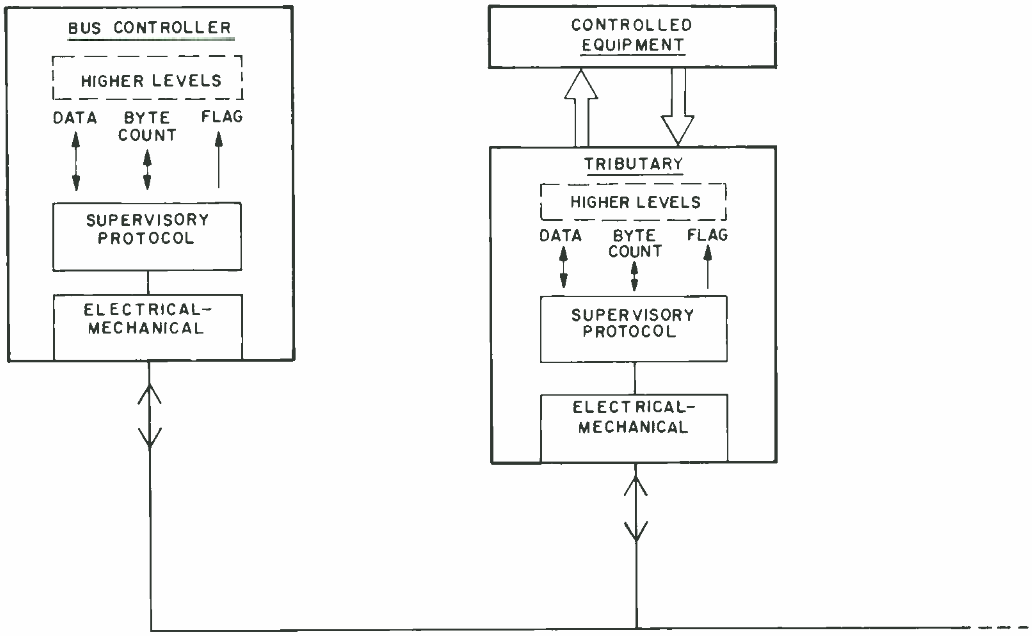


Fig. 1
System Elements

4. *Tributary Operational States*

A tributary shall be in one of five major operational states:

IDLE: The tributary shall not perform any communications. This state shall be exited only in response to BREAK.

ACTIVE: Prerequisite for transition to other operational states. The tributary shall enter this state whenever BREAK is received.

POLL: The tributary shall transmit a single status byte to the bus controller.

SELECT: A single tributary shall enter a communications mode with the bus controller.

GROUP SELECT: All tributaries or a selected group of tributaries shall enter a communications mode with the bus controller.

5. *Supervisory Messages*

Tributaries shall be directed to operational states through various communications sequences by supervisory messages as shown in Fig. 2. Supervisory messages consist of the following elements:

BREAK: Shall drive all tributaries to the ACTIVE state. (See American National Standard for Television—Digital Control Interface—Electrical and Mechanical Characteristics, ANSI/SMPTE 207M-1984, for code.)

(ADDR-POLL) : A tributary poll address (ADDR-POLL) shall drive the addressed tributary to the POLL state.

(ADDR-SELECT) : A tributary select address (ADDR-SEL) shall drive the addressed tributary to the SELECT state.

(GROUP ADDR-SEL) : A group select address shall drive a group of tributaries to the GROUP SELECT state.

Supervisory Characters: Identify communications sequences and provide status information. Supervisory characters are single bytes within the range 00_h-$7F_h$. Supervisory characters shall consist of:

01_h	(GRP)	Group assign
02	(STX)	Start of message
03	(ESC)	Escape
04	(ACK)	Acknowledge
05	(NAK)	Not acknowledge
06	(BSY)	Busy
07	(RST)	Reset
08	(SVC)	Service request from controlled equipment
09	(TEN)	Transmit enable

All other supervisory characters are reserved. The use of other characters for tributary supervision is noncompliant with this specification.

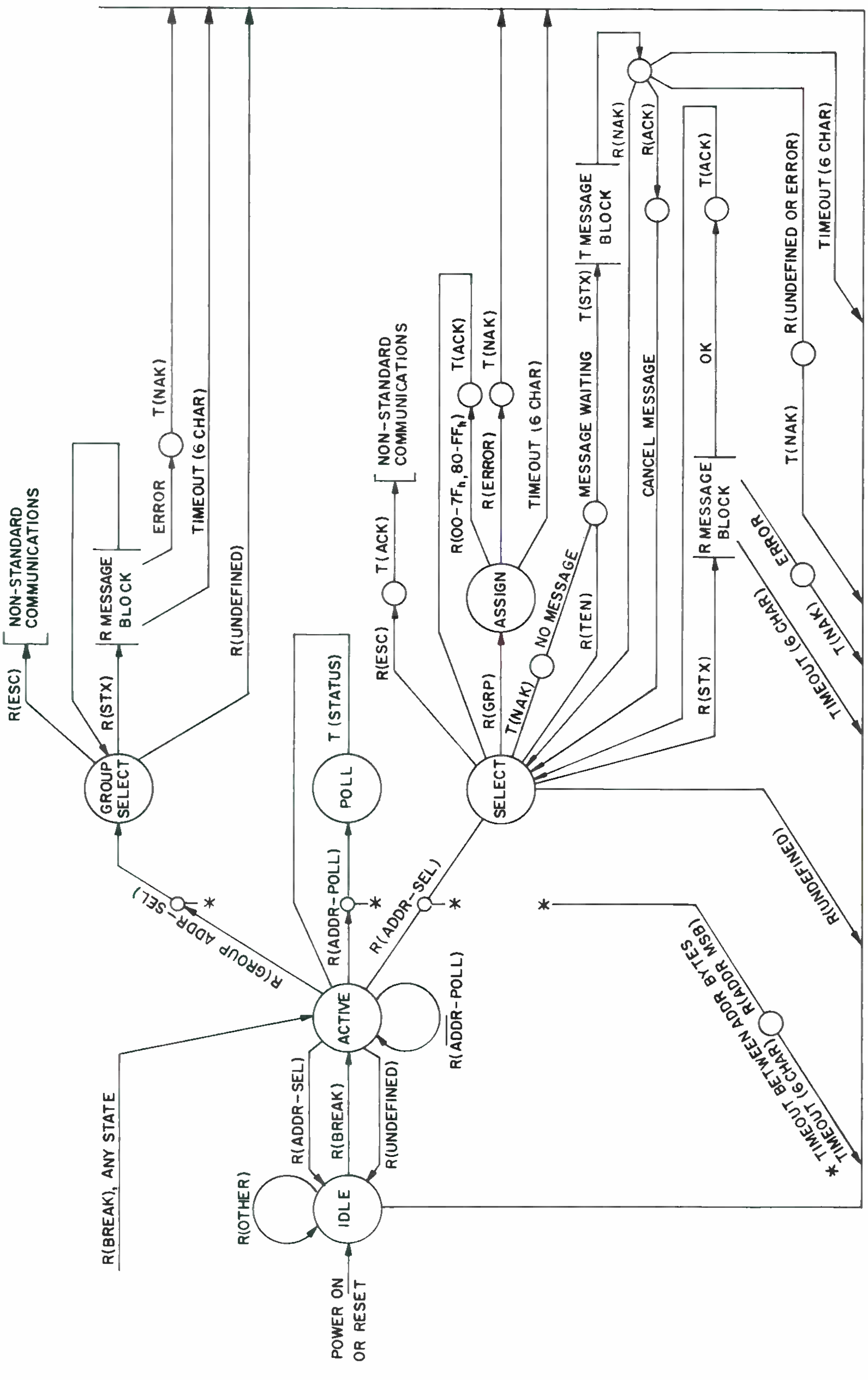


Fig. 2
Tributary Supervisory Protocol

6. *Operational Sequences* (See Fig. 2)

6.1 Idle. Tributaries in the idle state shall not perform any communications sequences.

6.1.1 Idle shall be entered under the following conditions:

Tributary power on or reset

Upon receipt of an (ADDR-SEL) not assigned to the tributary

Upon receipt of a GROUP (ADDR-SEL) not assigned to the tributary

When a specified time out of six words in duration occurs

On receipt of an undefined byte

On encountering a transmission error or ambiguous condition

6.1.2 A tributary shall exit the IDLE state only on receipt of BREAK.

6.2 Active. A tributary in the ACTIVE state shall perform communications sequences as directed by the bus controller.

6.2.1 All tributaries shall enter the ACTIVE state whenever BREAK is received and on completion of a poll sequence.

6.2.2 A tributary shall exit the ACTIVE state on receipt of an address which directs it to the POLL. SELECT, or GROUP SELECT states. A time out between the two address bytes shall cause the tributary to enter the IDLE state.

6.3 Poll. The POLL state shall be used to determine the presence and status of a tributary.

6.3.1 A tributary shall enter the POLL state on receipt of its poll address (ADDR-POLL). The tributary shall transmit one supervisory character to indicate its status, then return to the ACTIVE state. Tributaries not addressed shall remain in the ACTIVE state. Supervisory characters transmitted shall be of the following:

RESET (07_h): Tributary has powered up or been reset since last poll.

NAK (05_h): An exception (time out, undefined byte, etc.) condition has occurred since last poll or select.

BSY (06_h): Tributary not available to receive messages.

SVC (08_h): Service request from controlled equipment.

ACK (04_h): Tributary available to receive messages.

These characters rank in priority according to the order shown above.

6.4 Select

6.4.1 A single tributary shall enter the SELECT state on receipt of its select address (ADDR-SEL). All other tributaries shall transition to the IDLE state. A tributary in the SELECT state shall execute the communications sequences detailed in 6.4.1.1 through 6.4.1.4 as directed by the bus controller.

6.4.1.1 Receive Message. Supervisory character STX (02_h) shall be followed by a message block containing the following characters:

byte 1: byte count of bytes 2 thru n (0 = 256 bytes).

bytes 2 thru n: (256 bytes maximum) —device defined message.

byte n + 1: checksum = 2's complement of the least significant byte of the sum of bytes 1 through n.

The tributary shall indicate error-free reception by responding with ACK and shall return to the SELECT state.

On encountering an error during reception, the tributary shall respond with NAK, then transition to IDLE.

If transmission from the bus controller is interrupted for more than the time out period, the tributary shall transition to IDLE.

The tributary shall transfer a complete message, the byte count specified above, and a "block ready" indication to the entities using the system for control.

6.4.1.2 Transmit Message. A tributary shall notify the bus controller that a message is waiting by transmitting SVC (08_h) during POLL. Upon receipt of TEN (09_h) while in SELECT, the tributary shall transmit a standard message block as defined in 6.4.1.1.

6.4.1.3 Assign Tributary to Group. Supervisory character GRP (01_h) shall be followed by a single byte:

00_h deletes all previous group assignments of all tributaries

If the most significant bit of the byte following GRP (01_h) is a ZERO, the assignment of the tributary to the address represented by the following seven bits of that byte is deleted (1_h to 127_h).

80_h assigns the tributary to groups 1-127.

If the most significant bit of the byte following GRP (01_h) is a ONE, the tributary is assigned to the address represented by the following seven bits of that byte (1_h to 127_h).

A tributary may be removed from or assigned to more than one group by repeating the assignment sequence.

If transmission from the bus controller is interrupted for more than the time out period between receipt of GRP and the group assignment byte, the tributary shall transition to IDLE.

All group addresses except "all call" (8080_h) shall be deleted at tributary power-up or reset.

6.4.1.4 Nonstandard Communications. Supervisory character ESC (03_h) shall release a tributary to nonstandard communications sequences. The tributary shall respond with ACK; it shall exit ESC mode only in response to BREAK.

6.4.2 The tributary shall exit SELECT on receipt of BREAK or in response to the exceptional conditions noted in 6.4.1.1 through 6.4.1.3 above.

6.5 Group Select

6.5.1 Groups of tributaries shall enter the GROUP SELECT state on receipt of their group select address (GROUP ADDR-SEL). All tributaries not assigned to the group shall transition to IDLE. Tributaries in the GROUP SELECT state shall execute the communications sequences detailed in 6.5.1.1 and 6.5.1.2 as directed by the bus controller.

6.5.1.1 Receive Message. Supervisory character STX shall be followed by a message block as defined in 6.4.1.1.

Each tributary returns to GROUP SELECT state after error-free reception of the block; no response shall be transmitted.

On encountering an error during reception, a tributary shall respond with NAK, then transition to IDLE.

If transmission from the bus controller is interrupted for more than the time out period, tributaries shall transition to IDLE.

6.5.1.2 Nonstandard Communications. Supervisory character ESC shall release a group to nonstandard communications in accordance with 6.4.1.4. Tributaries shall exit this mode only in response to BREAK.

6.5.2 Tributaries shall exit GROUP SELECT on receipt of BREAK or in response to the exceptional conditions noted in 6.5.1.1.

7. *Bus Controller Operation*

7.1 System Synchronization. The bus controller shall transmit BREAK when power is turned on and after being reset.

7.2 Tributary Response Time Out. The bus controller shall transmit BREAK when a tributary fails to respond within the following time-out periods:

In response to ADDR-POLL, GRP (#), ESC, TEN, END OF MSG BLOCK 6 words

8. *Guidelines*

8.1 Function of This Practice. This practice specifies the supervisory protocol used within the communication channel. The protocol is the sequence of characters used to transfer messages between the bus controller and tributaries, provide recovery from error conditions, and generally supervise the usage of the communication channel. This practice is concerned only with channel supervision. Electrical/mechanical characteristics are specified in separate standards since many types of channels which can deliver eight-bit binary bytes and a unique BREAK condition can operate under supervision of the protocol Message content is specified by standards which are independent of both electrical/mechanical and supervisory characteristics of the communication channel.

8.2 System Configurations. This supervisory protocol permits supervision of point-to-point and multipoint systems.

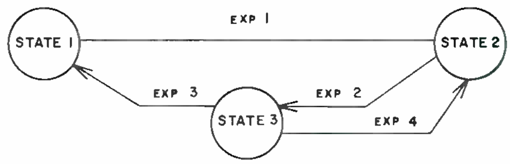
A point-to-point configuration is one in which a communication channel is connected to only one tributary. The bus controller may be connected to more than one channel, each having one tributary. This configuration has the advantage of speed since the dedicated channels provide access to all tributaries simultaneously.

The multipoint bus configuration is one in which more than one tributary is connected to a channel. This configuration has the advantage of reduced cabling costs and complexity. The main disadvantage of multipoint is that messages to different tributaries must queue up and be sent serially on the bus. This configuration is therefore slower in response time than point-to-point systems.

8.3 State Diagrams. The supervisory procedures are described by means of state diagrams that show how the interfacing hardware and software in a tributary follow sequences of bytes as they are received from the communication channel.

8.3.1 Each state (condition) that a tributary can assume is represented graphically as a circle; major states are identified by an upper case label or mnemonic within the circle:

8.3.2 All possible transitions between states are represented by arrows between the states; each transition is qualified by an expression which will produce the transition.



8.3.3 Expressions can be messages received from or transmitted to the communication channel, or local messages generated within the tributary.

8.3.3.1 Messages received from the channel are represented by R followed by the received message in parentheses:

indicates transition from state ONE to state TWO on receipt of the message GRP.

8.3.3.2 Messages transmitted to the bus are represented by T followed by the transmitted message in parentheses:

indicates transition from state THREE to state FOUR after transmitting the message ACK.

8.3.3.3 Local messages are represented by lower case labels:

indicates transition from state FIVE to state SIX when reset occurs.

8.4 Channel Synchronization. Data density is maximized by allowing the transmission of binary data in all device messages. This means that there must be no combination of transmitted bytes which can be interpreted as a channel synchronization command. The channel synchronization command is a unique transmission sequence called BREAK. This sequence cannot be accidentally generated by normal communications. Tributaries receiving BREAK are required to immediately transfer to the ACTIVE state regardless of what they are currently doing in relation to the communication channel. On power up, a tributary enters the IDLE state and ignores all bus transactions until it receives BREAK. Electrical specifications appropriate for use with this supervisory protocol assure that BREAK cannot be generated accidentally.

8.5 Supervisory Message Components. The protocol uses BREAK, tributary addresses, and a small number of predefined supervisory characters to manage the communication channel. Since the addresses and supervisory characters are eight-bit binary bytes, they must be recognized by being received immediately after BREAK. The only supervisory message that is unconditionally recognizable is the BREAK sequence.

8.5.1 Tributary Addresses. Tributary addresses consist of two bytes. Up to 8064 tributaries can be addressed uniquely. A one-byte addressing scheme would have served most small system applications with a saving in channel overhead, but complex reassignment strategies would have to be employed in order to accommodate larger users.

The address bytes are characterized by a 1 in the most significant bit. Each tributary is assigned two addresses, a SELECT address and a POLL address. The least significant bit of the least significant byte is set to 0 for SELECT and 1 for POLL.

A unique two-byte address serves as an all-call SELECT address. When this address is transmitted all tributaries in a multipoint system simultaneously receive and act on system messages.

Tributaries can be assigned to one or more of 127 group SELECT addresses. These addresses allow simultaneous operation with selected groups of tributaries in multipoint systems similar to all-call.

During all-call or group operation, transmission by the tributaries is allowed only when an error condition is encountered, since other transmission could cause channel errors as several tributaries attempted to transmit at the same time. When error conditions are encountered, tributaries transmit the supervisory character NAK; reception of the NAK, or an error indicating channel contention, alerts the bus controller to an error condition in one or more tributaries. The bus controller must assert BREAK and poll individual tributaries to determine which tributary (ies) has encountered an error and the nature of the error.

8.5.2 Supervisory Characters. The only supervisory characters used are those given in Sec. 5. Supervisory characters are single eight-bit bytes in which the most significant bit is 0. Implementations of this protocol must not use any other supervisory characters for nonspecified functions as such use would render a tributary incompatible with other systems and could occasion serious operational failures if other supervisory functions are added to this practice in the future.

8.6 Poll Sequence. The POLL sequence is used to verify tributary presence and status. In multipoint systems, the POLL sequence allows all tributaries to be scanned quickly to see if servicing or attention is required by any of them.

Status characters transmitted by a tributary inform the bus controller of the tributary's current condition. Characters associated with specific conditions are detailed in 6.3.1. The tributary is required to send the highest priority status character applicable to its condition if more than one applies. All status characters except service request (SVC) apply to conditions within the interface function. SVC is a pass-through condition which indicates a service need by the equipment controlled through the interface. Device messages are used to identify and provide the service required.

8.7 Message Receive or Transmit Sequences. Device messages are received or transmitted by a tributary by means of the message receive or transmit sequence from the SELECT state. This sequence offers message lengths of 1 to 256 bytes with checksum protection. Groups of tributaries can receive messages from the GROUP SELECT state.

On receipt of a message, the bus controller will transmit an ACK or NAK. It then waits for six characters for any exceptional condition. (See Fig. 2.)

All equipment control and status information is exchanged by means of device messages.

8.8 Escape Sequence. The escape sequence is provided for those users who wish to remain compatible with the electrical and supervisory protocol characteristics of the interface system but require nonstandard operational sequences or messages. Single tributaries or groups of tributaries may be placed outside the normal protocol limits using this sequence. The only protocol requirement which must be observed by devices while using this sequence is the requirement to enter the ACTIVE state whenever a BREAK is received from the communication channel.

8.9 System Design Considerations. This practice and associated standards specify characteristics for equipment compatible with the interface system. System function and configuration is left to the system designer. Certain cautions must be observed by the designer:

8.9.1 Device messages are specified by other standards. Only device messages which conform to those standards should be transmitted via the standard message receive/transmit facilities. Nonstandard messages should be transmitted via the escape sequence.

8.9.2 Switched Tributaries. This practice and associated standards consider operation of bus controllers and tributaries to be within one communication channel. If tributaries are transferred between channels, the system designer must provide means to place them in an appropriate state before connection to a new channel. It is recommended that the tributaries be forced to the IDLE state with all group address assignments cleared before connection. Procedures for notifying a bus controller of the attachment of a tributary will generally be required; these procedures are dependent on the nature of the system and are left to the designer's discretion.

American National Standard for television— digital control interface— electrical and mechanical characteristics

Approved January 27, 1984 Secretariat: Society of Motion Picture and Television Engineers

1. General

1.1 Scope. This standard defines the electrical and mechanical characteristics of an interface system comprised of a general purpose communication channel and interface device(s) used for the transfer of data and digital control signals between equipment utilized in the production, post-production, and/or transmission of visual and aural information. It is intended that the communication channel and device(s) described in this standard be part of an overall equipment interface, allowing interconnection of programmable and nonprogrammable control and accessory equipment as required to configure an operational system with a defined function. The standard is also intended to allow rapid reconfiguration of a system providing more than one defined function utilizing a given group of equipment.

1.1.1 The electrical and mechanical specifications set forth in this standard are intended for use in both fixed plant and field operational environments. These specifications take into account the requirement to function reliably without causing undue interference with other signals normally found in these environments.

1.1.2 This standard defines the electrical and mechanical characteristics of the communication channel and the associated interface device(s), to the exclusion of design specifications, performance requirements, safety requirements, and the communications protocol used in or by such equipment.

1.1.3 The primary intent of this standard is to establish an electrical and mechanical interface and communication channel for the purpose of interconnecting equipment by external means. This standard, or sections thereof, may be applied to the interconnection of elements within an item of equipment.

1.2 Definitions. For the purposes of this standard, the following definitions shall apply:

Equipment means either a single device which connects to the interface system or a group of interconnected devices, providing a specified operational function, having one common connection to the interface system.

Interface Bus refers to the communication channel.

1.3 Object. The intent of this standard is to:

Define a general-purpose interface system for use in the environment specified in 1.1

Specify equipment-independent electrial, mechanical, and functional interface characteristics which permit equipment to connect and communicate unambiguously via the interface system

Specify terminology and definitions related to the electrical and mechanical portion of the interface system

Enable the interconnection of independently manufactured equipment into a single functional system

Permit equipment with a wide range of operational capabilities to be connected to the interface system simultaneously

American National Standards Institute, 1430 Broadway, New York, N.Y. 10018

Define a system which is user configurable

Define a system based on readily obtainable standard components

1.4 Interface System Overview

1.4.1 This standard applies to systems, or portions of systems, which have the following characteristics:

A full-duplex four-wire communications channel is utilized

A nominal maximum bus length of 1220 m (4003 ft)

Data is transmitted asynchronously, bit serial, word serial

Standard transmission rate on the interface bus is 38.4 kilobits per second (kb/s)

Data exchange between devices is digital (as distinct from analog)

1.4.2 The function of the interface system is to provide an effective communications link over which messages are carried in an unambiguous way among a group of interconnected devices.

1.4.3 The interface system described in this standard assigns one of two operational characteristics to all devices:

Bus Controller. Each interface system contains one bus controller which supervises all tributaries in the system. This supervision is exercised through the use of interface protocol. The bus controller may also perform one or more functions in the operational plant in addition to its interface supervision. Although only one bus controller may be part of any particular interface system, it is recognized that an operational plant may make use of more than one interface system.

Tributary. A tributary transfers messages to and from an operational device via the interface system as specified in the interface system protocol. A tributary communicates messages through the interface bus only via the bus controller.

1.4.4 The basic message paths and the bus structure shall be as follows:

The basic message path utilizes asynchronous, bit serial/word serial transmission via a balanced wire pair

The interface bus may be utilized in either point-to-point or multi-point configuration including but not limited to

A point-to-point bus connecting one tributary to a bus controller

A multipoint bus connecting multiple tributaries to a single bus controller

The interface bus is a four-wire configuration which will effect two-way communication using a separate wire pair for each transmission direction; communication between tributaries is accomplished through the bus controller.

1.4.5 The data word and BREAK character utilized by the interface system shall be as follows:

The standard serial data word includes an eight-bit data byte; the complete serial data word consists of one start bit (SPACE), eight data bits (ONE BYTE), a parity bit (EVEN), and one stop bit (MARK). The least significant bit is transmitted first.

A BREAK character, comprised of 20 bits SPACE followed by a minimum of 2 bits MARK, is utilized to synchronize all devices connected to the interface bus.

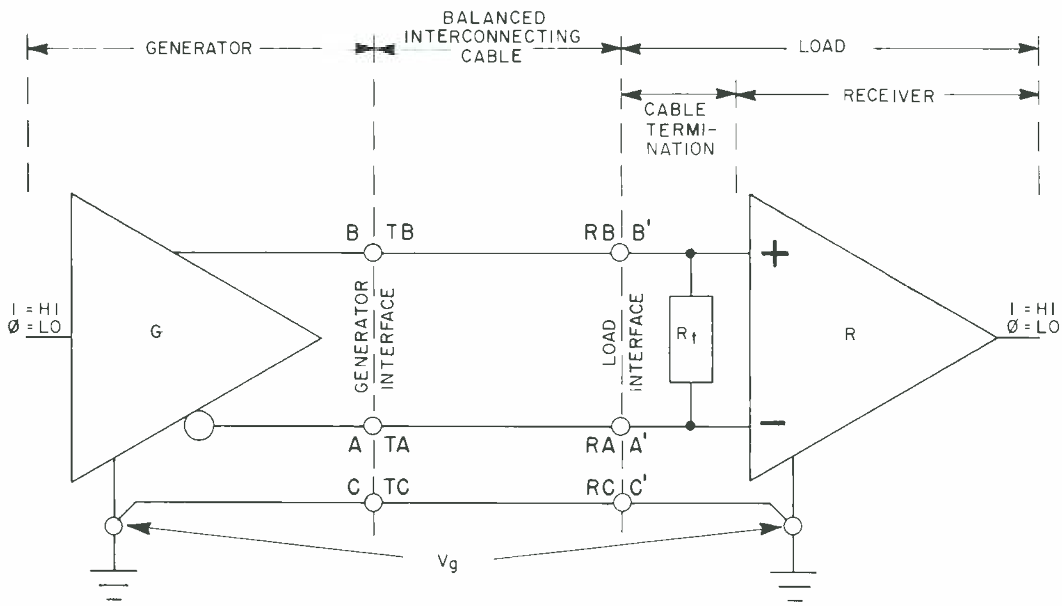
2. Electrical Characteristics

2.1 Interface Circuit. The balanced voltage digital interface circuit is shown in Fig. 1. The circuit consists of three parts: the generator, the balanced interconnecting cable, and the load. The load may consist of one or more receivers (R) and an optional cable termination resistance (R_t). The electrical characteristics of the generator and receiver are specified in terms of direct electrical measurements while the interconnecting cable is specified in terms of its electrical and physical characteristics.

2.2 Generator Characteristics. The electrical characteristics of the generator are specified in accordance with measurements described in 2.2.1 through 2.2.6 and illustrated in Figs. 2 and 3. A generator circuit meeting these requirements results in a low impedance (100 ohms or less) balanced voltage source producing a differential voltage applied to the interconnecting cable in the range of 2 to 6 volts. The signalling sense of the voltages appearing across the interconnecting cable are defined as follows:

The B terminal of the generator shall be positive with respect to the A terminal for a binary 1 (MARK) state.

The B terminal of the generator shall be negative with respect to the A terminal for a binary 0 (SPACE) state.



LEGEND

R_t	=	CABLE TERMINATION RESISTANCE
Vg	=	GROUND POTENTIAL DIFFERENCE
A, B	=	GENERATOR INTERFACE POINTS
A', B'	=	LOAD INTERFACE POINTS
C	=	GENERATOR CIRCUIT GROUND
C'	=	LOAD CIRCUIT GROUND

Fig. 1
Balanced Digital Interface Circuit

2.2.1 Open circuit measurement for either binary state shall be made in accordance with Fig. 2a. The results of this measurement shall be as follows:

The magnitude of the differential voltage (V_o) measured between the two generator output terminals shall not be more than 6.0 volts.

The magnitude of the voltage between either of the generator output terminals and generator ground (V_{oa} and V_{ob}) shall not be more than 6.0 volts.

2.2.2 The test termination measurement shall be made with a test load of two resistors, 50 ohm ± 1%, connected in series between the generator output terminals as shown in Fig. 2b. The results of this measurement shall be as follows:

The magnitude of the differential voltage (V_t) measured between the two output terminals shall not be less than either 2.0 volts or 50% of the magnitude of V_o, whichever is greater. For the opposite binary state the polarity of $\overline{V_t}$ shall be reversed (V_t).

The magnitude of the difference in magnitudes of V_t and $\bar{V}_t$ shall be less than 0.4 volts.

The magnitude of the generator offset voltage, V_{os}, measured between the center point of the test load and generator circuit shall not be greater than 3.0 volts.

The magnitude of the difference in the magnitudes of V_{os} for one binary state and $\bar{V}_{os}$ for the opposite binary state shall be less than 0.4 volts.

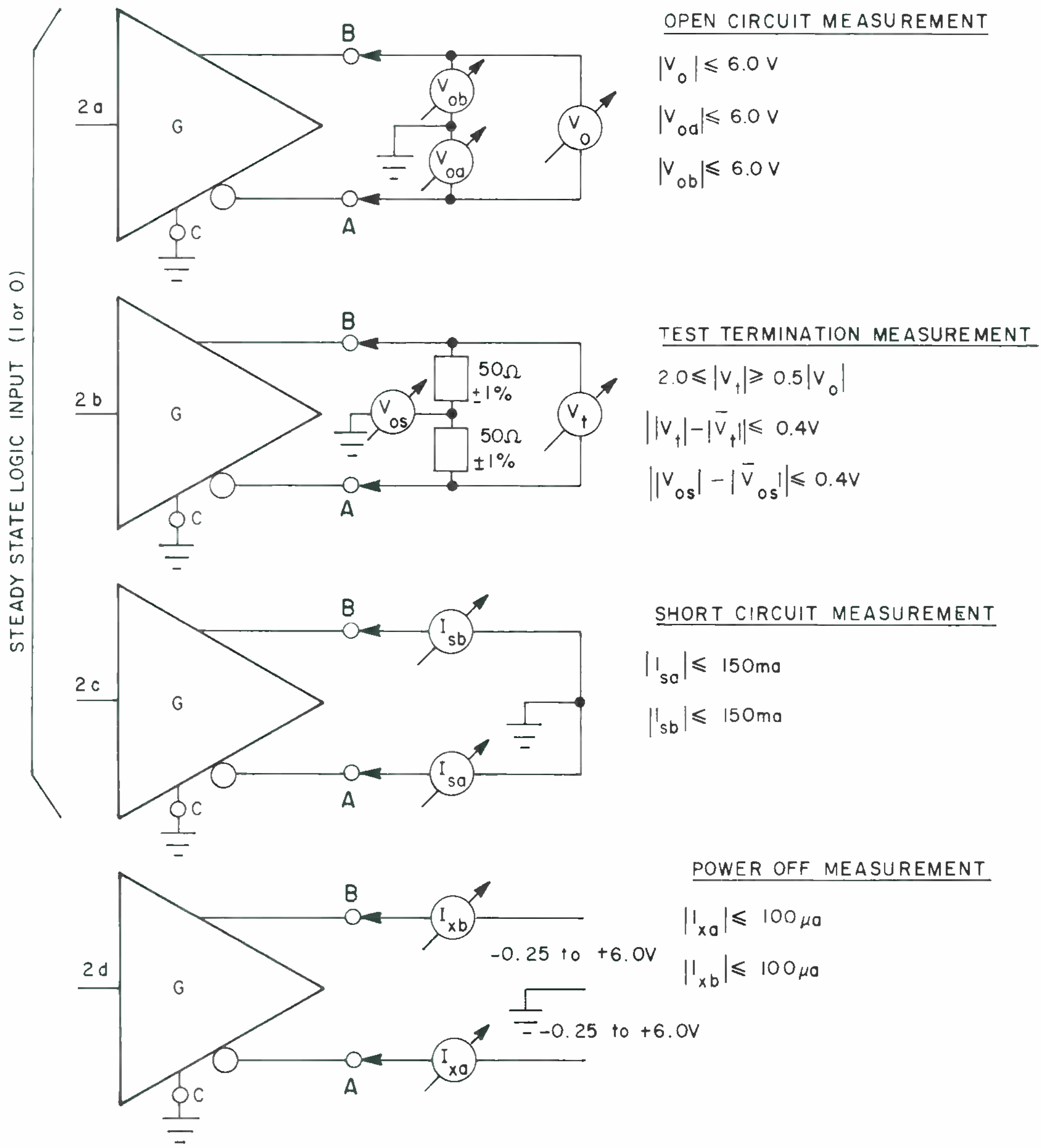


Fig. 2
Generator Parameter Measurement

2.2.3 The short circuit measurement shall be made with the generator output terminals short-circuited to generator circuit ground as illustrated in Fig. 2c. The magnitudes of the currents flowing through each generator output terminal during this test shall not exceed 150 milliamperes for either binary state.

2.2.4 The power-off measurement shall be made under power-off conditions and as illustrated in Fig. 2d. The magnitude of the generator output leakage currents (I_{xa} and I_{xb}), with voltages ranging between +6.0 and —0.25 volts applied between each output terminal and generator circuit ground, shall not exceed 100 microamperes.

2.2.5 The output signal waveform measurement shall be made using a test load consisting of a noninductive resistor with a value of 100 ohms ± 10% connected between the generator output terminals, as illustrated in Fig. 3. During transitions of the generator output between alternating

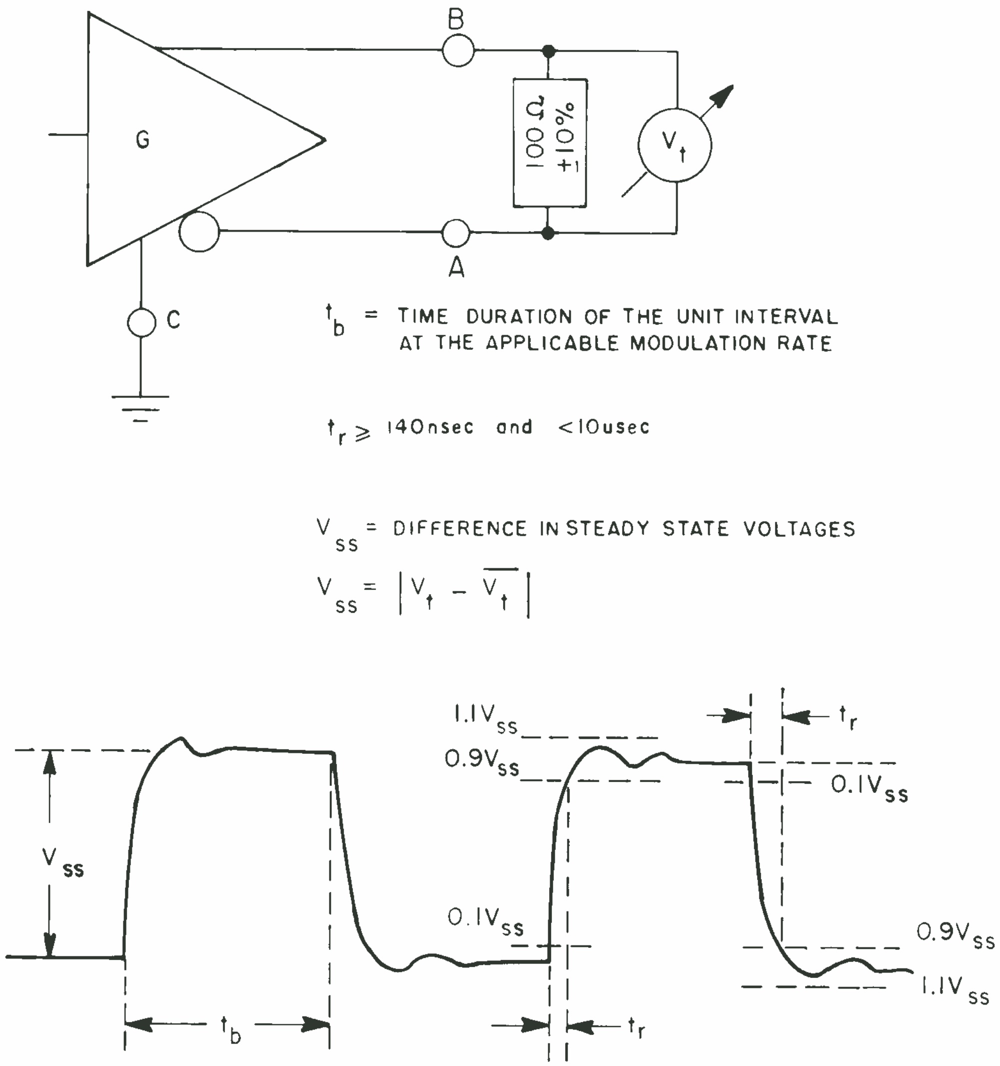


Fig. 3
Generator Output Signal Waveform

binary states (one-zero-one-zero, etc.), the differential signal measured across the test load shall be such that the voltage monotonically changes between 10% and 90% of V_{ss} at not less than 140 nanoseconds. Thereafter, the signal voltage shall not vary more than 10% of V_{ss} from the steady state value, until the next binary transition occurs. At no time shall the instantaneous magnitude of V_t or V_t exceed 6.0 volts nor be less than 2.0 volts. V_{ss} is defined as the voltage difference between the steady state values of the generator output.

2.2.6 The generator output shall be capable of being placed in a high-impedance state and when in such state shall withstand a common mode voltage swing of up to 7 volts.

2.3 Load Characteristics. The load consists of one or more receivers (R) and an optional cable termination resistance (R_t) as shown in Fig. 1. The electrical characteristics of a single receiver excluding both cable termination and fail-safe provision are specified in terms of the measurements described in 2.3.1 through 2.3.7 and illustrated in Figs. 4 through 6. A circuit meeting these requirements results in a differential receiver having a high-input impedance (> 4 kilohms), a small input threshold transition region between —0.5 and +0.5 volts and allowance for an internal bias voltage not to exceed 3 volts in magnitude.

2.3.1 The input current/voltage measurements shall be made with the voltage V_{ia} (or V_{ib}) ranging between —10.0 and +10.0 volts, while V_{ib} (or V_{ia}) is held at 0.0 volts (ground). This measurement shall be made with the power supply to the receiver in both the power-on and power-off condition. The resultant input current I_{ia} (or I_{ib}) shall remain within the shaded region shown in Fig. 4.

2.3.2 The input sensitivity measurement shall be made as illustrated in Fig. 5 over the entire common mode voltage (V_{cm}) range of —15 to +15 volts. The receiver shall not require a differential input voltage of more than 500 millivolts to correctly assume the intended binary state. Reversing the polarity of V_i shall cause the receiver to assume the opposite binary state. The receiver is required to maintain correct operation for differential input signal voltages ranging between 500 millivolts and 6 volts in magnitude. The maximum voltage (signal plus common mode) present between either receiver input terminal and receiver circuit ground shall not exceed 25 volts in magnitude. Application of voltages less than the maximum voltage (signal plus common mode) of 25 volts or a maximum differential signal of 15 volts at the receiver input terminals shall not result in operational failure of the receiver. The common mode voltage (V_{cm}) is defined as the algebraic mean of the two voltages appearing at the receiver input terminals (A' and B') with respect to the receiver circuit ground (C'). (Designers of terminating hardware should be aware that slow signal transitions with noise present may give rise to instability or oscillatory conditions in the receiving device and appropriate techniques should be implemented to prevent such behavior. For example, adequate hysteresis may be incorporated into the receiver to prevent this condition.)

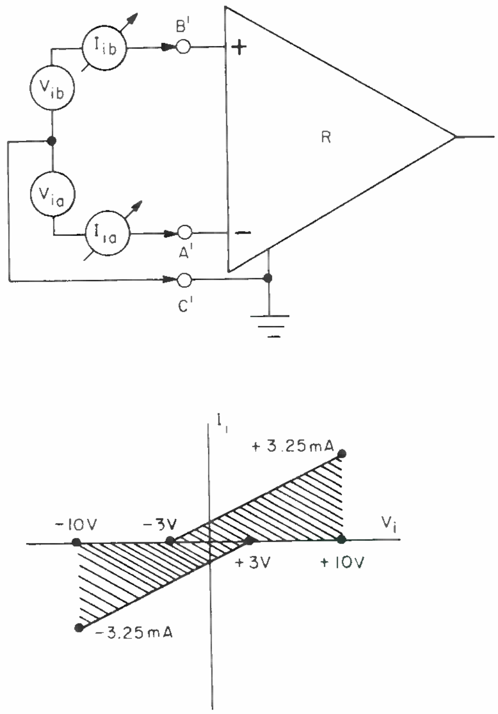
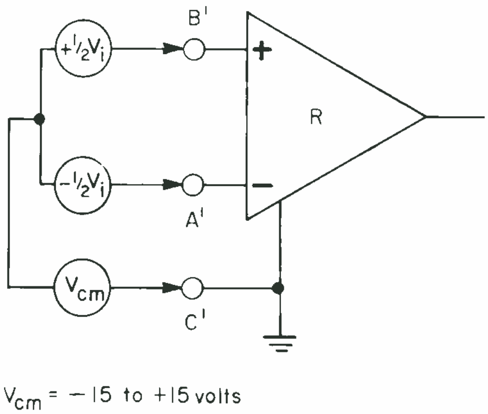


Fig. 4
Receiver Input Current-Voltage Measurement



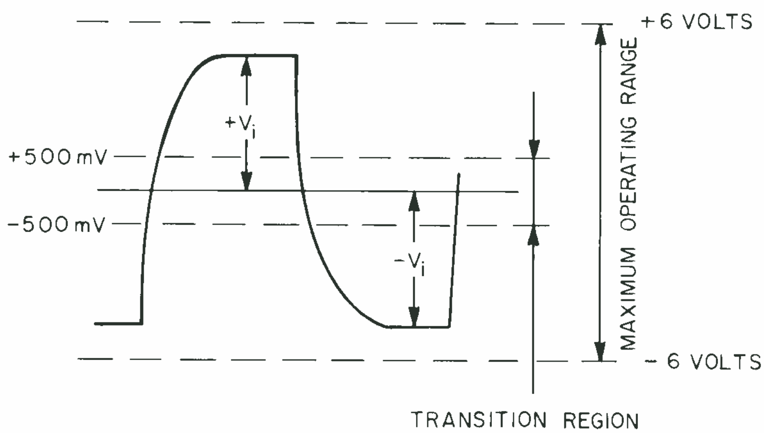


Fig. 5
Receiver Input Sensitivity Measurement

2.3.3 The Input balance measurement shall be made as illustrated in Fig. 6. The balance of the receiver input voltage/current characteristics and bias voltages shall be such that the receiver will remain in the intended binary state when a differential voltage (V_i) of 500 millivolts is applied through 500 ohms $\pm$ 1% to each input terminal and V_{cm} is varied between —15 and +15 volts. When the polarity of V_i is reversed, the opposite binary state shall be maintained under the same conditions.

2.3.4 The use of a noninductive cable termination resistance (R_t) is recommended. A distributed resistive load or a combined R/C load may be required in some cases (see 4.2.1) and the use of an active cable termination resistance is desirable for the purpose of reducing cross coupling when the bus controller is placed in a high-impedance state (see 4.2.2). Care must be taken not to exceed the limits on total load resistance or sensitivity. Refer to 2.3.7 for limits on the total load resistance.

2.3.5 The use of multiple receivers may be employed. Caution must be exercised to avoid performance degradation due to signal reflective effects from stub lines emanating from the load interface point to the receivers.

2.3.6 The interface system shall fail safe. This shall be accomplished by automatic disconnection of a tributary from the interface system in the event of a malfunction or power failure and incorporating in the receiver provisions to provide a steady binary MARK to protect against the following conditions:

Generator power-off

Generator in high-impedance state

Both signal wires open or shorted (Signal common return still connected)

Generator not implemented (Signal leads may or may not be present)

Open connector (Both signal leads and the common signal return are open simultaneously)

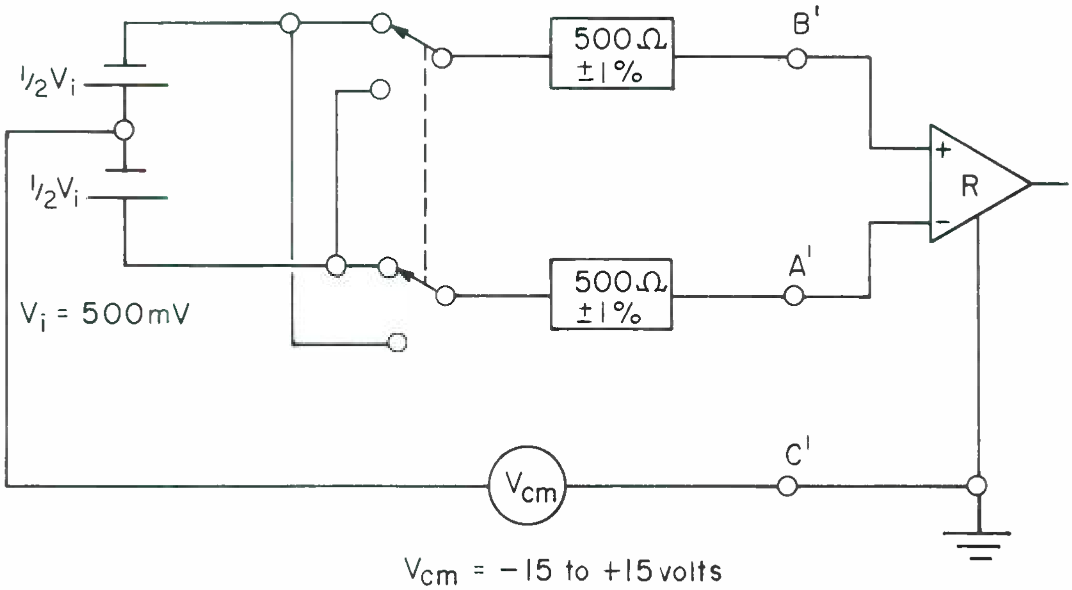


Fig. 6
Receiver Input Balance Measurement

2.3.7 The total load characteristics, including multiple receivers, fail-safe provision, and cable termination shall have a combined resistance greater than 90 ohms between its input points (A′ and B′ [Fig. 1]) and shall not require a differential input voltage of more than 500 millivolts for all receivers to assume the intended binary state.

2.4 Interconnecting Cable Characteristics. The physical and electrical characteristics of the interconnecting cable are given in 2.4.1 through 2.4.4 with additional guidance given in Sec. 4. An interconnecting cable conforming to this standard will result in a transmission line with a nominal characteristic impedance in the order of 100 ohms at frequencies greater than 100 kilohertz, and a DC series loop resistance not exceeding 240 ohms over an operational loop length of nominally 1220 m. The cable may be composed of twisted or nontwisted (flat cable) conductors possessing the characteristics described in 2.4.1 through 2.4.4. Most commonly available cable used for telephone applications (nonloaded) will meet these specifications.

2.4.1 Each conductor of the interconnecting cable shall be composed of either a stranded or solid copper wire conductor with uniform overall diameter of at least 0.5 mm (0.02 in). Use of non-copper conductors is allowed providing they are of sufficient size to yield a DC wire resistance not exceeding 10 ohms per 100 m (30 ohms per 1000 ft) per conductor.

2.4.2 Mutual pair capacitance, that is, the capacitance between one wire in the pair and the other wire in the pair, shall not exceed 65 picofarads per meter (20 picofarads per ft) and the value shall be reasonably uniform over the entire length of the cable.

2.4.3 Stray capacitance, the capacitance between one wire in the cable and all others in the cable sheath with all others connected to ground, shall not exceed 130 picofarads per meter (40 picofarads per ft) and shall be reasonably uniform over the entire length of the cable for any given conductor.

2.4.4 Pair-to-pair balanced crosstalk is defined as the crosstalk between one pair of wires and any other pair of wires in the same cable. This crosstalk shall be attenuated a minimum of 40 dB when measured at 150 kHz with each cable pair terminated in its characteristic impedance tested in the circuit in which it is to be used.

2.5 Environment. A balanced-voltage digital interface circuit conforming to this standard will perform satisfactorily at a data rate of 38.4 kb/s providing that the following operational constraints are simultaneously observed.

The interconnecting cable length is a nominal maximum 1220 m and the cable is appropriately terminated.

The common mode voltage at the receiver is less than 15 volts (peak). Common mode voltage is defined as any uncompensated combination of the generator/receiver ground potential difference, the generator offset voltage (V_{os}), and the longitudinally coupled peak random noise voltage, measured between the receiver circuit ground and cable, with the generator end of the cable short-circuited to ground.

2.6 Circuit Protection. The balanced-voltage digital interface generator and receiver device, in either the power-on or power-off condition and complying with this standard, shall not be damaged under the following conditions:

Generator open circuit

Short-circuit across the balanced interconnecting cable

Short-circuit to any other lead using electrical characteristics in compliance with this standard

Short-circuit to ground

(The faults above may cause the power dissipation in the interface device to approach the maximum power dissipation tolerable by a typical integrated circuit (IC) package. Caution should, therefore, be exercised when multiple generators or receivers are implemented in a single IC package since only one such fault per package may be tolerated at any one time without experiencing IC failure. It should also be noted that the generator and receiver device(s) complying with this standard may be damaged by spurious voltages applied between the input/output terminals and circuit ground. In applications where the interconnect cable may be subject to severe electromagnetic environment or the possibility exists that it may be inadvertently connected to circuits not in compliance with this standard, additional protection should be employed as may be appropriate.)

3. Mechanical Characteristics

3.1 Interface Connector. The interface connector shall be a 9-pin D-subminiature female (DE-9S) with metric (M3) female screwlock. A single interface connector shall be associated with any particular tributary device. Multiple interface connectors may be utilized on a bus controller in the case of a multipoint system or when the bus controller must communicate with more than one interface system.

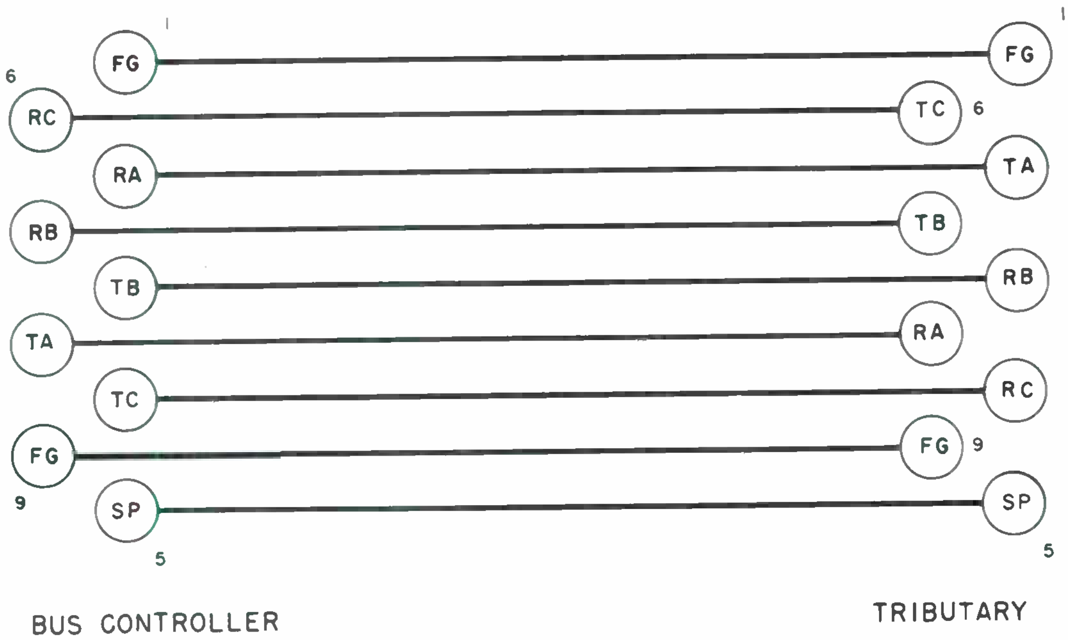
3.2 Pin Assignment. The pin assignments for the bus controller and tributary shall be as shown in Fig. 7. Use of the spare pin for unspecified communication or supervision is not in compliance with this standard. If used, it may not interfere with the normal operation of the standard interface system.

4. Guidelines

When interconnecting equipment using the interface system described in this standard, consideration should be given to some of the problems that may be encountered due to the characteristics of the interconnecting cable, cable termination, the number of devices in use, and grounding arrangements.

4.1 Cable. The interconnecting cable electrical characteristics are specified in 2.4. The following is intended to be used as additional guidance when considering the operational constraints placed on the system by the cable parameters.

4.1.1 The maximum length of cable separating the generator and receiver (load) is a function of modulation rate (influenced by the tolerable signal distortion), transmission losses, the amount of longitudinally coupled noise, and ground potential difference introduced between the generator and receiver circuit grounds as well as by cable balance. Increasing the physical separation and interconnecting cable length between the generator and receiver (load) interface points increases the system exposure to common mode noise, signal distortion, and the effects of cable imbalance. Users are advised to restrict cable length to the minimum consistent with the generator/receiver (load) separation requirements, and whenever possible to utilize cable specifically designed for balanced data circuits.



FG	FRAME GROUND
TA	TRANSMIT "A"
TB	TRANSMIT "B"
TC	TRANSMIT SIGNAL COMMON
RA	RECEIVE "A"
RB	RECEIVE "B"
RC	RECEIVE SIGNAL COMMON
SP	SPARE

A < B "MARK" STATE
A > B "SPACE" STATE

Fig. 7
Connector Pin Assignment

4.1.2 The nominal maximum cable length of 1220 m (4003 ft) is based upon:

Empirical data for nonloaded, twisted-pair telephone cable with copper conductors 0.5 mm (0.02 in) in diameter terminated in a 100-ohm resistive load

Signal rise and fall times equal to or less than 10 microseconds.

A maximum voltage loss between the generator and receiver (load) of 6 dB

(The user is cautioned that the nominal limit of 1220 m does not take into account cable imbalance or common mode noise beyond the limits set in this standard. Operation within the limit of 1220 m should result in a degradation of signal quality that will not exceed a zero-crossing ambiguity of 0.05 unit interval. It is recognized that many applications can tolerate a timing and amplitude distortion greater than this amount and in these cases correspondingly greater cable lengths may be employed. The use of cables specifically designed for the transmission of balanced data signals can also result in the ability to operate over substantially increased cable lengths.)

4.2 Cable Termination. The characteristics of the cable termination are specified in 2.3.4. The following is intended to be used as additional guidance when considering the operational constraints placed on the system by the termination resistance.

4.2.1 The determination of the type of cable termination utilized and its value must take into account the characteristic impedance of twisted-pair cable which is a function of operating frequency, wire size, wire type, and the kind of insulating materials used. The characteristic impedance of nonloaded, plastic-insulated, twisted-pair telephone cable with copper conductors 0.5 mm (0.02 in) in diameter is in the order of 100 ohms when measured with a 100 kHz sine wave.

The characteristic impedance of any cable typically contains an inductive component which could adversely affect the wave shape over extended cable lengths. Use of a composite R/C cable termination with a time constant of approximately 3 times the propagation delay of the cable may result in a significant improvement in the wave shape and a reduction in driving power requirements.

4.2.2 The presence of stray capacitance between the interconnect cable and any adjacent cable (Fig. 8) can result in interference being coupled to an adjacent cable when a transition occurs that is other than balanced. During normal operation, inter-cable coupling is at a minimum due to the use of a balanced transmission mode along with control of signal rise-times (2.2). When the driving device is placed in a high-impedence state, however, this control is no longer applied and the conductor in the pair that was at a positive value will transition to a less positive value. This transition, being uncontrolled and of an unbalanced type, may be coupled into adjacent cables.

Capacitive coupling of this type can be reduced or eliminated by utilizing shielded cables specifically designed for the transmission of digital data information. Although this approach is practical when installing the interface system in new plants, it may not be possible when working in existing facilities. When utilizing nonshielded cable, the use of an active termination is recommended. An active termination of the type shown in Figs. 9 and 10 will result in a balanced transition to a voltage that is equal on both conductors of the cable pair. The circuit shown in Fig. 9 also provides a fail-safe bias on the bus and allows for higher impedance terminations while the circuit shown in Fig. 10 is balanced with respect to the interface bus and does not introduce a differential bias on the line.

4.2.3 In general, reliable operation of the balanced interface circuit is not particularly sensitive to the presence or absence of the cable termination resistor when operating at 38.4 kb/s. The termination of the cable with a noninductive load in the order of 90 to 250 ohms tends to result in the preservation of the rise time in the generated signal and a reduction of line noise. Caution must be exercised in the value of termination selected as too low a value would result in a reduction in signal amplitude to the point where reliable operation of the system would be affected.

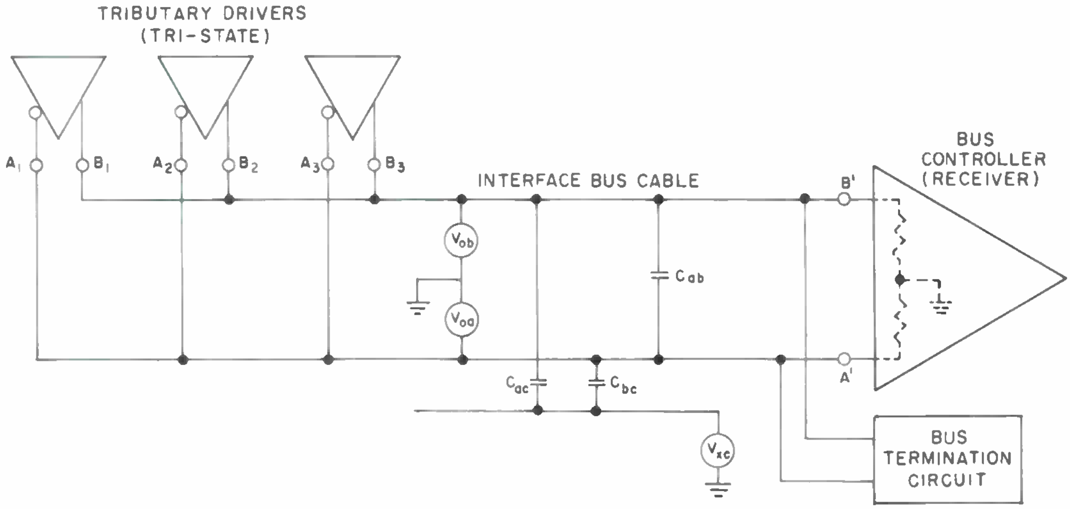


Fig. 8
Cable Model

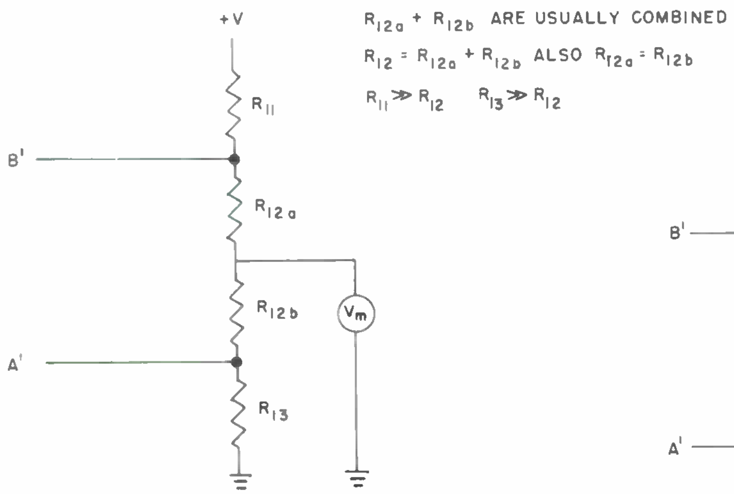


Fig. 9
Terminator Circuit 1

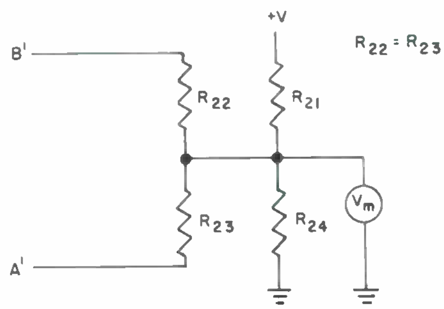


Fig. 10
Terminator Circuit 2

4.3 The data rate of 38.4 kb/s can result in a system response time within the equivalent of one television frame when the total number of tributaries is less than thirty-two. Higher data rates may be used when the operation of a specific system indicates this need. When higher data rates are used, the device(s) shall first establish communication at 38.4 kb/s. Operation at data rates lower than 38.4 kb/s is expressly prohibited due to the possible confusion of certain data patterns with the BREAK sequence.

5.11

Audio Magnetic Recording

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INTRODUCTION

The goal of any audio recording system is to store and reproduce sound as faithfully as possible. Analog magnetic tape is the storage medium on which audio signals are represented as variations in the magnetization level along the tape. As with any analog storage system, performance is limited by the maximum undistorted output level on one hand and noise level on the other.

The purpose of this chapter is to trace the development of magnetic recording and provide you with some of the "why" behind the operation of magnetic audio recorders. If you wish more practical information on maintenance and machine operation, the balance of this chapter contains a wealth of useful information.

AN HISTORIC PERSPECTIVE

In 1897, Danish inventor Valdemar Poulsen invented a rudimentary magnetic wire recorder, which he called the Telegraphone. Unable to market the concept successfully, he sold his Telegraphone patents in 1905.

In subsequent years, organizations in several countries slowly improved magnetic recording, progressing from steel wire and steel tape to paper and celluloid tape. Even with significant improvements made in tape manufacture, primarily by BASF and AEG, the audio quality of magnetic recording was very poor. As late as 1948, 78 rpm lacquer and wax transcription discs, with their associated cutting lathes and reproducing apparatus, were still widely used for broadcast program storage and editing. The poor fidelity of dc bias magnetic recording stalled its commercial acceptance.

The German radio network, Reichs Rundfunk Gessellschaft [RRG], purchased several AEG Magnetophon recorders to see if audio performance could be elevated to broadcast levels. While investigating an improved record amplifier circuit he designed, RRG engineer Walter Weber found performance incredibly improved. Low noise, good frequency response, and low distortion; true high fidelity from magnetic tape. Investigation revealed that his amplifier circuit was oscillating at a high frequency, not an uncommon happening, even today! So, in 1939, Walter Weber stumbled on ac bias of magnetic tape quite by accident. In the same year, Marvin Camras independently discovered ac bias while working on improving wire recorders at the Armour Research Institute in Chicago. AC bias was employed on a Poulsen Telegraphone by U.S. Navy researchers Carpenter and Carlson sometime in 1921, but the project was dropped, and it took another 20 years for ac bias to be "rediscovered".

Late in 1947, the tiny Ampex corporation revolutionized the American recording industry by introducing their model 200, the first truly successful high fidelity tape recorder. Since then, magnetic recording quality has steadily improved, with better head design, tape formulations, and advanced electronic design.

TAPE COMPOSITION AND TYPES

All magnetic tapes for audio recording are composed of microscopic needle-shaped magnetic particles that are attached to a plastic backing material with an adhesive binder. Variations in physical properties of the magnetic material, binder, and plastic backing result in the wide variety of tape types currently manufactured. The magnetic material itself largely determines tape electrical performance.

The oxide or metal particles used in magnetic tape are actually microscopic needle-shaped bar magnets about 0.5 microns long and 0.05 microns wide. (A micron is 1×10^{-6} meters). Each of these tiny particles produces a magnetic field of constant intensity. These particles can be thought of as binary switches which can exist in only two magnetic states. When an external magnetic field of opposite polarity is applied to an individual particle, that particle's magnetic polarity will switch at a certain field intensity. The magnetic field intensity which needs to be applied to a particle to make it switch states is called the coercivity of the particle. Coercivity is measured in Oerstads and abbreviated H.

Not all particles in a batch of magnetic material have the same coercivity. The familiar bell shaped curve describes the overall distribution of particle coercivity in a magnetic tape.

A tape is considered erased when the magnetic particles are randomly oriented so there is no net polarization. That is, there is no external magnetic field produced by the tape, because all its tiny internal magnets are opposing each other. When an increasing external field is applied to an erased tape, nothing happens at first because the field is not strong enough to switch any of the particles. (See Fig. 1) As the external field intensity increases, particles on the low end of the coercivity distribution curve start to switch. More and more particles switch as field intensity increases. When all of the particles are switched, increasing the external field intensity does not change magnetization. The tape is then said to be saturated. The magnetic field retained by the tape after saturation is called the retentivity. Retentivity is measured in Gauss and abbreviated B. The higher the retentivity of a tape, the higher its output. When a certain width of tape is specified, retentivity is expressed as remnance or fluxivity, and is measured in nanoWebers per meter.

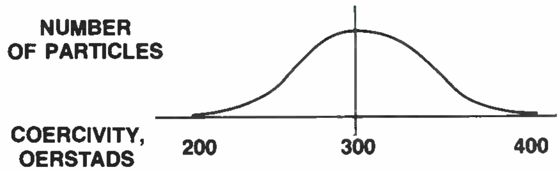


Fig. 1A. Coercivity distribution of magnetic particles in a ferric oxide tape (Gaussian distribution).

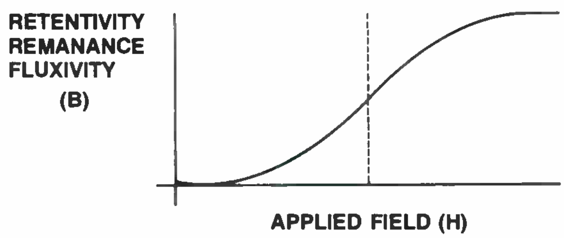


Fig. 1B. Magnetization curve of a sample ferric tape.

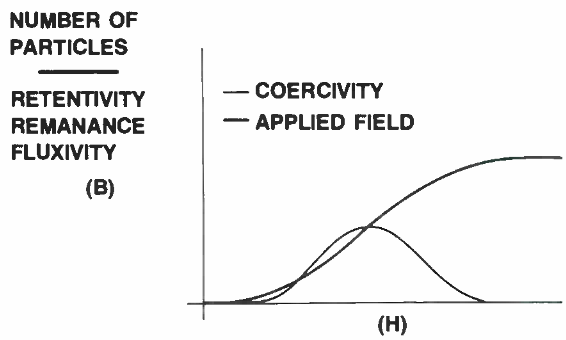


Fig. 1C. DC magnetization curve of the tape is derived by summing all the contributions of the individual particle coercivities and retentivities. More correctly, the DC magnetization curve of a tape is the integral (area sum) of the coercivity distribution of the individual particles.

This usage is the most common. So why is all this important? There are two reasons.

First, the magnetization curve in Fig. 1 is very nonlinear, which means that something needs to be done in the recording process to reduce distortion and produce a linear output. That something is ac bias, which will be covered later.

Second, different tapes have different values of coercivity and retentivity, which affects how they can be used. Tapes with high coercivity require higher magnetic fields for recording and erasing. These tapes are called high bias tapes. High bias tapes cause compatibility problems with

TAPE TYPE	RETENTIVITY	COERCIVITY	CURIE POINT
Ferric Oxide	1200 Gauss	300 Oerstads	470 C
Ferric Oxide Cobalt Doped	1500 Gauss	600 Oerstads	470 C
Chromium Dioxide	1500 Gauss	1500 Oerstads	130 C
Metal Particle	3000 Gauss	1100 Oerstads	770 C

Fig. 1D. Performance of common magnetic tape materials.

older equipment incapable of producing the large field strengths these tapes require. High coercivity tapes are less subject to a phenomenon called self-erasure, which degrades high frequency response.

The compact cassette, with its slow tape speed and inherently poor high frequency performance, led to the development of chromium dioxide and metal particle tapes, which have higher coercivity than conventional ferric oxide tapes. These new tape types have not been a problem in consumer applications, as models and features are changed with the seasons. Broadcast equipment lasts much longer, and most installations standardize on one type of tape so that recorders do not have to be constantly realigned. Improvements in audio mastering tape for broadcast have been directed toward increasing retentivity, and thus output level. High coercivity tapes are starting to become more popular as tape machines are updated to handle increased recording field intensities.

Tapes used in reel-to-reel and cartridge machines are usually ferric oxide types with little or no cobalt doping. The cassette format generally uses chromium dioxide or doped ferric oxide tapes, with some trend toward metal particle tapes for high quality recording.

A caution should be raised concerning the use of chromium dioxide tapes for archival storage of program material. The low Curie temperature of the magnetic material means that this type of tape may be more affected by heat. A controlled temperature environment is important for storage of all magnetic recording materials.

The overall performance of magnetic recording is limited by the tape itself. Maximum output level is limited by tape retentivity (fluxivity) after saturation, and noise is limited by the magnetic particle size. Particle size cannot be arbitrarily reduced to improve noise. Thermal energy would tend to randomize the orientation of very small particles, effectively erasing the tape. Also, electrical noise generated in the tape heads and reproduce circuitry is just a few dB below the noise level of modern tapes. Tape development has focused on increasing retentivity to increase output level, thus improving signal-to-noise ratio. High coercivity tapes improve high frequency response by reducing self-erasure.

THE RECORDING PROCESS

For any given type of tape, the physical track width recorded on the tape and the speed at which it is pulled across the heads will determine the best possible signal-to-noise ratio. Signal-to-noise performance improves approximately 3 dB for every doubling of the area recorded on. Increasing tape speed and increasing track width improve signal-to-noise ratio. Wide track tape running fast will outperform narrow track tape running slowly.

Fig. 2 is a chart depicting several tape types and their relative best-case performance. This chart will help you to evaluate different tape systems to determine which one suits your needs. Maximum level is considered to be the point where total harmonic distortion reaches 3 percent.

THE NEED FOR BIAS

The need for ac bias is best explained by looking back to Fig. 1B, which shows the magnetization characteristics of the tape. The curve shown is very nonlinear, indicating the severe distortion that would be produced if some type of bias were not used. DC bias would reduce distortion by operating in the linear portion of the transfer characteristic, but the limit to this technique seems to be about 5 percent THD. As described above it was accidentally discovered that imposing a high frequency bias signal on the audio waveform improved performance tremendously.

This ac bias frequency is usually about 5 to 10 times the highest frequency to be recorded, and the level is several times the peak amplitude to be recorded. With no audio signal applied, the

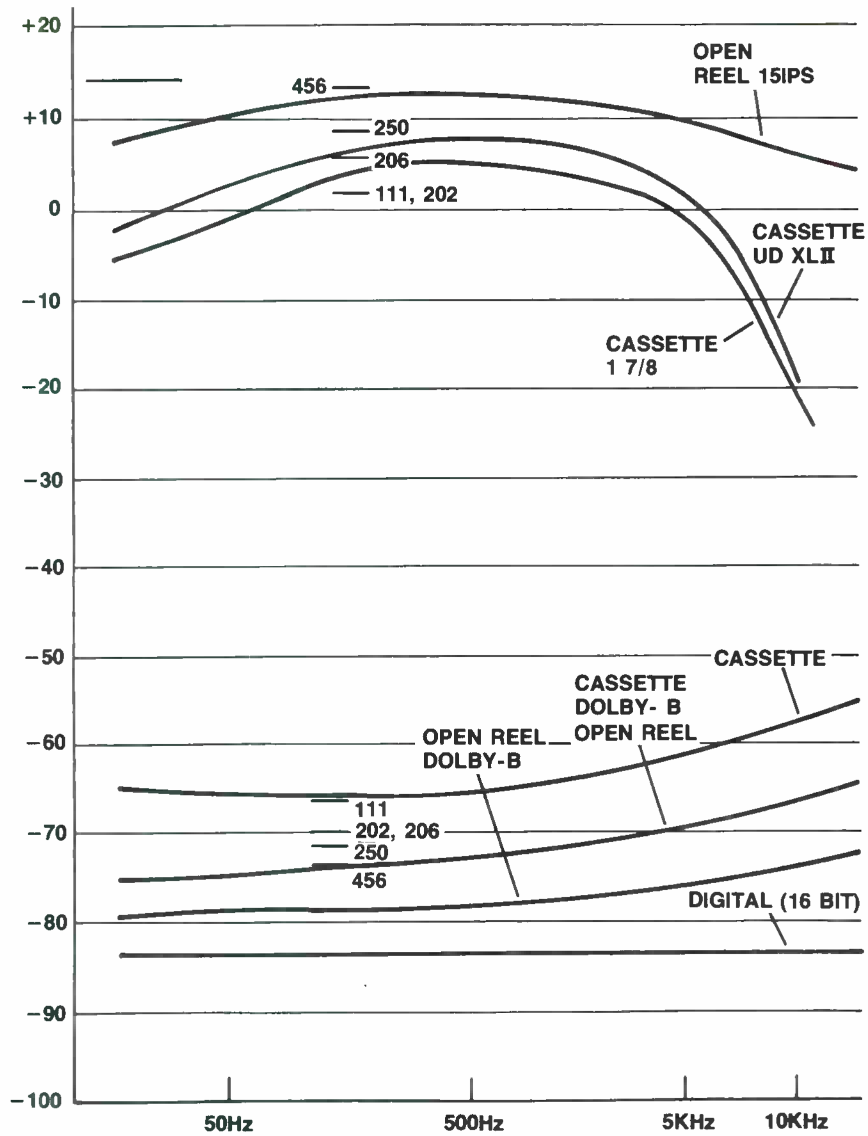


Fig. 2. Maximum output level and noise level vs. frequency for common tapes and formats.

bias waveform alternately flips the polarity of the tiny magnetic particles as they pass the record head. The field strength decays with distance from the trailing edge of the record head gap, resulting in a random distribution of magnetic particle orientation and no net magnetization on the tape. When audio is applied, the bias waveform is offset by the audio signal amplitude, producing a net magnetization of the tape at any given point. When the tape is played back, the audio will be recovered via those magnetization changes, which vary in frequency and intensity according to the original input signal.

Optimum Bias

Tape types have different bias requirements depending on their coercivity. If the bias field is too small, the tape is said to be underbiased. If the bias field is too large, the tape is said to be overbiased. When bias is reduced below its optimum level (underbias), distortion rises dramatically, noise increases and high frequency sensitivity improves slightly. The overall result is one of improved high frequency response with noticeable distortion. When bias is increased beyond its optimum value (overbias), overall sensitivity drops,

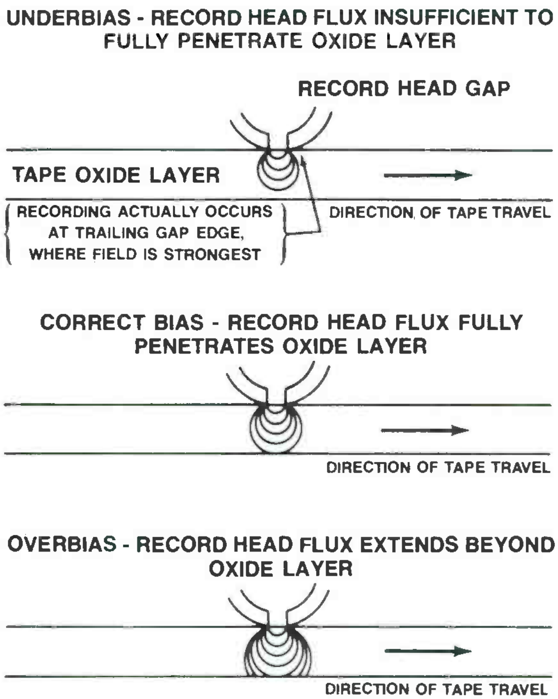


Fig. 3. Bias field penetration.

high frequency response is degraded, noise level drops, and distortion increases slightly. See Fig. 3. Optimum bias can be determined by using a distortion analyzer. Bias should be adjusted for minimum harmonic distortion in the range of 1 to 6 kHz. Since bias affects high frequency response, equalization must be adjusted after determining optimum bias. When setting up machines, it is safer to set bias slightly higher than the optimum for the tape used to make measurements. Then, normal variations in tape composition from run to run will not cause an underbias condition, resulting in excessive distortion.

SELF ERASURE

The magnetic field at the trailing edge of the record head gap diminishes rapidly with distance. However, at very short recorded wavelengths (high frequencies, slow tape speeds), the field is often strong enough to cause partial erasure of information already past the gap. This effect results in high frequency loss and increased distortion. High bias (high coercivity) tapes are less susceptible to self erasure. This means that less equalization is required during recording, improving signal-to-noise ratio and high frequency response. For example, the compact cassette format specifies 70 usec equalization for chromium dioxide tapes, and 120 usec for ferric oxide tapes. Chromium tapes require less high frequency equalization on playback, improving signal-to-noise ratio.

THE PLAYBACK PROCESS

Electrical and mechanical design of the playback head is much more critical than the record head. Since all recording is done on the trailing edge of the record head gap, the gap width is not as important as for a reproduce head. The only critical requirement is that the combination of bias and audio waveforms do not saturate the magnetic material in the record head.

In playback, gap width is critical because it determines the highest frequency that can be reproduced. Making the gap width extremely small reduces the sensitivity of the head. All tape heads have electrical resistance and magnetic reluctance, which produce thermal noise. As the gap width is reduced, output level will drop and signal-to-noise ratio will be degraded past the point where the tape itself is the limiting factor. When the recorded wavelength is equal to the width of the playback head gap, the *net* magnetic field inside the gap will be zero and there will be no output voltage.

Output voltage begins dropping long before the recorded wavelength is equal to the playback head gap width. This effect is called *gap loss*. A good rule of thumb is that the playback head gap should be from 1/4 to 1/10 the shortest recorded wavelength, resulting in a gap loss of 0.9 dB and 0.14 dB, respectively. The following formula may be used to calculate the gap loss knowing the gap width and recorded wavelength.

$$\text{Gap Loss (dB)} = 20 \text{ Log} \frac{\sin (180\ g/l)}{\pi\ g/l}$$

Where:
g = Gap Length in inches or microns
l = Recorded Wavelength in inches or microns

Recorded wavelength is determined by the tape speed and audio frequency in the following relationship:

$$\text{Recorded wavelength} = \frac{\text{Tape speed}}{\text{Audio Frequency}}$$

Wavelength drops as audio frequency increases or tape speed decreases. The wavelength recorded on tape for an audio frequency of 15,000 Hz and different tape speeds is shown below.

TAPE SPEED	RECORDED WAVELENGTH FOR 15 kHz
1.875 ips	0.000125 inches, or 3.175 microns
3.75 ips	0.000250 inches, or 6.35 microns
7.5 ips	0.000500 inches, or 12.7 microns
15.0 ips	0.001000 inches, or 25.4 microns
30.0 ips	0.002000 inches, or 50.8 microns

You can see that head gaps need to be vanishingly small as tape speed drops, leading to inevitable compromises in design that result in poor high frequency performance at low tape speeds.

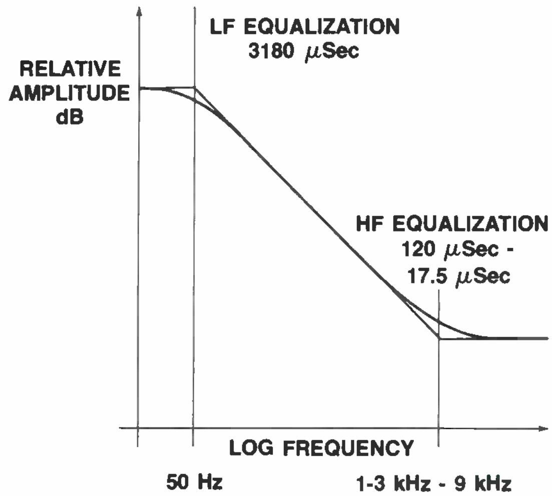
EQUALIZATION

The output voltage of a perfect playback head is directly proportional to the rate of flux change in its gap. The flux change is caused by the changing magnetization of the tape. For constant recorded flux intensity, the rate of change of flux increases with increasing frequency, producing an output voltage that increases linearly with increasing frequency. This rising response, along with the high frequency roll-off due to gap loss, is shown in Fig. 4.

In order to obtain flat playback response, a complementary equalization curve must be employed in the playback amplifier to boost the low frequencies. A playback equalization curve with a 50 Hz turnover frequency is widely used. This is also referred to as the 3180 usec curve. If the record and playback heads were perfect, and in perfect contact with an ideal tape, no other equalization would be necessary.

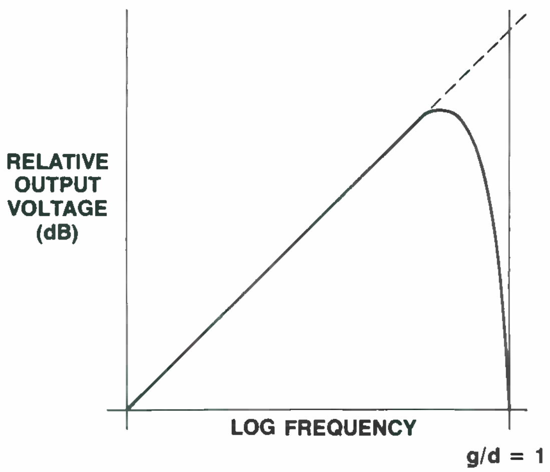
Tape self-erasure, head electrical resistance, inductance, and magnetic reluctance, along with imperfect head to tape contact result in the need to provide additional record and playback equalization.

Equalization is specified as a playback response curve for the total head, amplifier, and equalizer system. High frequencies are effectively boosted in playback by adding a high frequency RC time constant to the normal 3180 usec (50 Hz) RC time constant. What really happens is the roll-off provided by the 3180 usec time constant is stopped by this high frequency RC time constant. The equalization frequency varies from 9 kHz to 1.3



PLAYBACK EQUALIZATION CURVES SHOWING 3180 μSec LF TIME CONSTANT. HF TIME CONSTANT VARIES WITH TAPE SPEED, FORMAT, AND TAPE TYPE.

Fig. 5A. Ideal playback equalization curves.



OUTPUT VOLTAGE FROM IDEAL PLAYBACK HEAD FOR CONSTANT RECORD FLUXIVITY, DOTTED LINE IS THEORETICAL RESPONSE, SOLID LINE TAKES INTO ACCOUNT GAP LOSS.

Fig. 4. Playback head response.

SPEED	U.S. LF HF	EUROPE LF HF
1/4 in open reel		
30.00 ips	3180, 17.5	3180, 35.0
15.00 ips	3180, 50.0	3180, 35.0
7.500 ips	3180, 50.0	3180, 70.0
3.750 ips	3180, 90.0	
1.875 ips	3180, 90.0	
Cassette format		
1.875 ips	3180, 120.0	ferric oxide tapes
1.875 ips	3180, 70.0	chromium dioxide and metal tapes

Fig. 5B. Magnetic tape playback equalization time constants in microseconds.

kHz according to tape speed. See Fig. 5A. Lower tape speeds require more equalization and so have longer time constants (equalization starts at a lower frequency).

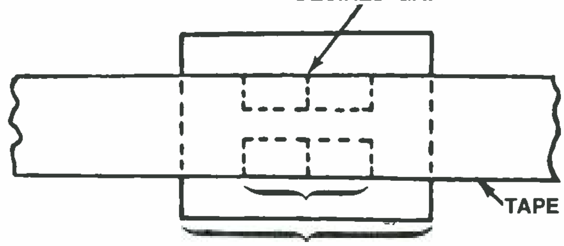
These system time constants and the equalization curves they represent are all referenced to constant fluxivity versus frequency in the playback head gap. These values are for gamma ferric oxide tapes. Many of these values are now considered obsolete, since newer high energy tapes do not require as much high frequency equalization as ferric tapes. The 3180 usec LF time constant is widely used although it is not a recognized standard. Consult the tape electrical data sheet, available from the tape manufacturer, for suggested equalization settings. Because of the variations in tape formulations, establishing a standard brand and type for use throughout the station will simplify setup and maintenance and give more uniform results.

Since equalization has been standardized somewhat, it can be generally ignored, except when setting up and checking the performance of a recorder-reproducer. In order to accomplish this task properly, the recorder-reproducer must be properly aligned mechanically, degaussed and adjusted for flat playback response using a standard test tape. The recorder equalizer is then adjusted to produce flat playback response.

When using cassette equipment, the tape type must be noted because of the different equalization curves employed. A chromium tape (70 usec) played back on a machine set up for ferric tape (120 usec) will sound overly bright due to the resulting 50 usec overall preemphasis (3 dB boost at 3 kHz rising to 10 dB boost at 10 kHz). Conversely a ferric tape played on a machine set up for chromium tapes will sound dull, being 3 dB down at 3 kHz and 10 dB down at 10 kHz. Some cassette players have automatic playback equalization switching keyed by a cutout in the tape housing.

PLAYBACK HEAD CONTOUR EFFECTS

There is a limitation on the lowest frequency that can be reproduced by a playback head. As frequency drops, the recorded wavelength increases and becomes large compared to the length of the entire playback head structure. The playback head structure, including the pole pieces and the case itself, becomes a parasitic gap. (See Fig. 6). Low frequency flux cuts this parasitic gap and produces an output that adds to or subtracts from the desired gap output, depending on frequency. The result is a series of low frequency response dips and peaks called head bumps. The pole piece length can be made very large, moving the effect out of the audio frequency range.



POLE PIECE AND CASE LENGTH PRODUCE PARASITIC MAGNETIC GAPS WHICH CAUSE LOW FREQUENCY CONTOUR EFFECTS, COMMONLY CALLED HEAD BUMPS.

Fig. 6. Playback head contour effects.

Contour effects are most pronounced on tape machines which have insufficient space for heads with long pole pieces. Professional reel-to-reel tape machines running at high speeds have pole pieces with faces in excess of one inch long. Special contour reproduce heads have been developed by several manufacturers for cartridge machines. These heads have been available since the late 1970's and largely overcome contour effects.

MECHANICAL ALIGNMENT

Correct mechanical alignment of magnetic recorders is absolutely necessary for proper electrical performance. Mechanical alignment is performed to insure that the tape contacts the head in the correct orientation. Head orientation is specified in terms of *height, zenith, azimuth,* and *meridian*. Fig. 7 shows each type of physical orientation.

Azimuth is the most critical of all the mechanical adjustments and the most difficult to hold correct in day-to-day operations. When record azimuth is correctly set, the tape will be magnetized in vertical bars exactly perpendicular to the direction of tape travel. At high frequencies and slow tape speeds, the width of these vertical bars is extremely narrow. Any departure from perfect azimuth in recording or playback will cause the tape head gap to cut across one or more these magnetized bars, reducing high frequency output. This azimuth loss effect increases with increasing track width and decreasing wavelength. It can be calculated using the following formula:

$$\text{AZIMUTH LOSS} = 20 \text{ Log } \frac{\sin 180\, w\tan (a/l)}{\pi\; w\tan (a/l)}$$

Where:

w = track width in inches
l = recorded wavelength in inches
a = azimuth error angle in degrees

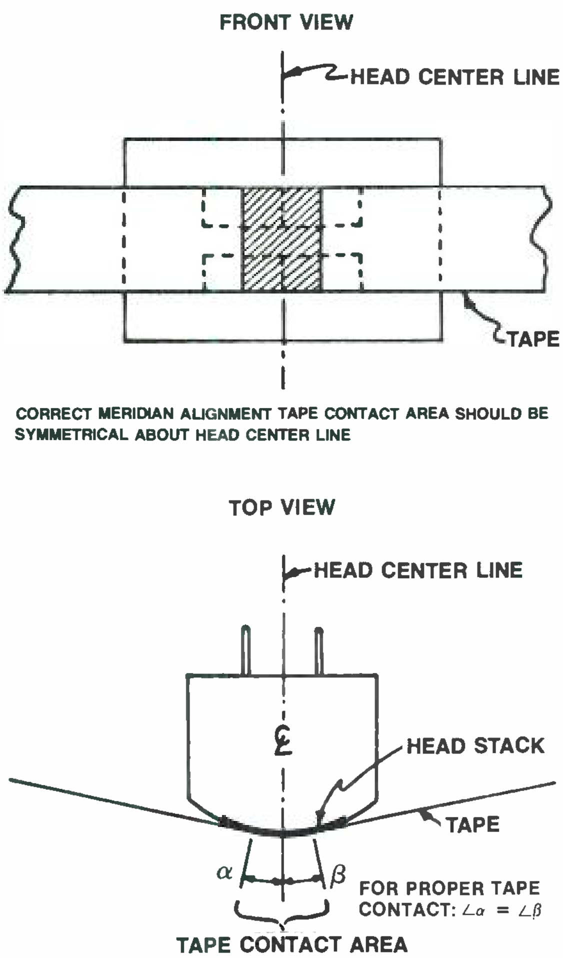


Fig. 7A. Mechanical alignment—meridian adjustment.

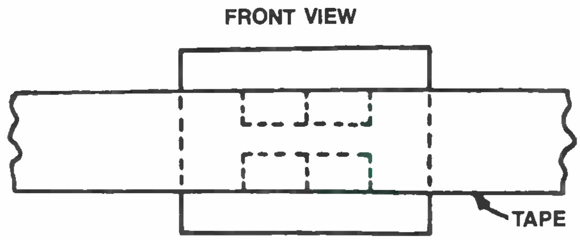


Fig. 7B. Mechanical alignment—height adjustment.

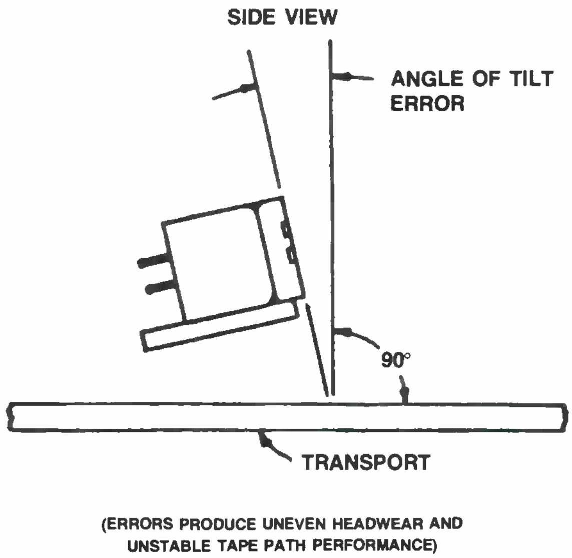


Fig. 7C. Mechanical alignment—zenith adjustment.

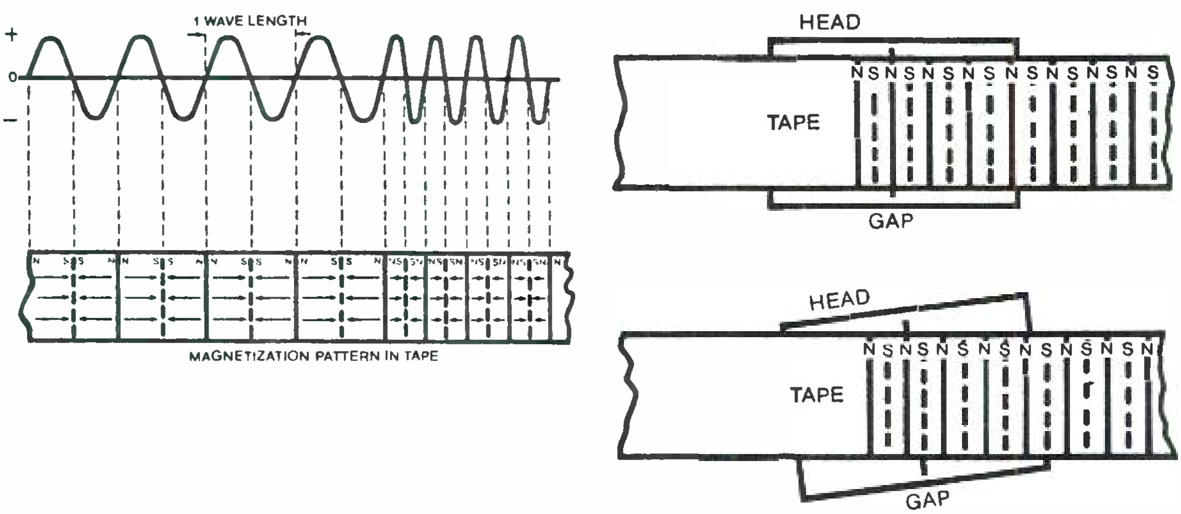


Fig. 7D. Mechanical alignment—incorrect azimuth can cause the head gap to overlap enough of a high frequency cycle, or cycles, to permit some degree of cancellation.

Azimuth errors of 0.5 degree and 0.1 degree will cause the following response roll-offs at 15 kHz:

TAPE FORMAT AND SPEED	0.5 Degree	0.1 Degree
1.875 ips stereo cassette- (track width 0.0235 in)	15.1 dB	-1.6 dB
3.75 ips ¼ in 2 track- (track width 0.075 in)	18.9 dB	-4.3 dB
3.75 ips ¼ in 4 track- (track width 0.043 in)	13.5 dB	-1.3 dB
7.5 ips ¼ in 2 track- (track width 0.075 in)	13.9 dB	-1.0 dB
7.5 ips ¼ in 4 track- (track width 0.043 in)	10.5 dB	-0.32 dB
15 ips ¼ in 2 track- (track width 0.075 in)	7.3 dB	-0.25 dB
15 ips ¼ in 4 track)- (track width 0.043 in)	2.1 dB	-0.08 dB

As you can see, there is no substitute for high tape speed when good high frequency performance is desired. The relatively good performance of the cassette format is due to its narrow track width. Most quality recorders can hold azimuth error angle down to better than 0.1 degree.

In stereo tape machines, azimuth error produces an additional high frequency effect called *mono sum loss*. Azimuth error in a stereo machine produces a slight time delay between the two channels. This time delay causes a corresponding phase shift between audio in the left and right channels. If the phase shift reaches 180 degrees at some frequency, and the two channels are electrically summed to mono, total cancellation will occur, creating a response notch.

SYSTEM PERFORMANCE SUMMARY

Tape performance is determined by the tape type itself, the track width, and the speed. High bias (high coercivity) tapes are less susceptible to self erasure than ferric oxide tapes and so provide better high frequency performance. In addition, they have higher output due to higher retentivity (remenance, fluxivity), which results in better signal-to-noise ratio. All other things being equal, doubling the track width will improve signal-to-noise ratio 3 dB. Increasing track width, however, results in greater azimuth loss given a fixed azimuth error angle. Increasing speed improves high frequency performance by reducing azimuth loss and gap loss. Increasing speed degrades low frequency performance due to head contour effects.

Improving System Performance

Analog tape performance has steadily improved over the years, but signal-to-noise ratio is still a concern. Several noise reduction systems have been developed to improve tape system noise performance. Noise reduction systems fall into two categories. The most effective are encode-decode systems that process the audio signal during recording and playback. Dolby™ and dBx™ are both encode-decode systems. Single ended noise reduction systems operate on playback only, eliminating the need for encoding.

Encode-Decode Systems

Dolby™ noise reduction was the first electronic system designed to reduce tape noise. The basic principle is to boost the high frequency content of low level passages during recording. On playback, a complementary high frequency cut is employed to restore correct frequency response. The end result is about 10 dB better signal to noise ratio (B system). Dolby Labs™ has three main systems for tape noise reduction. Dolby A™ is a four-band system designed for maximum performance and is primarily used in recording studios. Dolby B™ is a two-band system that has found universal acceptance due to its simple setup and good performance. Dolby C™ is similar to Dolby B™, but provides more correction.

The dBx™ noise reduction system uses complementary compression and expansion to improve tape signal-to-noise ratio. Signals are compressed 2:1 before recording. This means that a 20 dB input level change is transformed into a 10 dB level change on the tape. During playback, the recorded signal is passed through a 2:1 expander. This means that the 10 dB level change on the tape is transformed back to the original 20 dB level change. In addition to complementary compression and expansion, the dBx™ system uses preemphasis and deemphasis to further improve signal to noise ratio. Theoretically, a recorder with 60 dB dynamic range could be made to have 120 dB dynamic range.

Single-ended Systems

All single-ended tape noise reduction systems rely on the principle of auditory masking to reduce noise. High frequency noise and noise when no signal is present are the most audible forms of tape noise. Since tape noise is directly proportional to bandwidth, single-ended noise reduction systems reduce playback bandwidth during quiet passages when no high frequency program information is present. DNR and Dynafex are the two most popular single-ended noise reduction systems currently in use. The Dynafex system also uses additional amplitude expansion, an audio processing technique, to reduce playback noise.

All tape noise reduction is limited by a phenomenon called *modulation noise*. The surface of magnetic tape is not perfectly smooth. Irregularities in the tape surface produce a noise which is related to the signal. If you were to record a single tone and then filter out that tone on playback, you would notice that playback noise increases with increasing recorded level. This modulation noise is usually masked by the signal, and is not noticed. Tape noise reduction systems make modulation noise more noticeable. In general, the deleterious effects increase with increasing noise reduction. Additionally, scrape flutter components produced by idler wheels, tape stiction, tape guides, reel flanges, and even pressure pads all contribute to modulation noise components.

THE FUTURE OF ANALOG RECORDING

This chapter was written in 1985, a truly exciting year in the history of audio program storage. The digital audio compact disc and digital audio recorders are sweeping the consumer audio market. Many radio stations are programming extensively from CD's. One must ask the question: How much longer before analog magnetic tape will be obsolete?

Sales of prerecorded compact cassettes exceeded sales of LP records in 1982, about 15 years after the cassette became a popular storage medium. Presently, the high cost of digital audio tape recorders suitable for broadcast use is preventing their universal acceptance. A look back to Fig. 2 will reveal that present tape formulations, properly used, can come within a few dB of the 16 bit digital audio recording process. Since the best broadcast channel signal-to-noise ratios are presently about 80 dB, analog tape will provide excellent cost effective service well into the next decade.

TAPE MACHINE ELECTRONICS

Although the tape machine is a complicated electromechanical system, the main electrical functions of any tape machine can be broken down into four main areas:

1. Bias
2. Equalization
3. Control
4. Interfacing

Bias

A large ac bias current is mixed with the signal to be recorded on the tape to permit low distortion recording (see: THE NEED FOR BIAS). The bias signal is an ultrasonic (60 kHz—200 kHz), low distortion sinewave usually generated by a digital divider and low pass filter in modern equipment (Fig. 8) or a crosscoupled transistor oscillator in older equipment (Fig 9). The bias signal is then mixed with the audio signal either actively or passively and applied to the record head. In passive mixing (Fig. 9), a parallel resonant circuit (L3,C61) is needed to isolate the bias signal from the remainder of the audio circuitry. With active bias mixing (Fig. 10), the isolating property of an op- amp summing junction is used to add the bias signal, reducing or eliminating the need for a bias trap.

The adjustment of bias current is usually performed with a potentiometer in series with the bias line. There are several systems in production which vary the bias current in order to squeeze the last possible amount of head room from the tape in use. Since high frequency audio material acts as a bias signal, the ultrasonic bias can be reduced in proportion to the amount of high frequency audio material present. This allows the high frequencies to be recorded at a higher level than would be possible with a fixed bias system and reduces the possibility of tape saturation.

Equalization

Equalization is the electrical alteration of the amplitude versus frequency characteristic of a particular frequency or band of frequencies. In the absence of equalization circuitry, the response of a tape machine would be the inverted "U" caused by the combination of severe bass and treble loss inherent in the response of magnetic tape heads (See Fig. 4). The factors responsible include the increasing output of magnetic playback head with increasing frequency (6 dB/octave) low frequency "contour" effects and tape self-demagnetization. The equalization circuits of a tape machine create the necessary compensating bass boost and treble boost to achieve flat audio response at the machines's output (Fig. 5). Since the output voltage of a typical head is only 1 millivolt at 1 kHz, the equalization stage also pro-

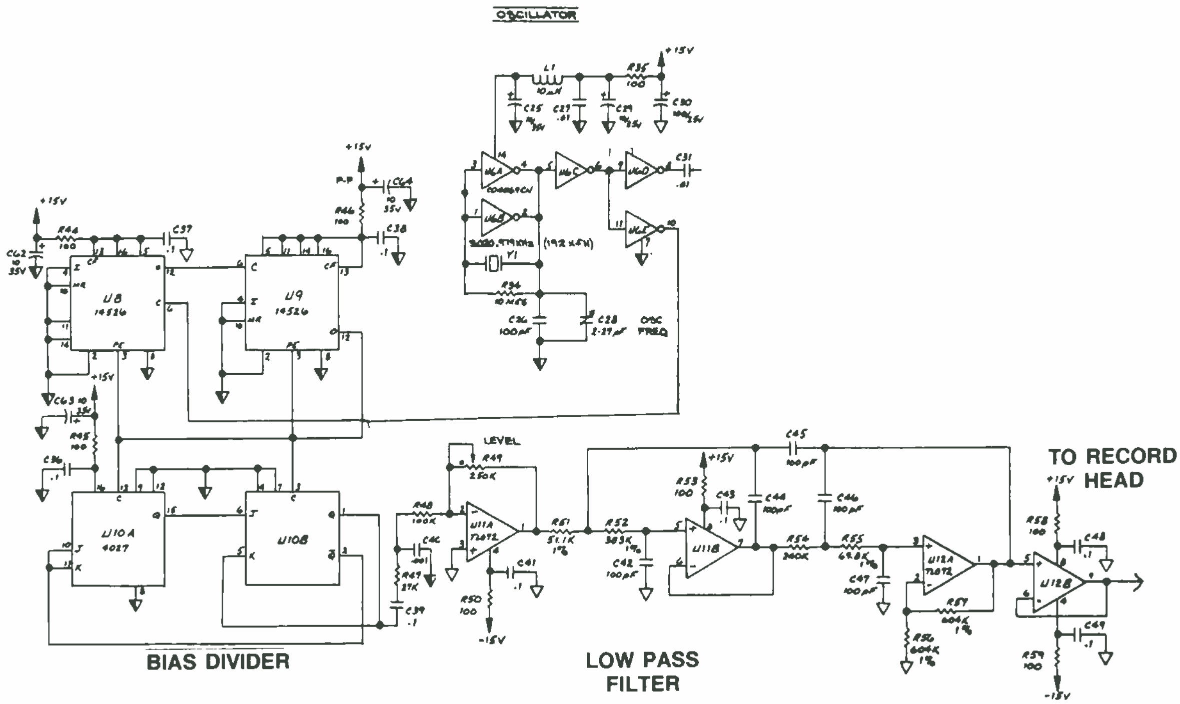


Fig. 8. Digital bias oscillator.

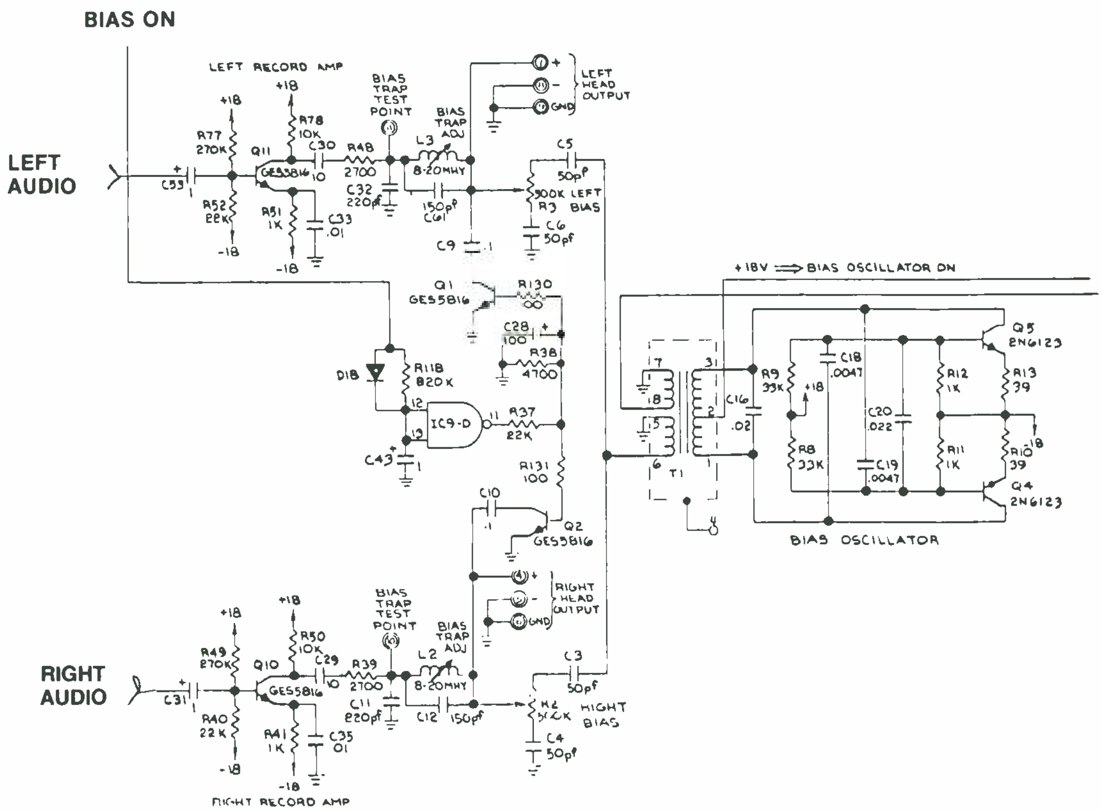


Fig. 9. Analog bias oscillator.

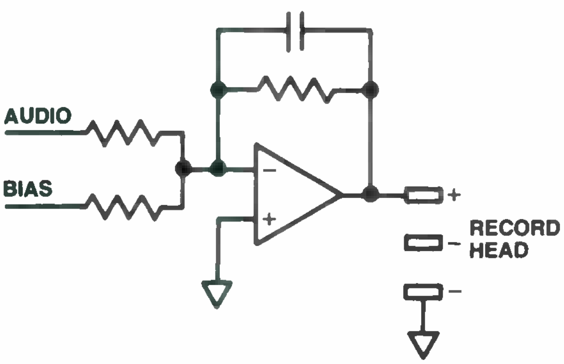


Fig. 10. Active bias summing.

vides some or all of the 60 dB of gain necessary to drive the output circuitry. The low head output voltage also requires that a low noise, high speed circuit be used as the equalization amplifier.

A typical circuit used to generate this playback response curve for a NAB cartridge machine is shown in Fig. 11. The resulting curves are usually referred to in terms of the time constants (or turnover frequencies) of filters which would result in the given amplitude response. The relationship between time constants and turnover frequencies is :

$$F = \frac{1}{2 \pi T}$$

Where:
F = Frequency in Hz
T = Time in seconds

For example, 50 microseconds (5×10^{-5} sec) is equivalent to 3180 Hz. In Fig. 11, the R3/C1 values set the low frequency turnover and R2/C1 sets the high frequency turnover. Both the high and low frequency turnovers allow a small adjustment range to compensate for variations in head performance. The gain of this circuit is about 40 dB at 1 kHz.

Control

The internal control of tape machines is now accomplished with digital logic in either discrete or microprocessor form instead of the mechanical or relay logic found in earlier tape machines. The amount of control included is dependent on the sophistication of the machine and its perceived applications.

All tape machines require the following control logic functions just to interface with the operator:

1. POWER ON
2. PLAY
3. STOP
4. RECORD

The following functions are included in reel-to-reel and cassette machines and some cartridge machines also:

5. FAST FORWARD
6. REWIND
7. TAPE SPEED SELECTION

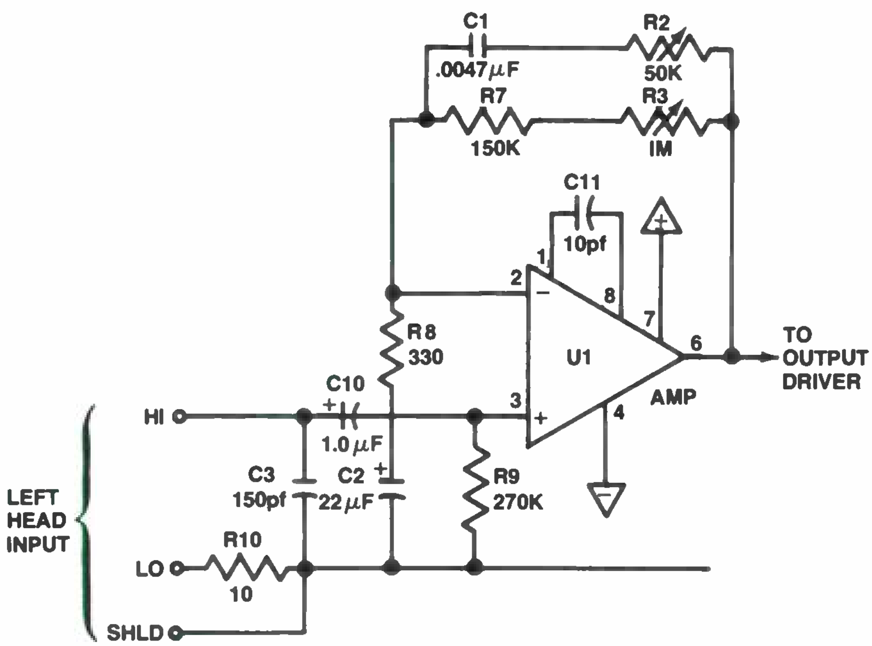


Fig. 11. NAB playback equalizer circuit.

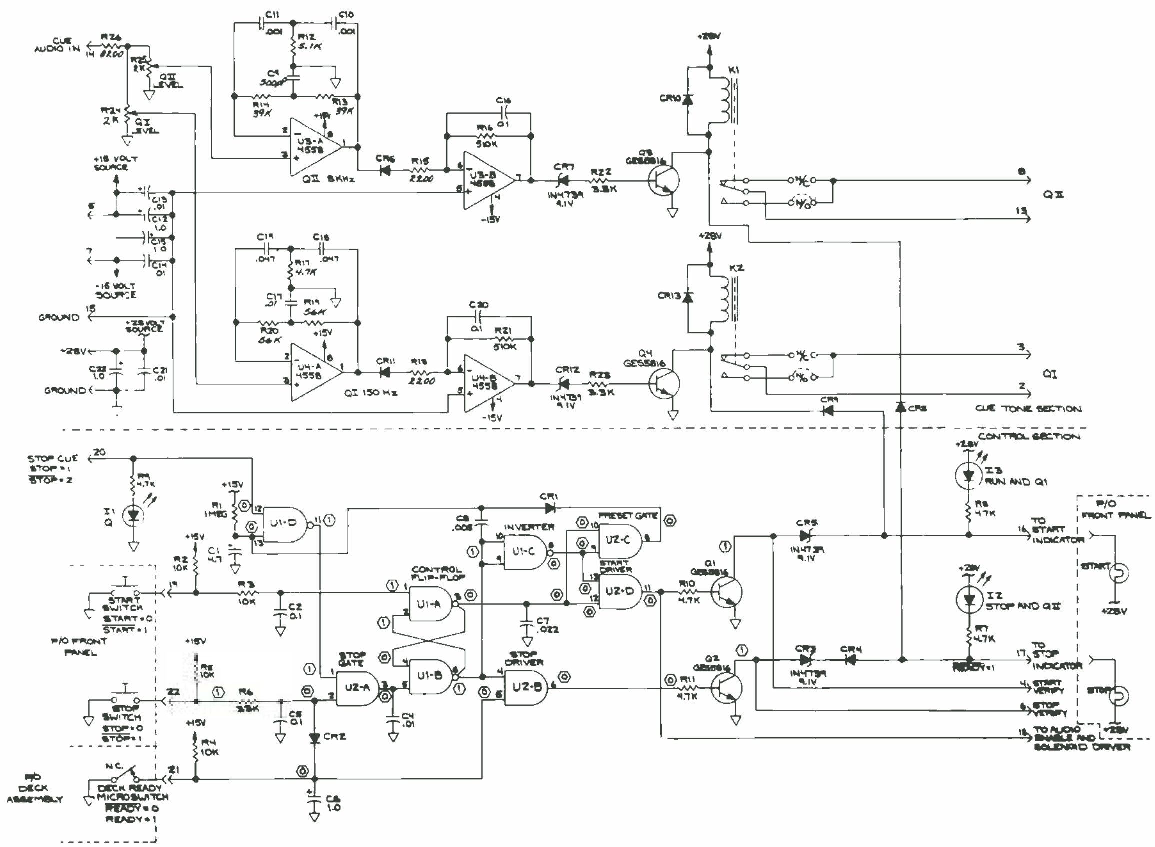


Fig. 12. NAB cartridge control circuitry.

8. EQUALIZATION/BIAS SELECTION

9. NOISE REDUCTION SELECTION

These functions are usually controlled by front panel pushbuttons or internal, user-selectable programmable jumpers.

Beyond these user interface functions, there are also the internal machine control functions to be performed. The control circuitry of a cartridge machine will be used as a example.

The *Control PC Board,* on upper deck, is shown in Fig. 12. Standard circuitry on this board provides:

(1) *START* and *STOP* action for the associated tape transport deck
(2) logic signals that activate front panel status lights
(3) *START* and *STOP* verify signals
(4) sensing and control logic for the Secondary and Tertiary cue tones (optional)

The control board responds to the start and stop pushbuttons, the ready micro-switch, and the 1 kHz tone sensor input.

Fig. 12 shows the logic status of the control board under initial conditions, ie., not loaded with a cartridge. The "0" and "1" figures at the IC terminals (not to be confused with the terminal numbers themselves) indicate the logic status during the unloaded condition. Loading a cartridge, pushing the start and stop pushbuttons, and sensing the 1 kHz tone will change the logic states.

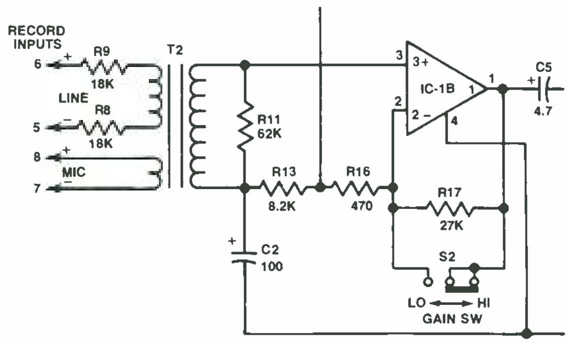


Fig. 13. Transformer coupled input circuit.

When a cartridge is loaded, the inserted cartridge opens the ready micro switch. The Control PC Board then lights the LED on the rear of the board and the ready light in the stop pushbutton on the front panel. CR8 permits the same two indicator lights to display the presence of a Cue II tone by lighting both to full brightness when the tone is present. CR4 provides isolation, while zener CR3 limits the dc voltage to the indicators to 20 volts. Q2 provides a stop/verify current sink to the rear panel connectors for use in applications requiring external status sensing (eg. automation). Other logic elements remain unchanged when a cartridge is loaded.

When the start pushbutton of a loaded deck is pushed, the logic status of U1A and U1B changes, causing the deactivation of all stop/ready indicators and the stop/verify output current sink. Integrated circuits U1C, U2C, and U2D also undergo logic changes; activating the solenoid, the start/verify, the start LED, the the start front panel indicator light.

U2C and U1D combine in a circuit that presets the Cue-stop. When a cartridge is started, U1D is set up to stop the tape motion when a decoded 1 kHz signal is detected by U1D (at terminal 12). R1 and C1 delay U1D for about 3 seconds after the deck starts to insure that a previous stop cue-tone will not stop a newly started cartridge. At the end of the recorded message, the 1 kHz tone will trigger a cue/stop signal in the Control Board, stopping the tape.

Alternately, pushing the front-panel stop pushbutton will also stop the running tape, but will trigger a stop/verify signal. While a tape is running, pushing the stop button changes the logic state of U2A, which stops the tape and turns on all stop indicator lights. This also provides a stop/verify signal to the output connector for external status sensing.

Interfacing

The interfacing requirements of tape machines can be divided into two basic areas; audio and remote control.

Audio

The nominal input level is in the range of 1 volt, but tape recorders must accept a large range of audio input amplitudes, usually between ±20 dB, without overload and with excellent common mode rejection. The two types of recorder input circuits used are the transformer isolated (Fig. 13) and the balanced differential instrumentation amplifier circuit (Fig. 14). With the declining cost of integrated circuits, an excellent transformer coupled circuit is more expensive to produce than an excellent balanced differential circuit. At high input levels low frequency distortion can be a problem in transformer coupled circuits.

There are three types of instrumentation amplifiers usually used as audio input circuits. As in most other applications, the most expensive three op-amp circuit has the best performance. In the circuit of Fig. 14A, the input impedance is low and unequal, the common mode rejection (CMRR) is low and is dependent on the balance of the source's output impedance and the matching of all four resistors. Also, two resistors need to be changed to vary the gain. The circuit of Fig. 14B overcomes most of the drawbacks of the first circuit. Input resistance is high, CMRR is dependent only upon the resistor match, and gain can be set using only one resistor. The major disadvantage is that the common mode voltage input range is gain dependent and can be very poor at high gains. The best choice is Fig. 14C, where the input impedance is high, the CMRR is dependent only on the ratio of R2, R2', R3, R3' not on R1, R1', making the CMRR independent of gain.

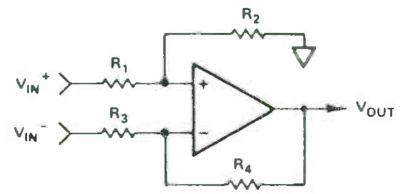
The professional tape machine must be able to drive long, balanced 600 or 150 ohm audio lines at levels between ±10 dBV, with low distortion, reasonable headroom and good dc isolation. These performance criteria require the use of some sort of balanced output driver circuits. The output circuit choices are similar to those for the input circuits, either transformer coupled (Fig. 15) or electronic balanced and floating (Fig. 16). The transformer coupled output offers very good CMRR and very good dc isolation with disadvantages in size, cost and higher low frequency distortion.

There are three choices in electronically balanced output configurations shown in Fig. 16. The circuit shown in Fig. 16A has the virtues of simplicity and one resistor gain adjustment, but is not truly balanced. If you ground the positive output terminal the output drops 6 dB, but if you ground the negative output terminal all output is lost! The circuit of Fig. 16B loses 6 dB of output if you ground either output terminal. The circuit in Fig. 16C is a balanced and floating design, which is an electronic approximation of a transformer. This design includes crosscoupled feedback which keeps the output level constant even if one output terminal is shorted.

Remote Control

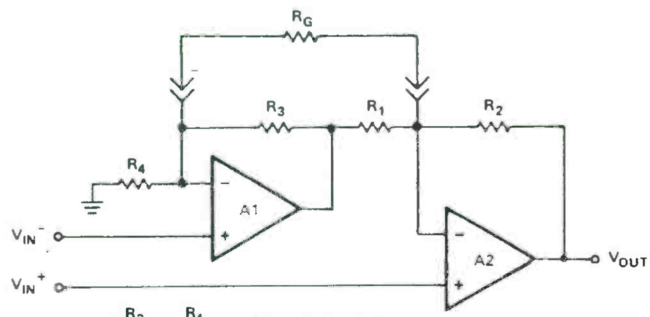
The remote control interface to most professional tape machines is a simple open-collector active pull-down transistor circuit, with the appropriate pull-up supply voltage available at the remote connector of the machine (Fig. 17). Some microprocessor-controlled machines do have serial interfaces, usually RS-232, available also. Although barrier strips and "Jones" connectors were once the connectors of choice, most new machine remote interfaces are the 25-pin "D" connectors made inexpensive and widely available by the personal computer industry. Some machines have dry relay contacts available for other interface controls.

For some machines with digital control logic, special optically isolated remote control interfaces



$$V_{OUT} = V_{IN}^{+}\left(\frac{R_2}{R_1 + R_2}\right)\left(\frac{R_3 + R_4}{R_3}\right) - V_{IN}^{-}\left(\frac{R_4}{R_3}\right)$$

Fig. 14A. Simple differential input circuit.



$$\text{IF } \frac{R_2}{R_1} = \frac{R_4}{R_3}, \quad V_{IN} = V_{IN}^{+} - V_{IN}^{-}$$

$$\frac{V_{OUT}}{V_{IN}} = 1 + \frac{R_2}{R_1} + \frac{2R_2}{R_G}$$

Fig. 14B. Two op amp differential input circuit.

may be necessary to prevent crosstalk between control circuit and avoid digital ground loops.

One of the problems in using cassette machines in professional applications is the lack of machines with easily implemented remote controls. The consumer units usually use a hand held infrared remote, which requires circuit surgery to use the simple contact closure remote controls available from consoles and other professional equipment.

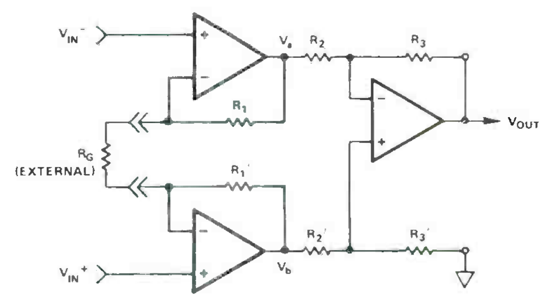


Fig. 14C. "Classic" three op amp differential input circuit.

TAPE MACHINE PERFORMANCE COMPARISON

The following material is presented in order to give an idea of the relative performance level of each format presented. It is representative of the median performance level of professional machines gleaned from industry test reports of machines in each category.

I. SIGNAL-TO-NOISE (3% THD Reference Level, No Noise Reduction)

Cassette	@55 dB
Reel-to-reel	@65 dB
Cartridge	@60 dB

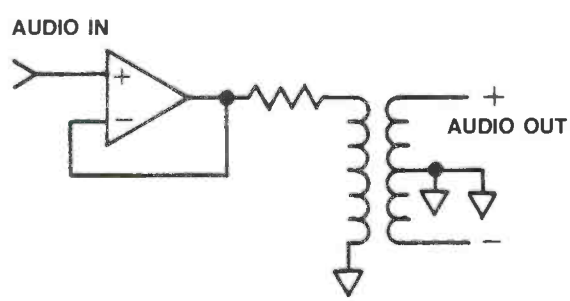


Fig. 15. Transformer coupled output circuit.

II. DISTORTION (THD referenced To Normal Operating Level)

Cassette	1.0%
Reel-to-reel	0.5%
Cartridge	1.5%

III. FREQUENCY RESPONSE (±3 dB)

Cassette	(1 7/8 IPS) 20-20 kHz (-20 dB below reference)*
Reel-to-reel	(7.5 IPS) 20-22 kHz (-10 dB below reference)
Cartridge	(7.5 IPS) 30-18 kHz (-10 dB below reference)

*The small tape area and slow speed of cassette machines mean that they are more susceptible to tape saturation, so cassette machine frequency response drops drastically at higher levels such as -10 dB or 0 dB below reference.

IV. HEADROOM

Cassette	3 dB (above 200 nWb/m)
Reel-to-reel	10 dB (above 320 nWb/m)
Cartridge	10 dB (above 250 nWb/m)

V. WOW & FLUTTER (DIN WTD PEAK)

Cassette	.05%
Reel-to-reel	.04%
Cartridge	.1%

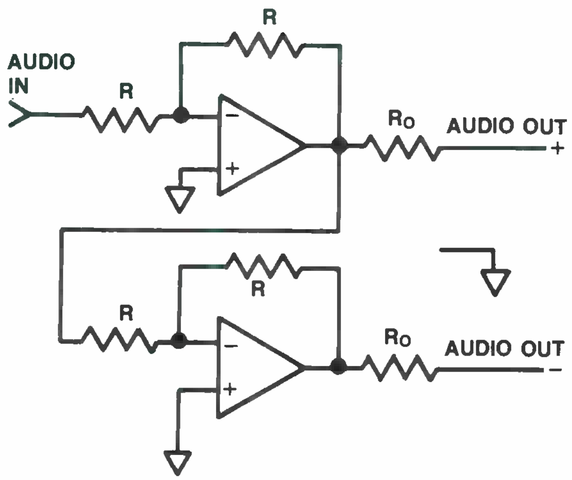


Fig. 16A. Simple op-amp line output circuit.

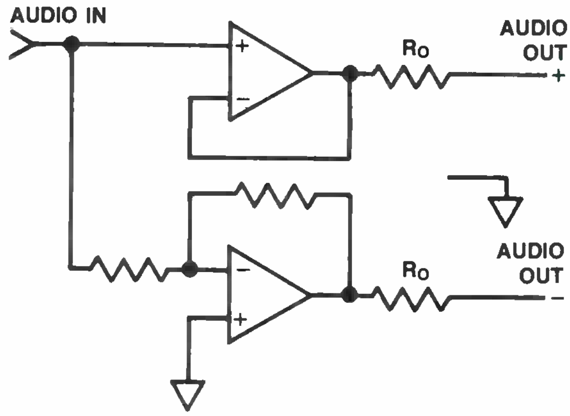


Fig. 16B. Balanced (not floating) line output circuit.

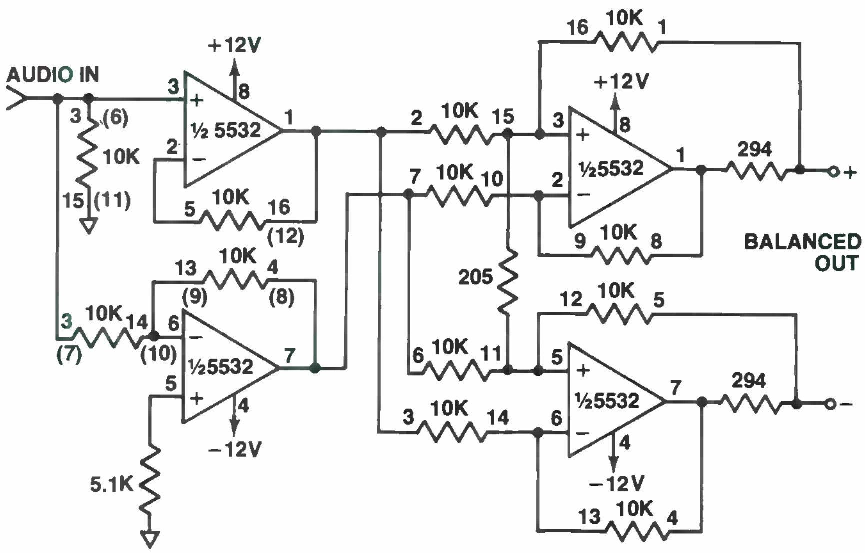


Fig. 16C. Balanced and floating line output circuit.

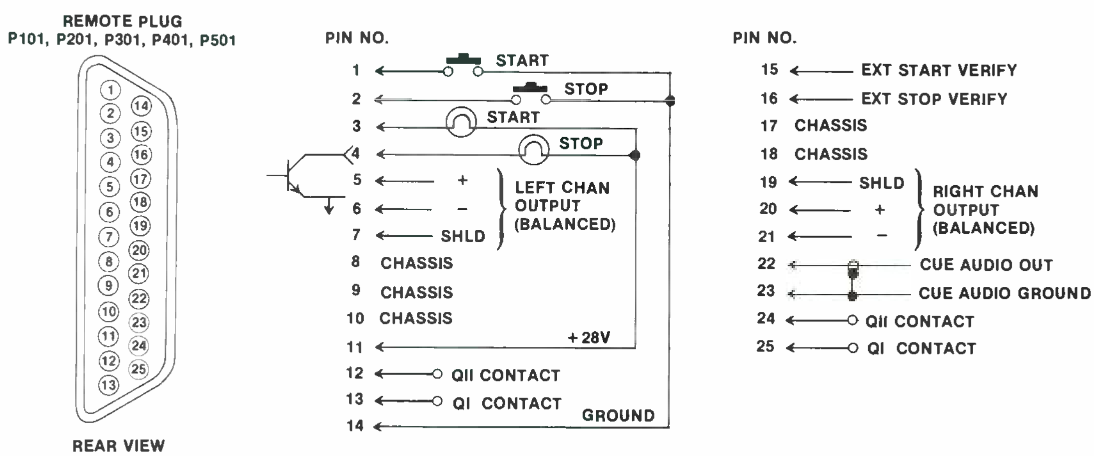


Fig. 17. Typical remote control interface circuit.

A Note About Reference Levels

Each format has its own particular "official" tape reference flux level. Manipulating the reference level in published material to enhance machine specifications (especially signal-to-noise) is a old and hallowed sales technique. The table below gives relative reference levels for several of the more popular levels and the formula for calculating the difference. Simply adding or subtracting the dB difference in reference levels between two different sets of specifications will allow a more accurate specification comparison.

FLUX LEVEL (nWb/m)	dB DIFFERENCE
160	(NAB 1975 Test Tape Reference)
185	1.3 (Ampex Reference Tape)
200	2 (MRL Reference Tape)
250	3.9 (Elevated Level #1)
320	6.1 (Elevated Level #2)

FLUX LEVEL (dB relative) = 20(log new level-log reference level)

Example: The cart machine being considered for purchase has a signal-to-noise of 65 dB referenced to 400 nWb/m instead of 160 nWb/m reference level for another cartridge machine which has a signal-to-noise of 57 dB.

Difference = 20(log 400-log 160) = 7.96 dB
57 dB + 8 db = 65 dB

This calculation shows they basically have the same specification.

FUTURE TAPE MACHINE DEVELOPMENTS

The performance of the conventional analog tape recorder has reached the point of diminishing returns. Recent advances in oxide particle size and packing, the introduction of alloy tape formulations and new head materials have given analog tape better dynamic range and high frequency performance. The addition of various noise reduction systems has slowly increased the sound quality of conventional tape systems. However, at the present stage of development, it is increasingly difficult to carry out any further major improvements in the performance of conventional tape recorders due to physical limitations.

One of the ways in which to overcome this problem is to use digital techniques of signal representation. The most prevalent form of digitizing for audio signals is pulse code modulation (PCM). There are three fundamental differences between digital and analog tape recorders.

1. A PCM tape recorder requires 30 times the bandwidth required for an analog recorder, requiring increasing recording density.
2. Because bits are recorded on PCM equipment, it is possible to disperse the data bits comprising one channel onto a number of recorded tracks.
3. Linearity is not important because only two states are recorded on the tape.

In May of 1967, NHK gave the first public demonstration of a 12-bit companded PCM digital audio tape recorder designed around a 1-inch, 2-head helical scan VTR.

In May of 1978, the EIAJ standard for PCM adapters was adopted and in 1985 a digital audio stationary head (DASH) tape recorder standard was proposed. With the advent of the Compact Disc, digital audio is becoming accepted in both the consumer and professional worlds.

There are basically two PCM recording systems in use, the rotary head (VTR based) and the stationary head (DASH) format recorder. The simpler, cheaper rotary head system will be used for consumer home recording. A DASH format recorder is necessary for recorders requiring sophisticated editing capability for 4 channels or more. The professional two-channel recorder is the area where the two systems are competing with each other. It will require more technological innovations to make two-channel stationary head systems cost competitive with two-channel rotary head systems.

AUDIO TAPE MACHINE MAINTENANCE

Introduction

This section will cover general maintenance procedures along with specific maintenance areas such as heads, pressure rollers, motors and electronics. There will also be a short discussion on troubleshooting and some general considerations. All tape machines have one thing in common; they cannot perform to their designed specifications if they are not properly maintained. Often the maximum performance of a machine is determined by the level and frequency of care rather than design or construction limitations.

General Maintenance Procedures

One of the best tools for keeping tape machines in good working order, especially in a broadcasting environment where many such machines are involved, is a scheduled maintenance system. The design and implementation of a good maintenance system can prevent most common types of problems from affecting the on-air signal.

The best type of system identifies each individual machine by serial number, location in the system and date of original installation. Since maintenance level varies with the area of use, it is essential to know the location of each machine at all times. This tracking of individual units allows rotating machines with spares, and it gives an accurate record for insurance and warranty information. The location record for each unit should also include its problem history record. This problem history can provide information on machines that would otherwise be lost. This information can help point out all types of trends such as the need for increased maintenance in a certain area, shortcomings in equipment design, and environment problems.

One of the most important pieces of information that can be provided by the equipment history card is the cost of repairs. This helps in deciding when it is no longer economical to continue to repair a tape machine.

There are many possible formats for logistically handling both the maintenance schedule and the problem history; however, with the advent of business and personal computers, the job has become much more simplified. It is now possible to set up your system so that at the beginning of each day, or each week, you can obtain a printout of the work scheduled for that given week or day. This allows efficient scheduling in either a large or small installation.

Putting a Machine in Service

The first step in setting up any system is establishing initial conditions: design specifications and the results of initial tests by the manufacturer and purchaser. When a new machine is received, it should have a test data sheet with it, usually included with or bound into the service manual. This will show results of final tests conducted by the manufacturer prior to shipment, typically:

- Signal-to-noise ratio for a given flux level,
- Frequency response at one or more flux or record levels,
- Operating levels,
- Wow and flutter and measurement standard,
- Distortion in percent for a given flux or record level and frequency,
- Crosstalk between channels at a given frequency and record level,
- Verification of functions such as tone sensing and start and stop times.

These data should be checked to the extent that time and available test equipment permit. Then the new machine should be re-equalized for the station's preferred tape and any new measurements recorded.

This overview of maintenance systems is intentionally broad. While a program for maintenance is essential, the specific system must be tailored to fit the individual installation. It must be set up so that it is easy to work with and convenient to use, otherwise it will not be followed. Use something that works for you.

Daily Maintenance

Daily maintenance tasks are those that must be performed routinely to insure continued good performance and prevent failures not related to mechanical wear or electronic failure. These are some of those tasks.

Head Cleaning

This is probably the most often performed maintenance task and one that is given possibly the least consideration. It is very easy to provide a cotton swab and a bottle of alcohol to the machine operators and tell them to clean the heads at the beginning of each shift. However, consideration should be given to several of the items involved in the head cleaning process.

First, head cleaning fluid; there are nearly as many head cleaning fluids available as there are heads to clean. Everyone has a favorite, and there are just a few basic things to consider when using head cleaners.

- Safety for the user. Some liquids, such as carbon tetrachloride, which might be used as head cleaners are dangerous to breathe. The more common alcohol cleaners are effective and safer.
- Safety for the tape, heads, and other hardware. Strong solvents might dissolve materials used in the construction of heads, rollers and other hardware. Tape, itself, may be vulnerable to solvent remaining after the cleaning process. Furthermore, a liquid may be spilled or oversprayed and cause damage to other parts of the equipment.

Lacking specific recommendations from the equipment manufacturer, the safest cleaners are those made with ethyl alcohol (ethanol) or isopropyl alcohol or freon. These substances evaporate quickly and leave no residue. Do not use rubbing alcohol as it may contain some substances which will not evaporate.

Remember that the more effectively a solvent dissolves the oxide buildup on heads, the more effectively it may dissolve tape and other materials.

Second, the cleaning utensil. Cotton swabs, available at drugstores and drug departments, are the most commonly used means of cleaning heads. These provide the user with some extra reach so that hard-to-get-to heads can be cleaned more easily and thoroughly; however, they have two main drawbacks. The first is that they tend to shed the cotton that is twisted onto the ends of the stick. Second, the surface that can be put in contact with the head at any one time is quite small. If swabs are used, the heads should be carefully inspected for cleanliness and debris before putting the machine back in service.

Another common head cleaning utensil is a clean, soft lint-free cloth. This method allows pressure to be applied as needed to remove dirt build-up. There are, however, two drawbacks to using a cloth. Sometimes it is not possible to get your fingers into some of the heads in some machines. Extending your reach by wrapping the end of a screw driver or pencil eraser is not advisable because of the possible damage that could be done to the head assembly. The other problem is possible excessive pressure on the heads. This may cause mis-alignment of the head assembly. Finally, tools such as screw drivers tend to become magnetized and should not be placed close to heads and other tape contact surfaces.

Tape Path Cleaning

Another daily maintenance chore that goes with head cleaning is the cleaning of the rest of the tape path. Pressure rollers, capstan shafts, idler wheels, guides, etc., should also be cleaned with the same diligence as the head. While they are not in the audio path itself, they contribute to the overall performance of your tape machine by affecting such parameters as wow and flutter and pulling torque.

Head and Tape Path Demagnetization

Another task that should be performed daily, demagnetization, or degaussing, is a two edged sword. Not only can heads become magnetized through normal use, but improper demagnetization can also leave residual magnetism in the head, the result of which is not only poor audio performance from machines, but partial erasure of high frequencies on the tape.

A variety of head demagnetizers is available from various sources; however, the following should be taken into account when purchasing a demagnetizer. Heads for tape machines have a wide range of hardness which affects the retentivity of the material. Some of the harder heads consequently require a much more powerful demagnetizer. An under-powered demagnetizer, no matter how many passes are made at the head, will not demagnetize an extended-life head. The other prime factor in buying head degaussers is the accessability of the heads. Some degaussers are available in cartridge form, and while they do an excellent job on cart machines heads, they are, of course, not suitable for any other application. Be sure your demagnetizer is powerful enough and will fit the units you need it to service.

After selecting the appropriate tool, it is no less important to use it properly. If operators are to perform the degaussing of heads, they should be properly trained. In general, the procedure should include the following steps:

- Turn off power to the tape machine. The degausser induces strong magnetic fields which could severely overload sensitive circuitry.
- Turn on or plug in the degausser at least 3 feet from the tape machine.
- Slowly wipe the tip across the heads, guides, rollers and other components in the tape path. Avoid touching tape contact surfaces unless the tip is protected with a clean, soft cover material.
- Slowly remove the degausser while rotating. The degausser should be at least 3 feet from the heads before it is turned off.

In order to verify that the heads have been completely degaused, take a blank tape and check the machine's signal-to-noise figure. If it shows an increase in the signal-to-noise over the normal level, repeat the demagnetizing process. Once the people performing this task have gotten used to the tool, it becomes an unneeded check. Just as in the head cleaning process, all other metallic components in the tape path should be demagnetized regularly.

These daily maintenance tasks should be performed without fail. Postponing these tasks can only result in poor on-air performance. Be diligent!

Weekly Maintenance

Weekly maintenance should consist mainly of a brief physical inspection of the unit for obvious problems such as noisy motor bearings, excessive debris on the deck or in the guidance from oxide shedding. In other words, just make it a point to look at the machine and verify that it is still in the building. Also talk to the operators. They can help you head off failures by letting you know if a machine is acting differently. Operators are not the enemy.

Monthly or Quarterly Maintenance

This section is given a dual heading because of the variation in usage at every installation. The frequency of the checks outlined here may vary. It will depend on how the machine is used, the environment it operates in, and the reliability of the individual unit. Some units may require that these checks be made more frequently than others, but this is something you will not know until you have developed some service history records. Start by using a relatively short interval; and if no corrections are required, you can lengthen the interval. However, remember that the primary purpose of a regular maintenance program is to *prevent* failures, not fix them. Do not allow the service interval to become too long. Also, most of these checks require that the machine be taken out of service, so it would be advisable to stagger the schedule.

General Cleaning

Each unit should be carefully dusted by using either a soft-bristled brush or low pressure air. Covers should be removed during this process, where it is convenient, so that debris is free to fall clear of the machine. This is also a good opportunity to clean the tape path thoroughly and demagnetize all the areas that are difficult for the operators to get to.

Visual inspection of the mechanical parts should be made to check for head wear, deck wear in cart machines, guidance wear, and any other mechanical parts that are subject to fatigue or constant abuse.

Motors should be checked for bearing noise. While small amounts of bearing noise may not be a sign of eminent failure, they should be considered a warning signal. Listening to the mechanical sounds of a machine can often tell as much as more sophisticated tests for wow, flutter and torque.

General Performance Check

Aside from any obvious faults that you or your operators may have noticed in a unit's on-air performance, there are a few quick checks which will determine if the unit is performing up to standard. First of all, typical performance numbers for each machine should be kept on file with the history card. Most manufacturers will provide test data sheets that provide all this information. Data taken at each performance check should be compared back to the original. Second, the test tapes used should be those recommended by the manufacturer and adjustments made should be done in accordance with the specific equipment manual.

Azimuth and Phase Response

The significance of these adjustments is discussed in another section of this *Handbook* so the only comment here is they are important to good performance. If you find that azimuth is unstable or hard to align, look for some mechanical problem in the head mounting assembly or the tape guidance system. The important thing is to get a good "feel" for how the machine reacts to adjustments. This is the best way to obtain an accurate setting.

Playback Frequency Response

A quick check of playback response will provide warning of the beginning of head wear and drift in response and output levels. A worn playback head will begin to lose the high end response. A certain amount of this can be compensated for by re-equalizing the playback electronics.

Record Frequency Response

Proper record frequency response is essential to the sound of your station. A quick check of this spec can tell if the unit needs complete record alignment and will provide information concerning wear of the record head.

Record/Playback Distortion and Noise

These two parameters will give more clues as to the overall performance of the tape machine electronics than any other measurements. Poor signal-to-noise figures or high distortion can be the forerunners of many more severe problems that can be prevented if action is taken early. Take the time to investigate these symptoms when they begin to appear.

Wow and Flutter and Tape Torque

These measurements can indicate impending failure of motors or bearings. The equipment to measure wow and flutter is expensive and usually not found in stations, but it can be rented for periodic checks. Listening to the sounds of motors and bearings can tell a lot and listening to a tape with a continuous tone will give a good qualitative check of wow and flutter.

For cart machines, a quick test of wow, flutter and torque can be made using a tone recorded on the longest cartridge available, thus placing the greatest load on the mechanical system.

Although the five performance checks described above represent a complete testing of the unit, they can be performed in less than 30 minutes per machine in most cases. The time spent on these tests is well worth it.

Semi-Annual Maintenance

Maintenance performed at this interval should be an in-depth version of the monthly or quarterly tasks. Complete specification verification should be done along with a thorough mechanical analysis. It should be determined that all motors have sufficient torque, both starting and pulling, that solenoids, brakes, and other electromechanical devices all operate smoothly, and that all mechanical operation is quiet.

Again, at this point, the timing of these procedures becomes dependent on the use of the machine. In machines that receive relatively light duty, this thorough examination would not be required nearly as often. However, in an "On-Air" tape machine, any maintenance that is done less frequently than on a semi-annual basis provides no protection at all against failures between checks.

Summary

These general maintenance procedures are those that can and should be performed on a regular basis. When a good program and system of records is set up and maintained, it will help make the engineer's job much easier and provide time for dealing with the few emergencies that are inevitable outside your tape equipment.

There are, however, a few general considerations that should be mentioned that are not directly involved with the tape machine proper, but will have a direct affect on the performance of the unit. The following is a brief look at some of these considerations.

Tape and Tape Cartridges

One of the major keys to good performance is consistency in all areas, tape and tape cartridges is no exception. Once a certain type of tape and one certain brand of tape cartridge has been chosen, stay with them. Optimize the machines to the type of tape and use it throughout the system.

Test Tapes and Alignment Tools

Test tapes are an essential part of the equipment needed for proper care of tape machines. The manufacturer usually recommends a certain test tape for their machine; however, if you have more than one brand of cart machine or reel-to-reel, by all means use only one test tape, and not a separate one for each brand of machine.

The care of test tapes is important, and a few basic rules should be followed. Aside from the normal care considerations given when the tape is not being used, *always* demagnetize all heads and tape path components before playing the tape. Also insure that no stray bias or dc voltage is present on the heads before using. The normal useful life of a test tape with proper care should be several hundred passes.

Aside from normal hand tools, some manufacturers recommend certain head, motor, and guidance alignment tools. These tools can be useful when used properly; however, remember that again consistency is the key.

Conclusion

In wrapping up this section on tape machine maintenance, there are two key factors to remember. First, be diligent in the maintenance program and most problems will never get onto the air or into a final production. Second, be consistent with procedures and all machines will perform the same in the news room, the control room, the remote van, or the production room.

Following these tips should give you the maximum life and performance out of every tape machine.

5.12

Microphones

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TRANSDUCER TYPES

Any discussion of microphones should begin with some understanding of what a microphone is and what it does. A microphone is a transducer, a device which, when activated by energy from one system, converts that energy to another form and supplies it to another system. In the microphone, acoustical energy (sound waves impinging on the diaphragm) is converted to an ac voltage. The method by which the microphone converts acoustical energy to electrical energy is one of the ways by which microphones are differentiated.

Carbon

The earliest transducer principle used to construct a microphone was variable resistance. This is the principle by which the carbon element operates. The carbon microphone contains carbon granules which are compressed and decompressed by the movement of the microphone's diaphragm as it responds to air pressure changes. As direct current is passed through the granules, the changes in density result in a variable resistance to the current flow. The resistance change bears the characteristics of both the amplitude and frequency of the sound waves that act upon the diaphragm.

While carbon microphones offer excellent durability, they do require a power supply. More serious deficiencies include their high distortion and limited frequency response range. Their common application in recent years has been in telephones and other communications areas, such as military and aviation communications. Even in these fields, carbon microphones are regularly being replaced by dynamic and condenser systems.

Dynamic Ribbon

The ribbon, or velocity microphone, utilizes a very thin, corrugated metallic foil ribbon suspended within the flux field of a permanent magnet (Fig. 1). The ribbon is referred to as a velocity microphone because it responds to air particle velocity at the ribbon. The term "pressure gradient" is also used to denote that the pressure exerted on the ribbon is in proportion to the difference between the pressures present on each side. While the ribbon's ends are held in place, the rest of it is allowed to move freely back and forth in a sympathy-induced mechanical recreation of the amplitude and frequency of the sound presented it. The ribbon's movement causes it to cut the lines of flux between the magnet's poles, inducing a small ac voltage onto the ribbon. The leads from the ribbon's ends connect it to a step-up transformer which converts the extremely low impedance of the ribbon (approximately 1 ohm) to a usable figure which might lie between 50 and 500 ohms.

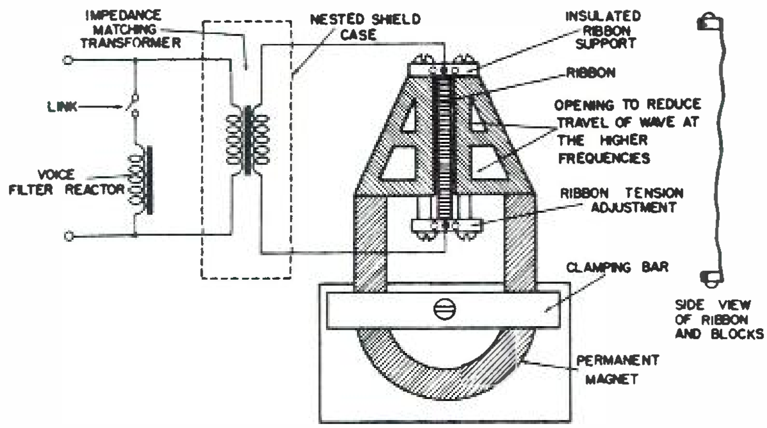


Fig. 1. Construction of a ribbon, or velocity, microphone. (Courtesy of Audio Cyclopedia)

While the inherent directional characteristic of the ribbon microphone is bi-directional ("figure-eight" polar pattern), other patterns have been derived to allow much more flexibility in their application.

Ribbon microphones have historically been noted not only for their potential for delivering a very "warm" sound, but for their susceptibility to damage from air blasts. Blowing across the ribbon, coughing into it, or subjecting the ribbon microphone to the wind in location recording could easily stretch or break the ribbon. Even rapid panning on a studio boom has caused ribbon failure. Newer designs, however, have provided considerable improvements in durability and a lower failure rate.

Dynamic Moving Coil

Although the ribbon microphone is a type of dynamic microphone, transducers of this type are usually referred to by the terms "ribbon" or "velocity microphones". In common usage, the "dynamic microphone" has come to mean a moving coil dynamic microphone.

The dynamic microphone has a diaphragm to the back of which is attached a voice coil (Fig. 2). This extremely light-weight coil of wire is suspended in a magnetic field supplied by a permanent magnet structure. The ends of this voice coil are brought out to stronger leads which connect to either a transformer or the microphone's output connector.

Sound waves acting upon the diaphragm cause it to move back and forth in sympathy. The attached voice coil is then forced to cut the lines of flux in the magnetic field causing a small ac voltage to appear at the ends of the leads. This signal should closely emulate the sound waves in terms of frequency and amplitude. The dynamic element is sometimes referred to as a generator or motor mechanism. It is, by design, very similar to a speaker. Loudspeakers are often used as both the speaker and the mike in an intercom.

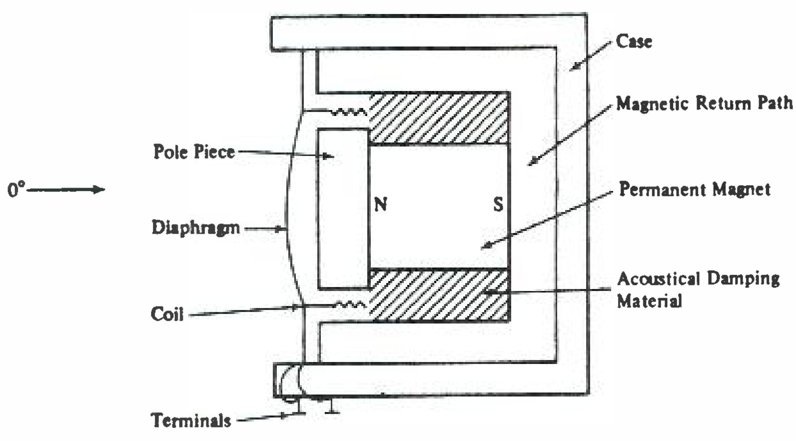


Fig. 2. Dynamic moving coil element. (Courtesy of Audio Cyclopedia)

The diaphragm design is crucial to good dynamic microphone performance. It must be highly compliant to allow effortless excursion at all frequencies of interest. In addition, this movement must be accomplished with maximum linearity and a minimum of break-up modes. Nonlinearities of diaphragm movement would include, for example, any tendency for a rocking motion to be set up which, at some frequencies, would allow one side of the voice coil to be travelling downward while the other side is moving upward. Such rocking results in phase cancellation and a dip in the frequency response.

Break-up modes occur when a portion of the diaphragm resonates independently of the rest of the surface. Again, phase cancellation results; and, with it, response anomalies occur.

The design and construction of a high quality dynamic microphone suitable for broadcast use is a blending of science and art. As with other transducers, much of its design may be modeled by the use of equivalent electrical circuits. The process is similar to the computer assisted design of loudspeakers using Thiel parameters. In the microphone, however, the seemingly infinite variety of complexities in certain parts of its design is so great that the artistry of the microphone engineer is constantly called upon.

In the design process, the goals set before the microphone design engineer are sometimes all but impossible to achieve within the same product. As is true in other areas of engineering, design trade-offs are numerous and the laws of physics tend to win in the end.

Dynamic Microphones: Effects of Small Size

Size plays an important role in the performance capabilities of the dynamic microphone. Small dynamic motor mechanisms tend to be very inefficient. Their acoustic sensitivity is low while their mechanical sensitivity is high by comparison. The result may be a poor system S/N ratio and a lot of handling or stand-born noise. Internal shockmount systems may be used to reduce the mechanical excitation but the design goal of small size may then be defeated.

Small size usually means sacrificed low-frequency response in dynamic microphones. This is not to say that large dynamic microphones will always have an extended low frequency response. Very small dynamics will almost certainly, though, be lacking in low frequency output.

Still another physical characteristic of the dynamic microphone that affects its performance is the mass of the diaphragm/voice coil assembly. The greater the mass, the more limited will be the high frequency response. Common design practice includes the use of Helmholtz resonators immediately in front of the diaphragm. These create peaks and effectively extend high frequency response beyond the normal limits of the system (Fig. 3).

Most broadcast quality dynamic low-Z microphones accomplish their impedance as a function of the turns and gauge of their voice coil wire. Some more public address-oriented older microphones employ a transformer within their housing to correct for design trade-offs in their voice coil. The transformer not only adds to the microphone's cost, however, but may restrict performance, limiting the frequency response limits and increasing distortion. Properly designed dynamic microphones can be the most rugged of the high-quality transducer types; some have truly become legendary for their ability to provide high quality broadcast audio with virtual bullet-proof construction.

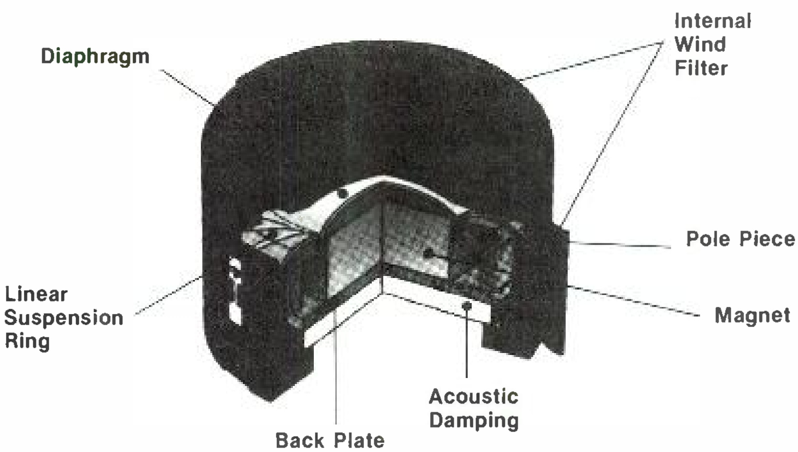


Fig. 3. Cutaway drawing of a dynamic element showing Helmholtz resonators.

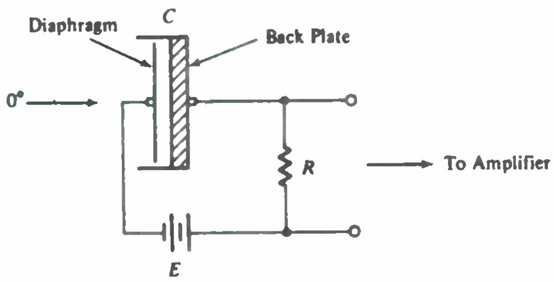


Fig. 4. Conventional capacitor microphone system.

Condenser

In the condenser microphone, a capacitor forms the generating element (Fig. 4). One side of the capacitor is the diaphragm; the other is the fixed backplate. Air between these two plates acts as a dielectric. The capacitor, of course, must possess a positive electrical charge on one and a negative charge on the other of the two plates. The conventional or "discrete" condenser gets this polarizing or bias voltage from an external dc power supply. In older condenser microphones, separate leads were required to deliver dc from the power supply to the microphone. Today, "phantom power" is used in most conventional condenser systems to deliver the required dc voltage to the microphone over the same conductors and their shield that are used to carry the audio signal. Upon activation of the power supply, a positive voltage is quickly built up on the rear surface of the diaphragm. This causes an electrical current to flow through the resistor until the surface of the backplate finally receives a negative charge of equal value.

As sound waves (air pressure changes) strike the diaphragm, causing it to move back and forth, the distance between the two plates rapidly increases and decreases. This causes proportional changes in the capacity of the condenser to hold a charge. The result is an ac current flow in the resistor and a voltage across the resistor that corresponds to the excursion of the diaphragm. While this voltage effectively represents the output voltage of the microphone, the source impedance is far too high to be carried for any distance over microphone cable. This output signal, then, is presented to an impedance converter (often an FET) inside the microphone. Power for the impedance converter is derived from the same source as that which provides the polarizing voltage for the element. The impedance converter delivers a low impedance output that can be fed down long microphone cables with minimal loss.

Electret Condenser

The electret condenser microphone utilizes a material which has the ability to hold a charge applied during the manufacturing process. Most high-quality electrets apply this material to the fixed back plate of the capacitor (Fig. 5). Some designs employ a charged diaphragm instead, but pay several performance penalties in doing so. Lowering the mass of the diaphragm by moving the electret material to the backplate results in lower handling noise, extended frequency response and improved transient response.

The electret functions much like the discrete condenser, but produces its output voltage without the need for an external high-voltage dc supply. An impedance converter is still required. The low voltage needed to power it, however, may be derived from internal or external batteries or an external ac-powered supply.

PHANTOM POWER

Phantom power, or "simplex" power, provides one means for remote powering condenser microphones. Phantom power requirements of various microphones may fall anywhere between 9 and 48 V. External supplies are most often designed to deliver 9, 12, 18, 24, 30 or 48 V. While many electrets will operate on any of these supplies, some modern discrete condenser designs require 48 volts. The phantom supply voltage in non-electret condenser microphones is often stepped-up by an internal circuit to provide a sufficient capacitor polarizing charge for good signal-to-noise figures.

In a phantom power circuit, the plus side of the dc supply is applied equally to both of the signal conducting leads of a balanced microphone line. This may be done by means of either a build-out of matched precision resistors, or via a center-tapped transformer. In each case, the return path

Fig. 5. Cutaway drawing of an electret condenser microphone element.

is the shield. In the microphone, the dc may be similarly tapped via the resistor or center-tapped transformer method to provide the power it needs. The dc is prevented from being seen at the impedance converter output by dc-blocking capacitors or the internal center-tapped transformer.

If a balanced output dynamic microphone is connected to a line with phantom power present, performance should not be altered; nor should damage occur to the dynamic element. The voice coil or output transformer winding connects across the two signal leads and should see no potential difference between them. Because there is no connection between either lead and the shield (the dc return path), there is no circuit. If an unbalanced dynamic microphone is connected to a phantom supply, the dc will pass through the voice coil, causing the microphone's whole life to pass by before it.

A less common powering system called "A-B" or "T" powering is not compatible with dynamic or phantom powered microphones. A-B power puts the positive side of the dc on one signal lead and the negative on the other. This is not a nice thing to do to a dynamic microphone.

FREQUENCY RESPONSE

One of the first specifications looked at, on a microphone data sheet, is the frequency response range or limits. The required low and high-frequency limits may depend upon the nature of the sound source that is being miked, the medium by which the signal is to be stored and transmitted and the environment in which the miking is to be done.

Often, unfortunately, more attention is given to the response limits than to how the microphone actually sounds in its intended application. Non-linearities of the response often contribute more to the listener's subjective impression of the microphone than do the response limits. A specification that reads "Frequency Response 40-18,000 Hz", by itself, says virtually nothing. Add to that some limits, like ± 3 dB, and our knowledge of the microphone, while improved,

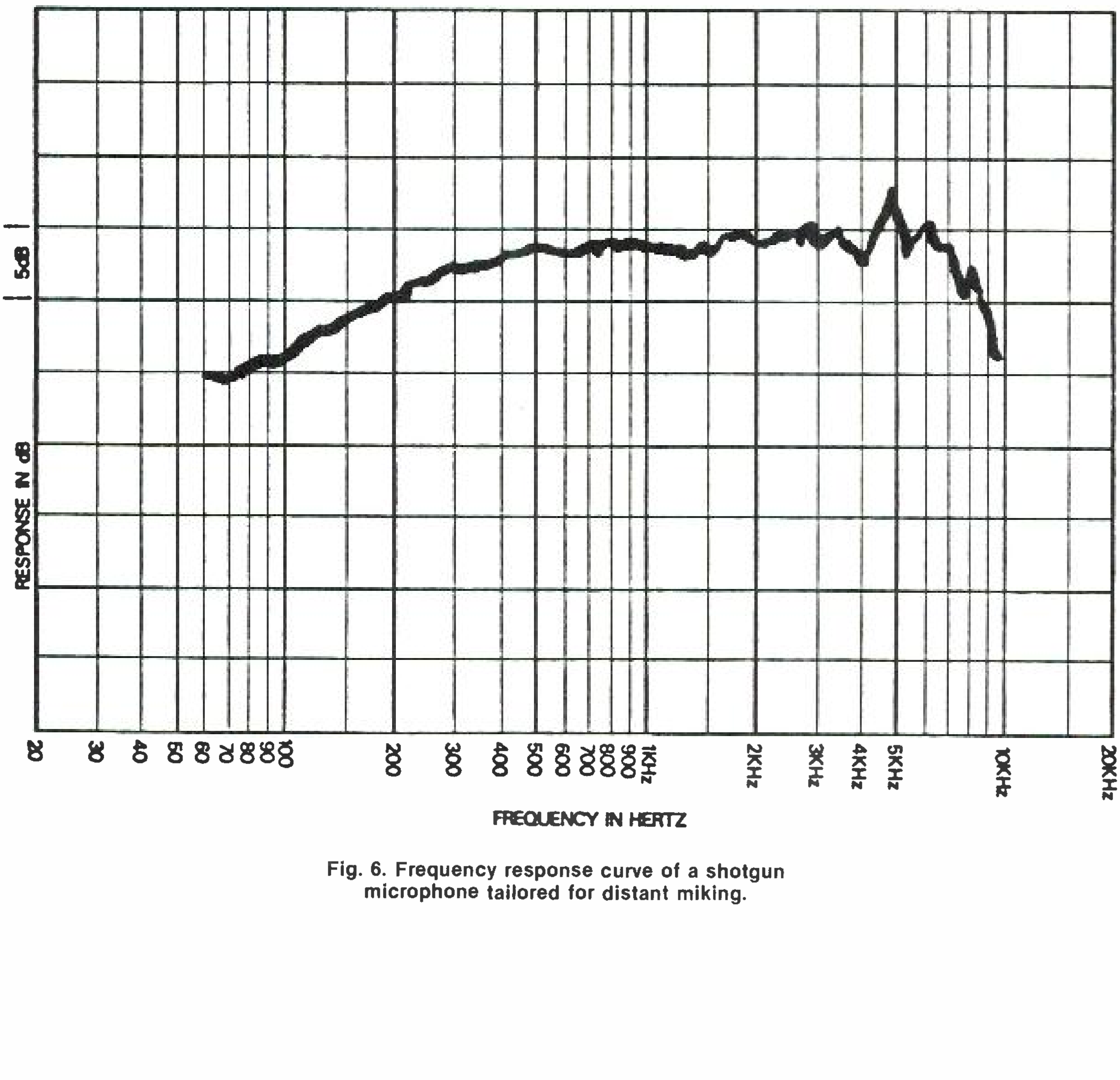


Fig. 6. Frequency response curve of a shotgun microphone tailored for distant miking.

is still extremely scant. The shape of the response will tend to contribute to the character or "personality" of the microphone's sound.

Non-linearities can create acoustic feedback in sound reinforcement, nasality, poor intelligibility, excessive sibilance, muffled sound or any of a variety of other acoustic problems. On the other hand, a microphone's response may be deliberately, carefully tailored by the design engineer to solve problems rather than create them. A rolled-off low frequency response and a rising high frequency response may be employed in a microphone that is intended for use at a considerable distance (Fig. 6). This tailoring can reduce the effects of unwanted low frequency information, such as traffic noise or the rumble of air handling systems, while boosting the high frequencies that are normally attenuated with distance.

Microphones that are intended to be worn on the body usually exhibit a response that compensates for their positioning where the lip and chin "shade" the microphone from the sibilance in speech.

A rolled-off low end response may help considerably in attenuating handling or stand-born noise as well as wind noise or the breath blasts of explosives in speech.

Unfortunately, published specifications can serve as only one guide in our efforts to understand a microphone. The extent to which they serve any function at all is, of course, dependent upon the credibility of the source. This may help to explain why a very poor sounding $10 microphone might show limited specifications that appear superior to an excellent-sounding $100 microphone.

Certainly, a response curve drawn with the microphone directly facing the sound source (on-axis response) will give us a clearer picture of the instrument than will a statement of its response limits. Even curves from well respected manufacturers, however, may be difficult to compare due to the variety of test procedures and standards. For example, the frequency response chart is an X-Y graphic that compares decibel output to frequency. If that chart is compressed vertically or stretched horizontally, the response curve will automatically appear more linear.

Frequency response plots for the same microphone may also vary greatly when run on the same equipment, with the same graph paper. Such variables as the pen speed, damping or even the direction of the tone sweep (low to high or high to low) may result in vastly different curves.

Directional microphones present still another possible anomaly in the testing process. These microphones exhibit a phenomenon known as "proximity effect" which results in a bass boosted output when close-miking a sound source. Proximity effect is neither good nor bad; its value is dependent upon the intended application. Various designs will differ in the amount of proximity effect that is possible to attain. A singular response curve of a directional microphone, tested at some particular (or worse, unknown) distance may be of little value to someone who wishes to use it at another. Ideally, directional microphone data should include close and distant curves (Fig. 7).

POLAR PATTERNS

As difficult as it might seem, at times, to pick up the audio signal that is desired, the real problem more often lies in eliminating those sounds we don't want to pick up. Microphones with various directional patterns are often enlisted to improve the ratio of desired signal to ambient noise.

For most miking applications, a suitable on-axis curve alone may not be satisfactory. Ambient noise, "leakage" from other instruments in a band or orchestra, room reverberation and feedback potential from PA floor monitors are some of the reasons why it is important to know what the off-axis response of the microphone is, as well. The best view of the microphone's off-axis response is obtained by examining several different polar plots. These should be drawn at low, mid-band, and high frequencies. Overlaid, these plots should reveal how well the microphone maintains its directionality at each frequency.

There are several broad classes of microphones, based on fundamental polar patterns, to which most microphones' directional characteristics conform, to some extent or another. These include:

1. Omnidirectional
2. Cardioid
3. Supercardioid
4. Hypercardioid
5. Bi-directional

The accompanying chart shows the relative data for each of the various patterns (Fig. 9). The data, however, is taken from mathematical models representing the perfect polar in each example. Actual microphone polars may vary from near perfection to close resemblance. In the real world, the microphone design engineer must go beyond the math equations to somehow accomplish a desired axial response and sensitivity with the proper polar pattern while maintaining polar uniformity. It is truly a blending of art and science.

Polar Scaling and Dynamic Range

Care should be taken when reading polar patterns to observe several variables in the way that they may be represented. These variables can

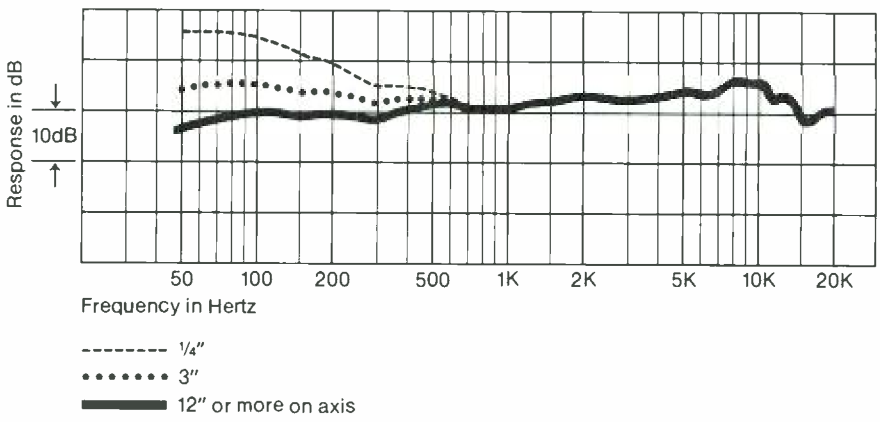
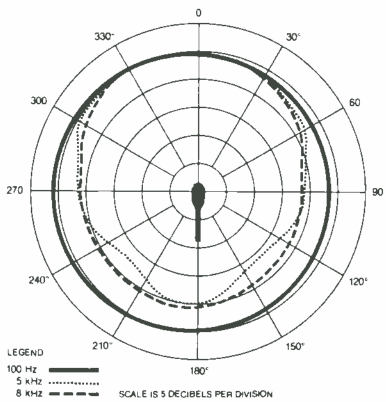
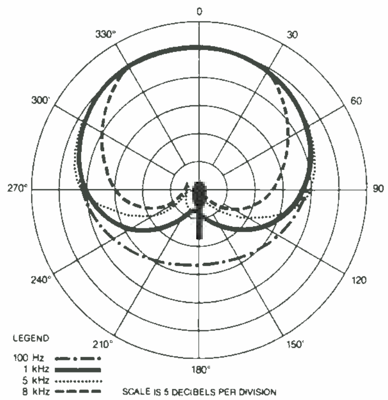


Fig. 7. Influence of proximity effect on a cardioid microphone



Typical Omnidirectional Pattern



Typical Unidirectional (cardioid) Pattern

Fig. 8. Polar patterns drawn at several frequencies.

drastically alter the perception of the mike's directionality if not examined closely.

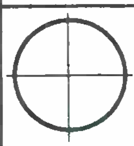
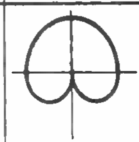
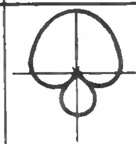
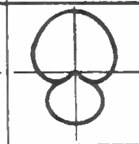
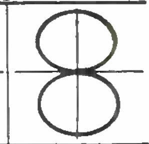
First, check to see whether the scaling is logarithmic or linear. A log scale (the most commonly used) will show a fairly modest inward curve of the cardioid pattern at 90 degrees, indicating a 6 dB drop in level. The linear scale polar for the same microphone will show a polar pattern that appears much more directional. The outside circle of the linear polar represents 100% while the center of the circle equals 0 output. Since a 6 dB loss is equal to a 50% drop in voltage, the polar curve at 90 degrees sweeps in to half the distance between the outside of the circle and its center.

Second, determine the graduations between concentric circles. Are the darker lines 5 dB or 10 dB apart? Finally, take note of the "dynamic range" of the polar pattern. This may be determined by counting graduation lines inward from the point where the polar crosses 0 degrees, in 5 or 10 dB steps (as marked) to the smallest inner circle. Polars may be found in most any range, with 25, 30, 40 db all being common. These differences will also alter the shape of a polar pattern.

Keep in mind that the polar pattern represents a cross-sectional, two dimensional diagram of a three dimensional function. The 131 degree arc, for example, that is described by the 3 dB down points on either side of the axis of the cardioid mike can really best be thought of as a conical area within which the mike is virtually uniformly sensitive. This area is often referred to as the mike's "angle of acceptance" or "included angle".

Omni Observations

The omnidirectional microphone is the easiest type to make. It consists of a diaphragm and gen-

CHARACTERISTIC	OMNI-DIRECTIONAL	CARDIOID	SUPER-CARDIOID	HYPER-CARDIOID	BIDIRECTIONAL
Polar response pattern					
Polar equation	1	.5 + .5 cos θ	.375 + .625 cos θ	.25 + .75 cos θ	cos θ
Pickup ARC 3 dB down (1)	—	131°	115°	105°	90°
Pickup ARC 6dB down	—	180°	156°	141°	120°
Relative output at 90° dB	0	−6	−8.6	−12	−∞
Relative output at 180° dB	0	−∞	−11.7	−6	0
Angle at which output = 0	—	180°	126°	110°	90°
Random energy efficiency (REE)	1 0dB	.333 −4.8dB	.268 −5.7dB (2)	.250 −6.0dB (3)	.333 −4.8dB
Distance factor (DF)	1	1.7	1.9	2	1.7

NOTE:

1 = Drawn shaded on polar pattern
2 = Maximum front-to-total random energy efficiency for a first order cardioid
3 = Minimum random energy efficiency for a first order cardioid

Fig. 9. Microphone polar patterns and their characteristics.

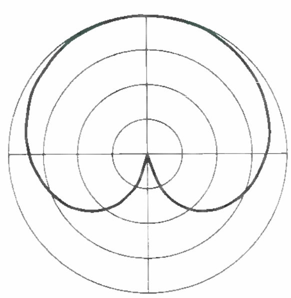


Fig. 10A. Cardioid log scale polar. Scale is 10 dB per division.

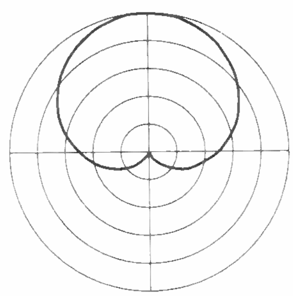


Fig. 10B. Cardioid linear scale polar. Scale is 10 dB per division.

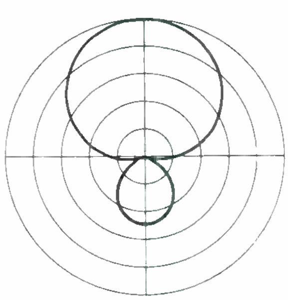


Fig. 11A. Hypercardioid linear scale polar. Scale is 10 dB per division. Dynamic range is 50 dB.

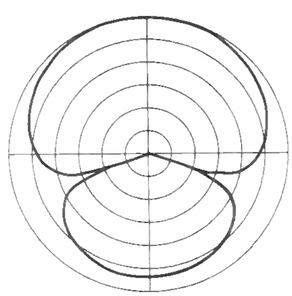


Fig. 11B. Hypercardioid log scale polar. Scale is 10 dB per division. Dynamic range is 40 dB.

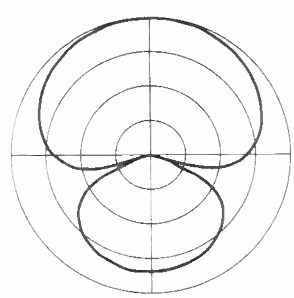


Fig. 11C. Hypercardioid log scale polar. Scale is 10 dB per division. Dynamic range is 60 dB.

erating element backed by a totally sealed case. When placed in a sound pressure field, the perfect omni disregards the direction of the sound's origin. A positive pressure (air expanding) at the diaphragm, for example, causes the diaphragm to move inward regardless of the sound's point of origin (Fig. 12). Such a microphone may also be referred to as a "pressure microphone."

So it would be with the perfect omni. Most omnidirectional microphones, though, are not truly omnidirectional. The case of the mike represents a barrier to the higher frequencies arriving from off-axis. So, due to this "case effect," most omnis are increasingly directional as the frequencies get higher. Likewise, the smaller the omni, the more truly omnidirectional it may be. In addition to case effect, energy arriving at the diaphragm from on-axis is reinforced at those frequencies to which the size of this baffle area is significant. This "baffle effect" causes a rise in the microphone's high frequency output, but only with respect to energy arriving on axis.

Cardioid Comments

The chart (Fig. 9) may be used to get some idea of how the various patterns should relate to the reference omni with respect to their ability to reject unwanted energy arriving from various points off-axis. A sound source which delivers 60 dB sound pressure level (SPL) to a cardioid on-axis from one foot away will "sound to the microphone" as if it has dropped in level to 54 dB, or been moved to two feet away, when the microphone is simply rotated to position the sound source at 90 degrees off-axis. Here a properly designed cardioid emulates well its mathematical model. At 180 degrees off-axis, however, the cardioid can't live up to its equation. The chart indicates 0 output. In reality, well designed cardioids are capable of significant cancellation at 180 degrees, but typically, more on the order of 20 dB or better.

Still, a 20 dB drop in the level of some unwanted noise may be enjoyed simply by turning the cardioid to face directly away from the noise. That's the equivalent of moving the sound source to ten times its actual distance from the microphone. Not bad.

The 180 degree response curves ("back curve") of many cardioid microphones show their tendency to more closely resemble omnis at both the low and high frequencies. The much more impressive cancellation in the midrange only is just the kind of temptation that sometimes causes a manufacturer to release a data sheet that shows one polar pattern only, and that at some unknown frequency.

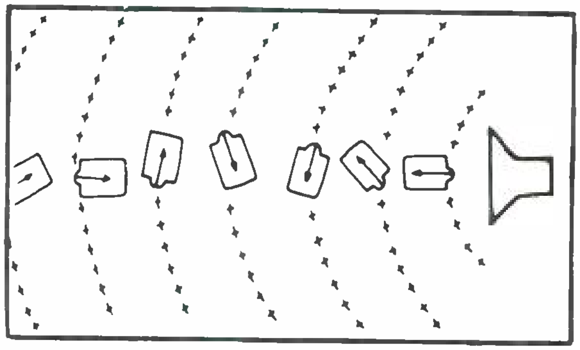
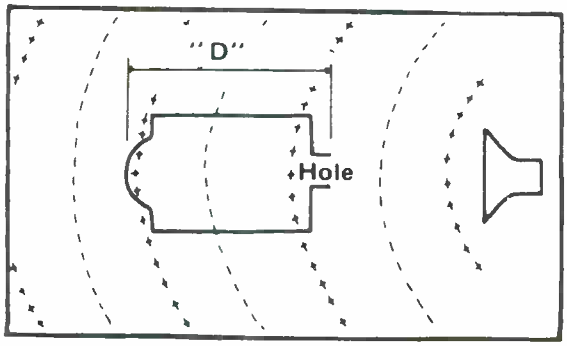
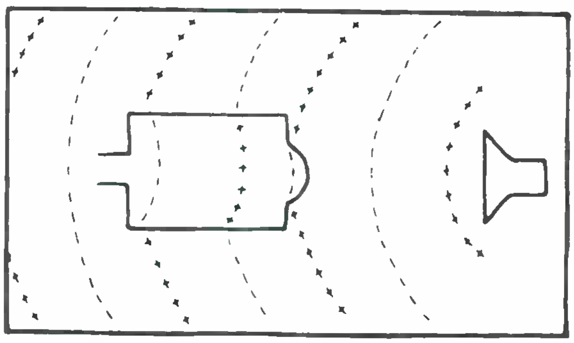


Fig. 12. Omnidirectional microphone principle.



SOURCE AT REAR



SOURCE AT FRONT

Fig. 13. Single-D cardioid microphone operating principle.

Omni versus Cardioid

Compared to the omnidirectional microphone, the cardioid has several characteristics that should be noted:

1. The cardioid reduces the pick-up of ambient noise and reverberant energy. A look at the comparison chart shows that the "random energy efficiency" (RE) of the cardioid is .333 as compared to a RE of 1 for the omni. The random energy efficiency is a measurement that compares a microphone's sensitivity to random, or reverb-

erant energy to its on-axis sensitivity. While this shows the cardioid to be one third as sensitive to random ambient noise as the omni, remember that discreet sound sources positioned at the null of the cardioid will be attenuated to a much greater extent. This will prove true outdoors where sounds arrive at the microphone directly with minimal reflections. Indoors, the advantage of the cardioid's deep rear null is only appreciated when the microphone is situated within "critical distance" of the offending sound source. Up to critical distance, the direct sound is greater in intensity than the reflected energy; after that point, the two remain approximately equal.

2. The cardioid is more susceptible to the problems of "P-pop" (the blast of "plosives" such as "P" and "T" in speech), wind noise and handling or mechanical noise.

3. The cardioid is more complex to design and construct. Due to this fact, expect to pay considerably more, sometimes, for a cardioid microphone than for an omni of seemingly equal audio quality. The cardioid's more complex construction results in the omni often being the more rugged of the two as well.

4. The cardioid offers greater immunity to feedback in most sound reinforcement applications.

5. The cardioid increases the effective "working distance." The comparison chart lists a "distance factor" (DF) of 1.7 for the cardioid. This means that a cardioid microphone has a working distance advantage over the omni of 1.7:1. An ideal cardioid may be used at a distance of 1.7 times that of the omni for a given ratio of desired on-axis signal to ambient noise. A non-linear back curve will, of course, reduce the microphone's advantages in terms of working distance, gain before feedback, and rejection of ambient or reverberant energy.

6. The cardioid exhibits proximity effect. While some designs are quite low in proximity effect, all exhibit some of this bass-boost phenomenon when used close. This may be considered an enhancement in many close-miking applications. Care should be taken, however, to avoid input overload or loss of intelligibility that may result from excessive proximity effect.

Other Patterns

An examination of the chart will quickly show how the three other polars relate to the omni and the cardioid. The hyper-cardioid, for example, combines a tight acceptance angle with superior side rejection and offers the lowest RE. The result is a good pattern for distant miking such as is often required on a boom mount. Most short shotgun microphones approximate a hyper-cardioid pattern.

Notice that the bidirectional pattern offers the best side rejection but with no advantage over the cardioid in RE. Bidirectional microphones are typically ribbons or dual-diaphragm condensers. They are excellent for the pick-up of two sound sources in music or dialogue miking with no phase problems.

Compared to the omni, each of the directional patterns shares all six of the characteristics described above.

LIMITING THE WORKING DISTANCE

Sometimes it is not enough to reduce ambient noise merely through the choice of a polar pattern that offers the lowest RE. In very noisy environments such as may be encountered in an aircraft, on a battlefield, in a factory, or at a sporting event, it may be desirable to differentiate between close sound (an announcer, for example) and distant sound.

Those microphones which offer considerable proximity effect may be used to advantage in these situations. Close-mike the talent and roll-off the low end as needed to flatten the response.

In extreme situations, a noise-cancelling (differential) microphone may be required. The design of the rear ports and back damping systems of the differential microphone causes sound which originates in very close proximity to the front of the diaphragm to arrive in phase at the rear of the diaphragm, allowing diaphragm movement and maximum output. Sound arriving from a distance, however, strikes both sides of the diaphragm with equal intensity and out-of-phase, causing the signal to be cancelled. The noise-cancelling microphone is able then to differentiate between close and distant sound sources. The audio quality of such systems normally limits them to voice communications applications.

Inverse Square Law

The easiest, and certainly the cheapest, way to limit the apparent working distance of a microphone is by simply positioning the microphone very close to the sound source. Inverse square law tells us that every time we decrease the distance between the microphone and the sound source by one half we increase the SPL from the source at the microphone by 6 dB (Fig. 14). Because of the logarithmic nature of this phenomenon, the ambient noise level from more distant sources remains rather constant. As the input sensitivity control of the mixer or recorder is lowered to compensate for the additional 6 dB available from

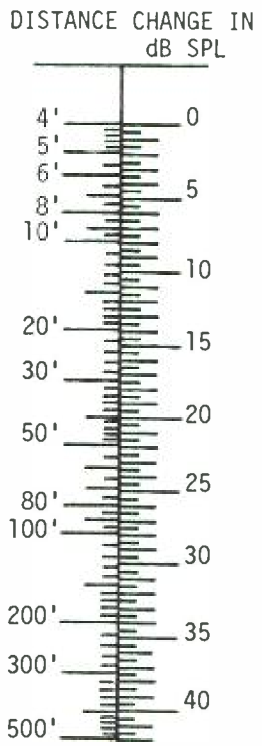


Fig. 14. Inverse square law.

the now-closer sound source, the microphone, in effect, becomes less sensitive to distant sounds.

Headset microphones

Headset microphones benefit from always remaining at a set distance from the talker's mouth. Background noise is reduced due to the use of inverse square law. Omni, cardioid and differential elements are available on headset systems. Cardioids offer the best combination of ambient noise suppression and acceptable broadcast quality.

GOING FOR THE DISTANCE

Shotgun microphones

Effective working distances beyond those afforded by cardioid, supercardioid or hypercardioid systems may be realized through the use of a shotgun or "line" microphone. The line microphone uses a slotted "interference tube" ahead of the element to provide a high degree of cancellation at the sides. Sound waves arriving on-axis are virtually unaffected by the tube. Sound arriving from slightly off-axis, however, is forced to turn and travel down the tube to the element. This results in numerous out-of-phase conditions being set up in the tube, with cancellation increasing as the microphone is rotated to 90 degrees. The length of the tube will determine the low frequency limit to which this cancellation can be effective. Below this limit, it is up to the capsule to offer the directional pattern for the shotgun microphone. All things being equal, longer tubes should maintain their cancellation at lower frequencies than short tubes.

Shotgun microphones tend to be much less uniform in their off-axis response than simpler, more conventional designs (Fig. 15). Even with their multi-lobed polar patterns, however, their very narrow acceptance angle can often save the day.

Shotguns work best outdoors and in controlled acoustic environments such as well designed studios. They do not function properly near reflective surfaces. Distant miking down a hallway will not be greatly assisted by the use of a shotgun microphone. A shotgun microphone pointed toward an open window will "see" all sounds originating on the other side of that window as if they are coming from a source the size of the window. No directional advantage will be realized other than to reduce, perhaps, the level of some nearby off-axis sounds originating from the microphone side of the window.

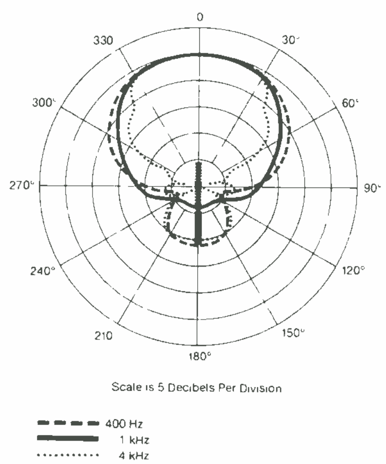


Fig. 15. Shotgun polars.

Acoustic Gain Devices

Shotgun microphones increase working distance by throwing away off-axis energy, thereby narrowing the acceptance angle. Some devices actually increase working distance by providing acoustical gain. Increased acoustical gain means less electrical gain is required and, therefore, less noise. Providing that a satisfactory signal-to-ambient noise ratio may be obtained, the improved signal-to-electrical noise ratio derived from the use of an acoustical gain device will in-

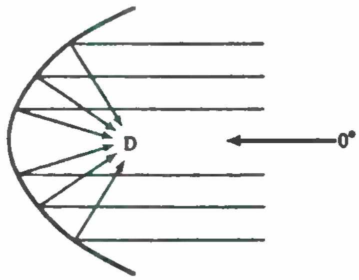


Fig. 16. Parabola principle.

crease the effective working distance of the overall system.

The most commonly used of the acoustic gain devices is the parabola or parabolic reflector (Fig. 16). The parabolic reflector is shaped so that sound is reflected onto a "focal point" a short distance in front of the center-point of the "dish". An omnidirectional microphone placed at this point receives multiple reflected sound waves in-phase which add together to produce significant gain. The response of such systems is very ragged and limited. Low frequency response is limited by the dish diameter. While totally unacceptable for most broadcast applications, the audio quality achieved with the parabolic microphone is often deemed adequate for sound effects pick-up such as at sporting events. In some situations, "acceptable" quality means the best level of performance possible through practical or affordable means.

A second type of acoustic gain device is the horn. Low cost re-entrant paging horns are often used for "talk-back" in paging systems. They can be quite directional and are extremely sensitive. Their audio quality as a microphone, however, is no better than as a loudspeaker. Horns are sometimes used as microphones in surveillance applications where natural sound is secondary to sensitivity and intelligibility. Installed on the side of a building, the small horn is virtually as inconspicuous as a light fixture and is seldom thought of as a microphone. Some horns are built with 45 ohm voice coils, providing higher output to microphone inputs.

Acoustic gain is also realized by using a microphone in the very-near vicinity of a large hard, reflective surface. Omnidirectional microphones may be flush-mounted into the barrier, facing out. In this position the microphone is in a "half-space" environment, or looking into only half the world. The output, for sound arriving on-axis, is increased by 6 dB. As the sound source is rotated off-axis, however, the microphone output drops. At 90 degrees off-axis, the output is down 6 dB, or equal to the omni in free space. The resulting polar pattern resembles a cross-section of a cardioid cut through the microphone at 90 degrees.

See the section on Acoustic Phase Interference later in this chapter for further details regarding surface mounting.

Frequency Response and Distant Miking

Distant miking may result in noticeable, or even dramatic, changes in the spectrum of the sound being recorded. High frequencies, attenuated by the air, may have to be boosted to restore a normal sound. Similarly, high-pass (low-cut) filters may prove helpful in reducing low frequency room reverberation or background noise, thereby extending the useful working distance. If possible, any boost in EQ should be done in the recording process to minimize noise that would be amplified in post-production equalization. Microphones with rising high frequency response or a rolled-off low end will reduce the need for EQ.

SENSITIVITY RATINGS

Certainly, one of the greatest sources for confusion on a microphone specification sheet has to be the sensitivity rating. Several different rating systems are currently in use.

Here are some figures to keep in mind when trying to interpret any of the ratings:

$$
\begin{aligned}
10 \text{ dynes/cm}^2 &= +\ 94\ dB\ SPL \\
&= 10 \text{ microbar} \\
&= 1 \text{ pascal} \\
&= 1 \text{ newton}/cm^2
\end{aligned}
$$

Open Circuit Output Voltage

Microphones are often specified to have a particular output voltage when looking into an open circuit. In most modern equipment, microphone inputs are at least 10 times the measured impedance of the microphone and may be regarded as an open circuit. Specifications may be given as an actual output voltage or as decibels below one volt at a sound pressure level of 74 dB (1 dyne/cm^2 or .1 Pa). These ratings are referred to as the "open circuit output voltage rating" or "open circuit sensitivity."

The open circuit sensitivity may be expressed in dB by means of the following formula:

$$V_{OC} = 20 \log E_O - \text{SPL} + 74 \qquad [1]$$

Where: V_{OC} = Open circuit voltage in *dB* (ref 1 *v*/0.1 *Pa*)
E_O = Microphone output in volts
SPL = Actual sound pressure level (in dB) at microphone

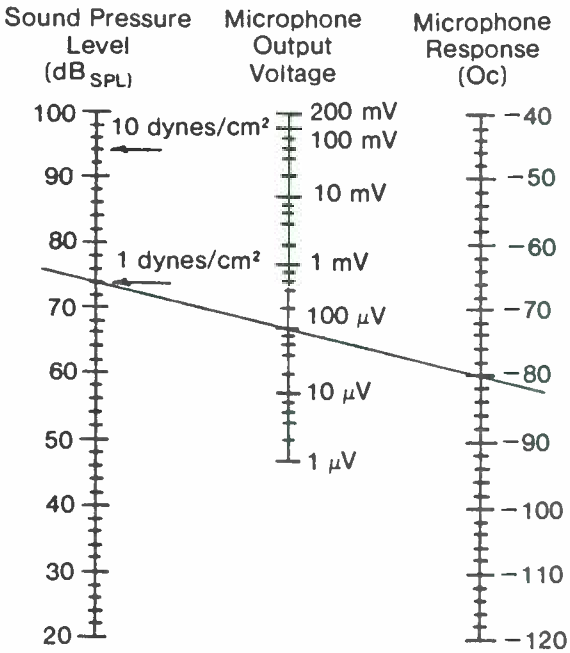


Fig. 17. Nomograph: open circuit voltage rating.

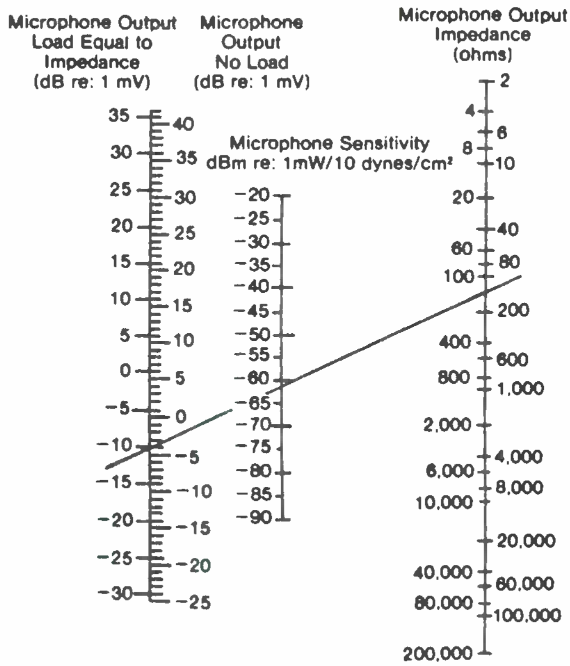


Fig. 18. Nomograph: equivalent power rating.

Power Level

A microphone's sensitivity may also be specified in terms of its output power level. This "equivalent power level rating" takes into consideration the open circuit rating and the actual impedance of the microphone. Specifications given would be in dBm (or just dB) referenced to 0 dB = 1 mW/10 dynes/cm² or 0 dB = 1 mW/Pa.

Calculating the power level rating may be done this way:

$$P_E = V_{OC} - 10 \log_{10} Z + 44\ dB \qquad [2]$$

Where: P_E = Equivalent power level
Z = Impedence of the microphone

EIA Sensitivity Rating

This is one of those "sometimes specified, hardly ever used" specifications. The formula for determining EIA sensitivity is as follows:

$$ESR = V_{OC} - 10 \log_{10} R_{MR} - 50\ dB \qquad [3]$$

Where: ESR = EIA sensitivity rating
V_{OC} = Open circuit voltage in dB
R_{MR} = Center value of the nominal impedance range

For R_{MR} you may use the following table:

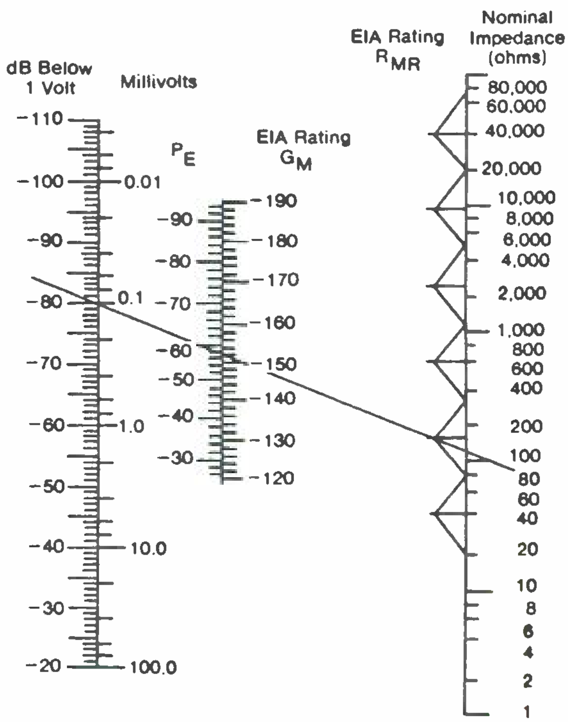


Fig. 19. Nomograph: EIA sensitivity rating.

RANGE (In Ohms)	CENTER VALUE
20-80	38
80-300	150
300-1250	600
1250-4500	2400
4500-20000	9600
20000-70000	40000

Line-Level Microphones

The line-level microphone incorporates a microphone, preamplifier and power supply into one hand-held package. The system may also include a limiter. Line-level may be required to get the signal above the level of induced noise (see section on Hum Rejection), to overcome resistive losses in long cable runs or merely to level-match to a system that does not provide a microphone preamp at its input. A line level microphone is sometimes preferred over a separate microphone and preamp for simplicity of operation.

Applications may include operation into extremely long lines, into unshielded twisted-pair cable, phone lines, or into line-level inputs such as those often found on microwave transmitters. (Also see 6.1-3 on Telco connection.)

OUTPUT IMPEDANCE

The *impedance* (Z) of a microphone is a measurement of its ac resistance looking back into the transducer. Broadcast microphones are virtually all low impedance, ranging typically from 50 to 600 ohms. Dynamic moving coil microphones may achieve their low impedance by either a low-Z voice coil winding or a transformer. Condenser microphones use an impedance converter circuit to step-down the capacitor's high-Z output.

Low impedance offers the advantages of low susceptibility to hum and electrical noise pick-up and the ability to run relatively long lines with minimal level or high-frequency loss.

Unlike matching power amplifier impedances to speaker systems which may be desired for best power transfer, microphones like to see load impedances on the order of 10 times their internal impedance or even higher. This assures maximum voltage transfer. A microphone that looks like a resistive source of 150 ohms, looking into a load resistor of 150 ohms, for example, will suffer a 6 dB voltage drop as compared to an open-circuit connection.

DYNAMIC RANGE

The difference between a microphone's own "self noise" and the maximum SPL it can handle is its dynamic range. In field applications, ambient noise usually provides sufficient masking to make the self-noise spec of minor interest. The importance of this specification increases, of course, as greater working distances are demanded or ambient noise levels are lowered.

The impedance converter of condenser microphones, like any active circuitry, will create some noise. The degree to which it does so will vary greatly from one design to another. The impedance converter design also determines the "headroom," or maximum SPL that the condenser microphone can handle. A maximum SPL of as much as 141 dB is achieved in several high quality condenser microphones. Such levels may never be encountered by lavalieres worn at a news desk or seldom by shotgun microphones that are more often attempting to pick up weak audio signals.

Dynamic microphones contribute virtually no self noise. When greatly amplified, only the noise of the thermal agitation of air molecules is detected. While this is very low in level, the dynamic microphone does not automatically win the award for first choice in a low-noise system. Because the output level of the dynamic is often lower than that of a condenser system, the user may end up working into the noise floor of the preamplifier in order to provide sufficient system gain.

Properly designed dynamic microphones are practically impossible to drive to audible distortion. The distortion heard when a dynamic microphone is subjected to the lips-touching proximity of a blow-your-brains-out rock and roll vocalist is the clipping of the electronics following the microphone. Outputs of a volt or more may actually be delivered under such abuse as rock and roll or hog calls.

Preamp or amplifier clipping may be avoided through padding the microphone output or adjusting the "trim" (gain adjustment) of the mixer. Be aware that many mixers offer adjustment only after a gain stage or transformer, either of which may then distort before any control is possible. In-line attenuators, or "pads", are commercially available that allow the selection of 10, 20 or 30 dB of attenuation. These plug directly into the microphone line. They will not pass phantom power.

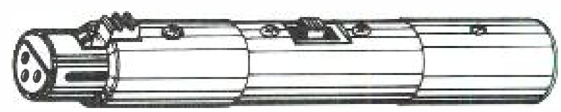


Fig. 20. Microphone in-line attenuator.

DURABILITY AND RELIABILITY

Modern day design and manufacturing techniques have provided the broadcaster with a variety of very durable microphone types. There are a few concepts that generally apply to a microphone design's potential for durability:

1. The moving coil dynamic system should offer more inherent durability than condenser or ribbon designs, although both of the latter systems have been employed recently in prod-

ucts that exhibit impressive improvements in durability.

2. Omnidirectional microphones are more easily built to be very durable than are more complex directional designs.

3. Internally shock-mounted microphones may present a compromise in durability relative to rigidly mounted, nested designs.

Reliability is influenced by a variety of factors including the original design, construction and quality control as well as the care that is taken in the use, transport, and storage of the microphone. Further discussion of this may be found in the last section of this chapter.

VOIDING AND AVOIDING NOISE PROBLEMS

This chapter has dealt already with improving the ratio of desired signal-to-ambient noise. Other unwanted signals include wind noise, "P-pop", mechanical or handling noise, ac hum and RFI (Radio Frequency Interference). The reduction or elimination of each of these can be handled through both proper microphone design and user technique.

RFI problems can usually be traced to a point in the low level circuitry at which the signal leads are unbalanced, Hi-Z or both. Condenser microphones, for example, may sometimes be sensitive to RFI at or around their impedance converter. In such an event, the manufacturer should be consulted for low-pass modifications or information.

P-pops may be reduced through several approaches:

1. Use an omni if possible. Directional microphones are much more prone to P-pop problems than are omnidirectional ones.

2. Position the microphone out of the "ground-zero" area of the breath blast. In an announce application, speak on-axis but at a slight angle to the diaphragm. Stand-ups and hand-held interview miking should be done with the microphone capsule below the axis of the mouth.

3. Use a "pop filter". This is often the same part as the manufacturer's windscreen. Avoid using foam filters that are not designed for the particular microphone model. If such a requirement arises, test the combination carefully for frequency response and directional characteristics before putting it into service. Windscreen/pop-filter foam is specially designed, reticulated foam which comes in a variety of densities. Even very acceptable open-cell foam may be too thick for use on some microphones. Non-reticulated foam (such as "nerf balls") will roll-off high frequency response and alter the polar patterns of directional microphones.

4. Fashion a pop filter. In an emergency, for radio and other off-camera miking, a piece of fabric such as silk may be suspended a short distance in front of the microphone diaphragm. This is often accomplished by using a wire ring for support. The fabric may be stretched loosely over the ring and the ring attached to an arm from the microphone or its support.

5. Use a high-pass filter. Most of the disturbing explosive energy of a P-pop is very low in frequency. Try using a very abrupt high-pass (low-cut) filter in the microphone line. Rolling this energy off before it gets to the board or recorder input will further serve to reduce harmonic distortion at higher frequencies.

Wind noise may be dealt with similarly to P-pop:

1. Omnis are preferred for low wind noise.

2. Use a properly designed windscreen. Most are made of reticulated foam. Superior results with shotgun microphones can be attained by using a well-engineered fabric/mesh cylindrical screen which provides an air space between the material and the microphone (Fig. 21). To handle severe cases (gale-force winds) special socks are available to wrap around the tubular windscreen. While this will result in some per-

Fig. 21. Zeppelin-type windscreen on shotgun microphone.

formance trade-offs, recordings made under such conditions are typically not intended for critical listening. Windscreens must cover all openings to the element: front and rear.

3. Use a high-pass filter.
4. A microphone with a limited low frequency response will help minimize wind noise. Extended response condenser microphones can produce very high outputs of infrasonic energy when panned or when air around them is moved by air handling systems, etc. The result may be preamp overload or undesirable compressor or limiter action. Again, windscreens and/or high-pass filters may solve these problems.

HANDLING OR MECHANICAL NOISE

The problem of mechanical, non-acoustic, noise is one that plagues the user, whether the microphone is hand-held or hardware mounted. The reduction of a transducer's sensitivity to such noise, or the improvement of its acoustic to mechanical-noise sensitivity ratio, starts with the basic element design.

First, omnidirectional microphones are lower in mechanical noise than comparable directional systems. Second, condenser microphones, because of their inherently lower diaphragm mass, are superior in this respect to dynamics. Also, as the size of a dynamic system is reduced, such as in dynamic lavalieres or some small hand-held systems, the acoustic sensitivity tends to go down while the mechanical-noise sensitivity may remain fairly constant.

When the resonant frequency of the microphone's mechanical sensitivity is low, a high-pass filter may be used.

Microphone elements are, of course, often internally shock-mounted by the manufacturer to avoid the transfer of noise from the case to the element. Lowest noise is achieved through the combining of omni or even omni condenser systems with internal shock mounts.

External shock mounts are often employed in stand or boom-mounted microphone applications. Properly designed shock mounts allow excursion on-axis, or perpendicular to the diaphragm plane. Excellent mechanical isolation may result from using an external shock mount on an internally shock-mounted microphone. Provided that the resonances of the two systems are dissimilar, they will stagger, increasing the effective isolation.

Another method of reducing mechanical noise is to raise the resonant frequency of the mechanical drive system. An example would be that of

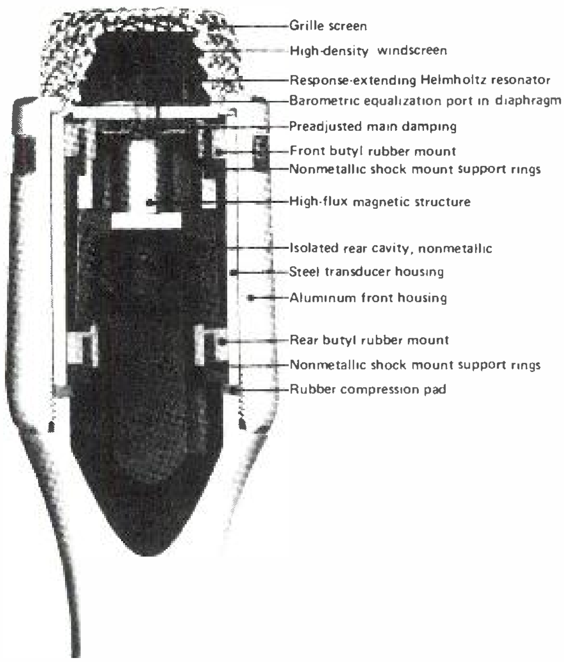


Fig. 22. Cutaway of shock-mounted omni.

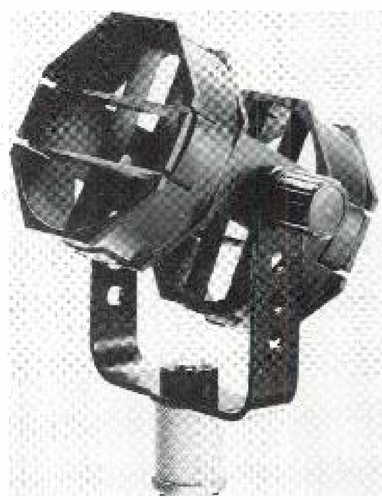


Fig. 23. An external shock mount.

bracing wooden platforms, tables or lecterns to eliminate the very audible "drummy" sound produced when struck. The use of very high-density materials for microphone support systems will result in a higher resonant frequency. A mike stand set onto concrete or into sand gains the advantage of this density.

Mechanical noise transfer to the diaphragm may also be reduced through decoupling the diaphragm from tensile forces, converting these to lateral forces. This may be illustrated as follows:

Select a microphone that has some noticeable handling noise problems and plug it into a talk-out system, raising the gain until the handling noise is evident. Now hold the microphone face up (diaphragm horizontal) with the cable hanging straight down and tap on the cable. You should hear a low thump. Next rotate the microphone 90 degrees so that the cable is hanging at a right angle to the microphone axis. Tap the cable again and the thump should be all but gone.

The mechanical drive that takes place longitudinally on the cable is no longer able to easily drive the diaphragm in the direction of its compliance. Handling noise output, therefore is greatly reduced. This principal may be applied to such areas as your own hardware designs and the dressing of cables as they enter stand or boom-mounted microphones. A loop of cable or a small coiled cord lowers mechanical noise transfer by this method. Direct interconnection of boom cabling and a shock-mounted microphone can create a direct mechanical short between the two, by-passing much of the shock mount's effectiveness. A coiled cord may be employed between these two points to eliminate this direct path.

AC HUM REJECTION

Microphones may also be sensitive to noise induced in them and their cables by electromagnetic or electrostatic radiation. This may be the result of their proximity to power transformers, fluorescent ballasts, high-voltage ac lines, SCR dimmers, etc. The following points should be considered in attempting to avoid or eliminate induced ac hum:

1. Insure that lines are balanced low impedance. The higher the impedance of the microphone, the greater will be the voltage of the electrostatically induced hum. The balanced line insures that nearly equal hum will be induced on each conductor. Little differential is seen, then, at the amplifier input to amplify, resulting in "common mode rejection" of the hum. Pins 2 and 3 of the 3-pin microphone connector should carry the signal, with pin 2 most often "high." This would mean that positive pressure on the diaphragm produces a positive voltage on pin 2. The shield connects to pin 1 (ground).
2. Route cables with caution. If possible, do not run low-level cables near high-voltage lines, avoiding especially long parallel runs of adjacent microphone and ac power cables. When such cables must cross, they should do so at right angles.
3. Use twisted pair cable. Leads should be twisted inside the microphone and out. The virtually identical positioning this provides them within any hum field and the fact that they are out-of-phase with each other will further insure electrostatic hum cancellation.
4. Use well shielded cable. Installed cable may utilize a foil shield as flexibility and low flex "memory" is not a factor. Good stage and field cable, though, normally has a braided shield. Some of the cables which offer the best combinations of flexibility and good shielding utilize conductive cloth or conductive vinyl under the braided shield.
5. Follow good grounding practices, avoiding ground loops.
6. In general, dynamic microphones are much more apt to be sensitive to induced hum than are condenser microphones. The voice coil can be a very effective inductor. "Hum-buck coils" are employed in some designs which lower electromagnetic hum sensitivity by about 20 dB. The hum-buck coil is wrapped around the outside of the motor mechanism and wired in series with the voice coil but out-of-phase. When both coils are placed into an electromagnetic field, equal energy is induced onto each. Because they are out-of-phase with each other, the offending signal is cancelled.
7. Transformers in microphones should be avoided if electromagnetic hum is a possible problem. Some transformers are constructed with hum-bucking characteristics, however, to greatly reduce their hum induction potential.
8. In severe problem situations, operation at line level rather than microphone level may be required.

ACOUSTIC PHASE INTERFERENCE

Another miking problem relates not to sounds that are added to the output, such as popping or hum, but to portions of the spectrum that are greatly attenuated. This change in the microphone's apparent frequency response is due to acoustic phase cancellation. Acoustic phase cancellation may occur when two or more microphones are mixed, or even when a single microphone is subjected to an overdose of reflected sound. Figs. 24A-24E show the severe phase cancellation problems that can result from several typical miking situations.

Although sound arriving at each microphone is identical, originating at the same source, it arrives at the two microphones by paths of varying lengths. This causes a difference in the arrival times and results in phase cancellation of certain

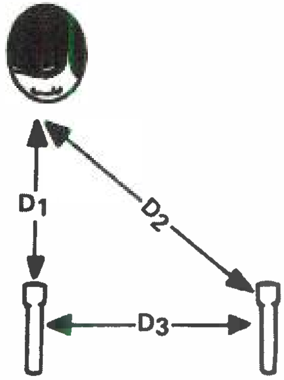


Fig. 24A.

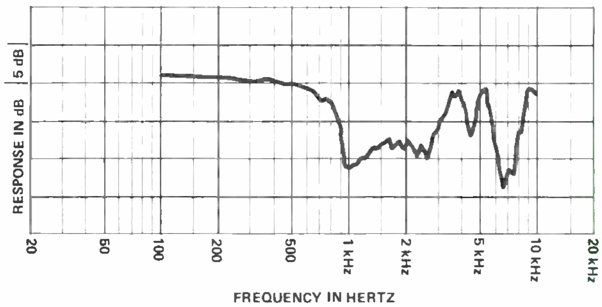


Fig. 24B. $D_1 = 12''$, $D_2 = 21.6''$, $D_3 = 18''$

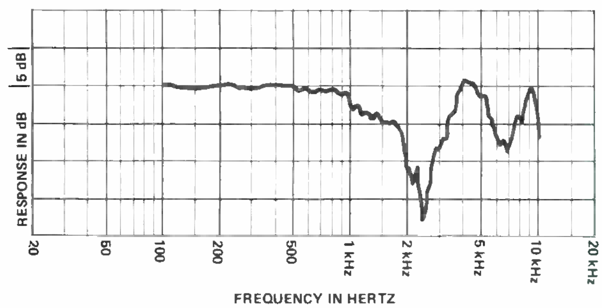


Fig. 24C. $D_1 = 18''$, $D_2 = 21.6''$, $D_3 = 12''$

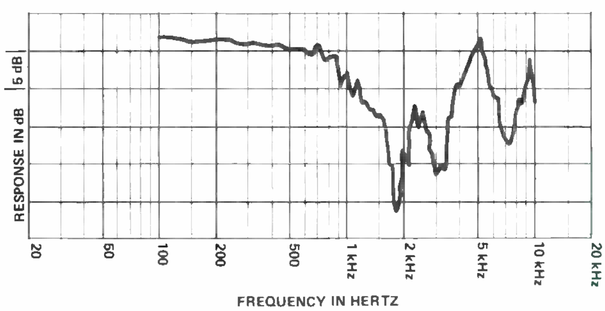


Fig. 24D. $D_1 = 24''$, $D_2 = 30''$, $D_3 = 18''$

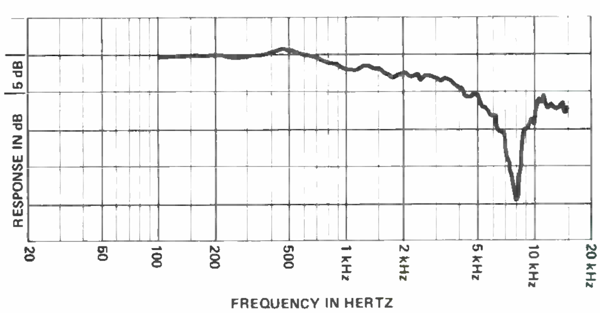


Fig. 24E. $D_1 = 5.6'$, $D_2 = 6'$, $D_3 = 2'$

frequencies. The curves given for each of these examples are FFT (Fast Fourier Transform) derived displays of the actual frequency response of the two microphones combined, with respect to a sound source positioned as shown. The FFT analyzer and its companion microprocessor were used to compare the combined output of two matched, calibrated microphones to the output of one of the two microphones by itself. If no phase cancellation occurred, no trace variations would appear on the X/Y plot, as its plot would be a straight line.

It doesn't take much study of the response charts to show that, no matter which way we have oriented the microphones and our sound source, the summed response of the two mikes is awful. These experiments reveal quite graphically what the ear often perceives as a comb-filter or notch-filter effect that sweeps up and down in frequency, and even changes Q, as the variables *D1*, *D2* and *D3* change with the movement of the microphones or sound source. In more subjective terms, the resultant sound may be described as hollow, as if the sound is being forced through an empty cardboard tube.

Unfortunately, situations that cause acoustic phase cancellation arise quite frequently. One classic example occurs on a podium with a pair of microphones, spaced apart to provide on-mike coverage as the speaker turns his head to address all of the audience in front of him. The curves shown in figures 24B and 24D are typical of the problems caused by this approach. If the output of these two microphones is summed and fed simultaneously to a house sound system, there probably will be gain-before-feedback problems, as well. Add the insistent creeping oscillation of a system on the brink of exceeding unity gain to the already intolerable frequency response of the microphone pair, and you'll have the quality of sound that makes the audience check the credits to see who handled the audio.

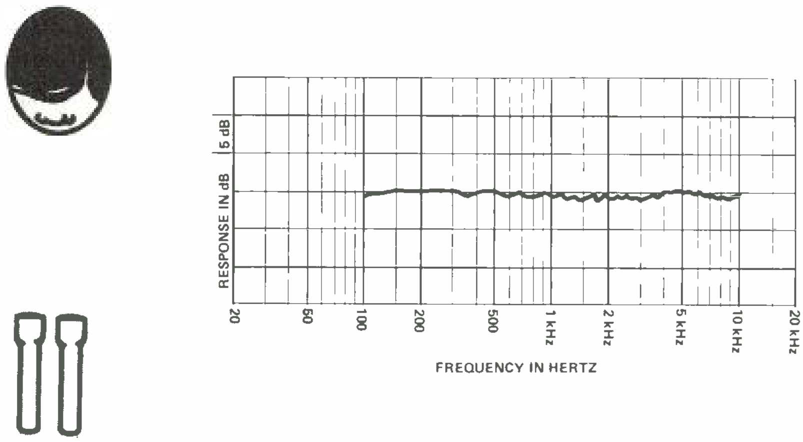


Fig. 25. Redundancy miking.

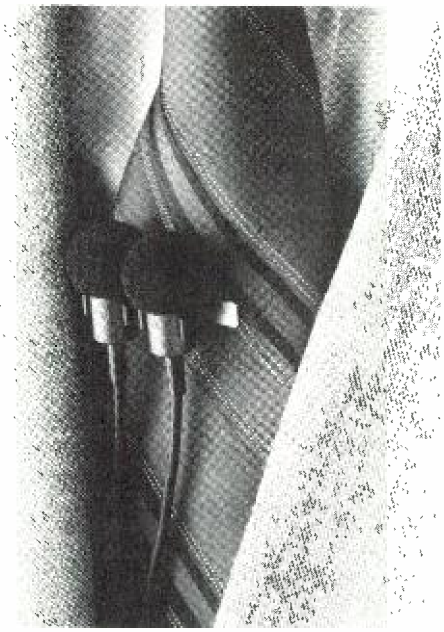


Fig. 25. Redundancy miking.

The simplest solution to the problems caused by this spaced-pair podium miking technique is to use one microphone only, placing it in front of the person speaking and toward the center of the podium. Fig. 25 shows two microphones immediately adjacent to each other. This type of positioning is often employed where redundancy miking is desired for critical applications. Normally, only one of these microphones would be "open" at a time. The second is strictly a back-up system. Sometimes the two mikes are used to feed separate systems, such as for house PA and broadcast. Each may still be used as a back-up for the other.

Adjacent pairs of cardioid microphones may at times be angled inwardly with their axes crossed and their diaphragms closely spaced. This may be done in order to broaden the acceptance angle of the two mikes while still maintaining some cancellation at the rear. The microphones' close proximity allows their diaphragms to occupy nearly the same point in space, thus reducing sonic time path differences. This ensures that negligible phase cancellation will occur should their output be summed. The same formation is often used as a 2-mike stereo pick-up technique, which has the benefit of good mono compatibility.

Obviously, there are many times when the outputs of two or more open microphones must be mixed. How do we then avoid phase cancellation? The phase cancellation problems we have been discussing occur when identical signals, at the same or nearly the same amplitude, are allowed to combine at something other than zero phase angle. We have shown the result of reducing phase angle error by placing the two pickup elements close together. We can also influence the amplitude difference between the two combined out-of-phase signals through careful microphone placement. A good "rule of thumb" to follow in microphone placement is called the "3:1 Ratio Rule."

To eliminate the problems demonstrated in Figures 24A-24E by employing the 3:1 Ratio Rule, *D3* must always be at least three times *D1*. Fig. 26 shows examples of both the violation and enlistment of the 3:1 Ratio Rule. Subjective tests have shown that an amplitude difference of at least 9 dB between the two signals will reduce the phase cancellation to an inaudible level. The 3:1 Ratio Rule is a means by which this 9 dB minimum difference may be approximated quickly in most multiple microphone set-ups.

The amplitude variance desired may also be achieved by judicious use of the mixer's gain or fader controls. Only those microphones in actual use should be opened to their normal operating levels; others should be lowered in level or preferably off. Attentive monitoring on an ac-

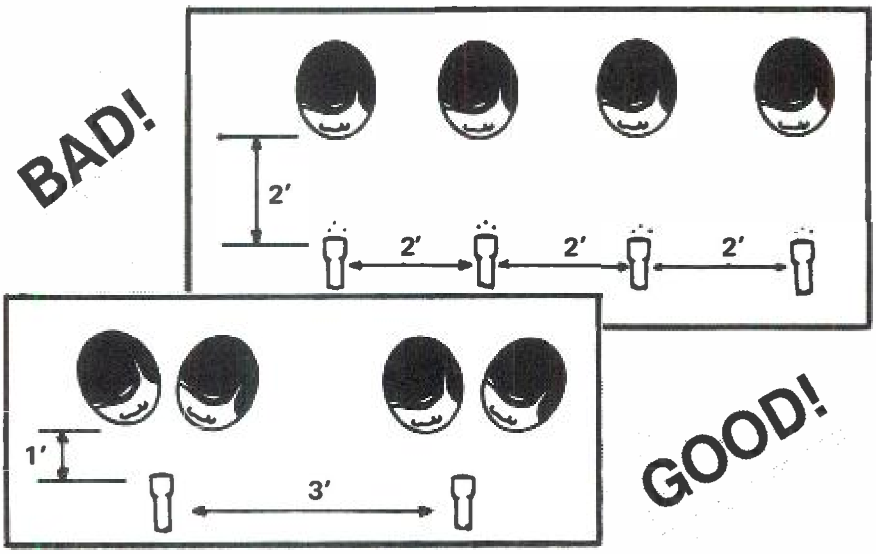


Fig. 26. Violating and obeying the 3:1 ratio rule.

curate control room speaker system will reveal audible phase problems, especially in relatively simple program sources such as speaking voices or most solo instruments.

Acoustic Phase Cancellation with a Single Microphone

Acoustic phase cancellation can also occur in a single microphone system when reflected energy from a nearby barrier such as a music stand, podium, table or floor is introduced at the microphone's diaphragm at a sound pressure level that is within 9 dB of the direct sound. Such problems may be avoided by four different means:

1. Increase the reflected path length.
2. Shorten the direct sound path length.
3. Reduce the reflectivity of the barrier. It may be possible to construct or cover the barrier with an acoustically absorptive material or construct the barrier of an acoustically transparent material. The latter may be used for opaque screens used in chroma keying to eliminate, for example, reflections into a weather person's lavaliere.
4. Position the microphone so close to the barrier surface that the direct and reflected sounds arrive at virtually the same time, causing them to be in-phase.

The latter method also results in a higher microphone output because of the additive effect of the two in-phase signals. As was discussed in the section on acoustic gain devices, the microphone is operating in or nearly in a half-space environment. This barrier miking technique may be employed with omnidirectional or directional elements. The omni may be recessed flush into the surface facing out for best performance. Some of the earliest use of the near-barrier technique was with the Electro-Voice Mike Mouse, an acoustically transparent foam holder which positions the mike inside (usually a cardioid) very close to the barrier, with the diaphragm perpendicular to the surface.

The results of barrier miking will vary due to the following influences:

1. The size of the barrier. (The barrier must be large to support low frequency response.)
2. The size of the diaphragm. (Very small diaphragms may be positioned extremely near the barrier, resulting in less high-frequency cancellation.)
3. The reflectivity and resonant characteristics of the barrier.
4. Ambient noise or reverberant energy problems.
5. Reflections from other nearby reflective surfaces.

MICROPHONE POLARITY REVERSAL

Phase cancellation will also occur if the outputs of two microphones, positioned in the same sound field, are combined with their polarities reversed (i.e. pins 2 and 3 are reverse wired at one end of one cable). The frequencies cancelled, and the degree to which cancellation will occur, will depend upon how far apart the microphones are spaced, how closely matched their frequency responses are and the relative levels of the two mixed signals. It should be noted that the terms "phase" and "polarity" do not mean the same thing. Phase refers to a time difference. Having noted this distinction, in common usage, the terms "in-phase" and "out-of-phase" are often used to refer to matters of polarity. Most microphones will be wired to what is sometimes called the "RCA pin count" which is:

PIN #1 = SHIELD
PIN #2 = HIGH
PIN #3 = LOW

This conforms to IEC standard 268-12 and 268-4. Refer also to RS-221 paragraph 3.3 which states that the in-phase terminal should be the red (or other than black) conductor and that the out-of-phase terminal shall be the black conductor. The terms "in-phase terminal" or "high" (pin #2) indicate the terminal that has a positive voltage present when a positive pressure is applied to the microphone diaphragm. While there are commercially available devices which use a pulse generator to check for polarity reversal, microphones and their cables may also be checked by simply bringing them together and summing their output while speaking into them from a foot or so away. Two mikes which are "in-phase" will deliver a higher output under such a test; if they are reversed in polarity, the output should drop noticeably.

Polarity Reversal as a Tool

While inadvertant polarity reversal in a mike line can result in some very bad audio, deliberate polarity reversal is sometimes employed as a problem solver.

Reducing Background Noise

A pair of mikes may be reversed in polarity to reduce the pick up of ambient noise. This technique is sometimes employed with two mikes in fixed locations, such as in a press box at a sporting event. If these mikes are brought together, a noise-cancelling or "differential" microphone is created. The talker must now speak into one of the mikes only, virtually in a lips-touching position. Due to inverse-square law, the amplitude of the voice at that microphone will be much greater than at the other, resulting in reasonable output level. Distant sound will be picked up equally well, however, by both elements and cancelled.

STEREO MIKING

When miking for stereo broadcast, it is of utmost importance to provide high quality stereo audio without compromising mono audio quality. This all-important mono compatibility normally excludes the use of spaced-pair microphone techniques involving omnidirectional or cardioid microphones. Coincident microphones can provide good stereo and with no phase cancellation problems in mono. Spaced microphones depend upon a combination of amplitude and timing (phase) differences to provide stereo separation. They do not sum well for mono as the very phase differences upon which they depend, to a great extent, to provide separation result in multiple comb filter effects in the mono mix.

Mid-Side Coincident Technique

The most useful of the types for stereo broadcasting is the M-S or "Mid-Side" microphone. Dependent only upon amplitude differences for stereo separation and imaging, the M-S microphone provides excellent stereo imaging and maintains total mono compatibility. The M-S microphone is a combination of a "Mid" microphone, which is typically a cardioid, a bidirectional microphone and a matrixing network which combines their outputs. Sound originating from directly on-axis of the M-S mike will be picked up by the M element but not by the S, typical of bidirectional systems. As the sound source moves off the center axis, one side of the bidirectional mike becomes "positive" relative to the other and, therefore, adds its output to the M signal with which it is in-phase. The result is a smooth transition from left to right and total mono integrity.

Three cardioid microphones may also be employed to perform the M-S function (Fig. 27). The outputs of the mikes, with one of the two opposing cardioids' polarity reversed, should be sent to a mixer. There the Mid mike may be

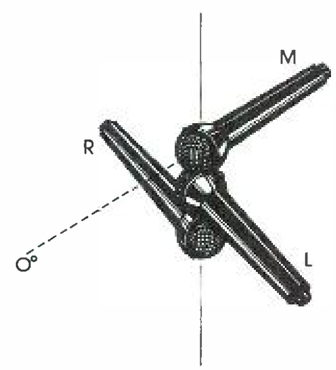


Fig. 27. Three cardioids used as an M·S microphone.

right. The M-S technique offers several control capabilities:

1. Adjusting the relative levels of the M and the S signals will narrow or broaden the perceived stereo image.

2. Panning the M signal off-center may be done to deliberately shift the stereo image. For example, crowd noise at a sporting event may be shifted to appear more closely balanced left and right of a mike position without moving either the mike or several thousand fans.
3. Substituting various patterns, from omni to hypercardioid, for the Mid microphone will affect the apparent mike-to-sound source distance as well as the signal-to ambient or reverberant-noise ratio.

Each of the first two of these controls may be exercised in production or post production: another big bonus.

Blumlein Miking

The Blumlein technique employs crossed bidirectional elements and responds, like the M-S, to amplitude differences to achieve stereo separation. The stereo sound achieved by this approach can be very natural and mono integrity is well maintained. The Blumlein is more sensitive, however, to ambient noise and reverberation than the M-S and placement is critical.

CARE AND FEEDING

Microphones require a certain amount of care in their handling and storage. Here are some basic factors to consider and tips on microphone care. Misuse, or even some attempts to service or clean the microphone, could affect some manufacturers' warranties. When in doubt, ask or return mikes to the manufacturer's recommended service organization for maintenance.

1. Use windscreen or pop filter to protect microphone if it is to be subjected to air-borne contaminates such as dust or smoke.
2. A foam windscreen will also protect a mike from exposure to rain or snow for a surprising period of time. Over time, the cells will fill with water resulting in high-frequency loss and level drop. The foam may be quickly squeezed to reduce the moisture content or a dry screen substituted as required.
3. Foam windscreens will accumulate deposits of dust and other contaminates. The result will be a deterioration of frequency response and, perhaps, even altered polar response. Foam may be cleaned with soap and water. Rinse well to remove all residue. Non-detergent soaps work well.
4. Many mikes may be carefully opened to remove a foam pop filter and sometimes a cloth insert. Do so only in a very clean environment. These filters should be cleaned as above.
5. Avoid allowing dynamic microphones to be set on work benches or other areas where metal particles or metallic dust may be attracted to their internal magnet structures. Very small particles can work their way onto the diaphragm and alter the response greatly. In some cases, the dynamic mike can be opened to reveal the diaphragm for examination. Metallic particles may be *very* carefully removed onto the magnetized tip of a screwdriver. The screwdriver shaft should be steadied on the edge of the mike case and the tip very carefully lowered to attract particles which would likely be held immediately above the voice coil gap.
6. Avoid subjecting electret condenser microphones to high temperatures. This means do not store in the trunk or glove compartment of a car left in the sun on a hot day. Also avoid leaving the electret mike on a boom very close to hot lights. The result may be a loss of charge on the capacitor element and a drop in level.
7. Avoid moisture with all mikes but especially with condenser microphones.
8. If given a choice between using mercury or alkaline batteries to power a microphone, remember that mercury cells die much more suddenly than alkalines. The gradual drop in level with an alkaline can be a life saver; no one likes surprises. Mercury batteries also drop in output level in cold weather. Furthermore, mercury batteries may give off a gas that can corrode the contacts.
9. Avoid unnecessary mechanical shocks. Store in clean padded enclosures.
10. Moving a condenser microphone from a cold environment to a warm one may cause noise problems from condensation.
11. Avoid moisture in cables and connectors, particularly where phantom power is being used.

SUGGESTED READING

For further information on microphones, the author suggests the following sources:

Audio Cyclopedia, Howard M. Tremaine, Howard W. Sams & Co., 1974

Sound Recording, John Eargle, Van Nostrand Reinhold Co., 1976

Microphones: Design and Application, Lou Burroughs, Sagamore Publishing Company, Inc., 1974

Microphones—2nd Edition, Martin Clifford, Tab Books, Inc., 1982

Layman's Guide to Microphone Specifications, James Long, Audio Magazine, August 1969

The Microphone Handbook, John Eargle, Elar Publishing Company, Inc.

The Recording Studio Handbook, John W. Woran, Elbar Publishing Company, Inc., 1982

Microphones, Jon R. Sank, J. Audio Eng. Soc., Vol. 33, No. 7/8, July/August 1985

Modern Sound Reproduction, Harry Olson, Van Nostrand Reinhold Co.

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