RADIO BROADCAST STUDIO EQUIPMENT

The quality of a broadcast station's only distributed product, its sound, is determined by the events taking place in the studio. The best source material and the best air talent will produce only marginal results when burdened by inadequate equipment. Even in television, audio can no longer be treated as a secondary technology. State-of-the-art equipment, properly used in the studio, equates to ratings and translates to revenue for the station.

This article describes the equipment and operation of audio facilities used in radio and television studios. First is an overview of typical studio layout and a discussion of the major components used in an audio facility. This article provides the novice engineer with guidance for the design and construction of a studio suited for the station's needs. We also make the leap to digital audio equipment and investigate the new terms and standards destined to change studio operations forever.

Broadcast studios now take advantage of digital audio source material, digital audio processing equipment, and now digital consoles. A major manufacturer recently introduced a digital studio-transmitter link (STL) system that operates without data compression. Just as compact disks (CDs) killed vinyl disks as the standard source material for radio broadcasters over an alarmingly short period of time, digital broadcast equipment is poised to eclipse analog studio systems. We have the potential to reduce the radio studio to a touch-screen computer operation, but the trend in the industry remains loyal to familiar function and feel. Broadcasters prefer their digital equipment to emulate the friendly analog devices that they have used for decades.

TYPICAL RADIO STUDIO LAYOUTS

Many stations in active markets have suffered as a result of changes in formats and managers and engineers who have left a crazy quilt of analog and digital equipment. The results have handicapped each station's ability to survive in a competitive market. Programming needs, available space, and creativity of former engineers often dictated the design of a studio. Design du jour, accompanied by galloping changes in technology, leaves many studios ripe for redesign and rebuilding.

The station's current format should dictate the main design parameters. The operational concepts will be much different for a music format than for a news/talk operation. It is possible to catalog radio station studios under a few general categories.

Music Formats

Studios built for music formats remain the most common, particularly in smaller markets. The basic configuration consists of an audio console, two or more CD players, and multiple cart players all arranged on a U- or L-shaped desk. Other associated equipment might include audio routing, transmitter remote control, and telephone interface equipment. Careful design places all equipment within arm's reach of the operator. This configuration handles the rather simple programming needs nicely. The announcer on duty mixes and switches audio, reads announcements, plays music from CD players, plays commercials from digital cart players, and switches news and network programming from a satellite receiver.

Some stations transfer music from CD to analog, or digital, cart, making the studio an all-cart operation. In this situation the studio contains four to six cart playback decks and no CD players. Having all program material on cart makes the operation more "idiot-proof." The operator needs only to talk, push plastic boxes into slots, and push buttons.

Going one step further, some stations retrieve their program material from digital audio storage systems. Commercials and music are stored digitally on a hard drive, retrieved when required, converted back to analog audio, and fed to the console. This eliminates the physical handling of any audio storage medium. The operator controls each audio event with keyboard and mouse.

Music formats require positions for one operator and possibly an announcement booth for the newsperson. With the popularity of the zoo format, many large-market stations designed their studios for two or three on-air personalities. One member of the team operates the console, one juggles phone calls and pulls music and spots, and the third handles news. The studio layout varies with the duties handled by each member of the team.

News/Talk Formats

All-news stations serve up information from live in-studio talent and short-duration feeds from a large number of sources. Here the board operator functions as an engineer rather than on-air talent. Small studios orbit the central control room allowing eye contact between the board operator and on-air talent. The console provides more of a switching function than audio mixing. Live news or talk programs dictate a larger number of console inputs than the typical disk-jockey operation. Digital audio storage systems help this format flow smoothly because the operator is busy enough without the mechanics of handling carts.

Technology now allows replacement of the audio console with a computerized audio switching system. Hard-disk audio storage systems use multiple computers linked to a central network server through a local area network (LAN). Each studio retains access to all stored program material. The operator sees the log displayed on the screen and can shift events around, control audio source equipment, adjust levels, and even read copy and tags directly from the screen.

Talk formats require a studio for the show host and the on-air guests. A large round table with the microphone booms

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mounted in the center allows the host and any guests working room plus affords eye contact between them. The show producer operates the mixing console and telephone hybrids in the control room. If not located in an adjacent room, the show's call screener also shares the space in the control room.

Large-market budgets afford lots of labor power and first-class equipment. A visit to a small-market talk show may reveal the show host running all the control room equipment while talking with callers. The station's telephone receptionist works frantically screening calls and shuttling caller names on yellow sticky notes to the frenzied show host.

The Production Studio

A separate studio provides an area for commercial production work and transfer of music to a cart or a digital audio storage system. The production studio provides access to CD players, a digital audio workstation, cart recorders, equalization, patch panels, reel-to-reel recorders, and possibly turntables. The production studio provides more flexibility and handles a wider variety of audio media than the on-air studio.

The 4, 8, and 16 track consoles and multitrack recorders found in major market production facilities often intimidate the typical disk jockey accustomed only to segueing music and commercials. These consoles feature submaster mixing busses, allowing the mix down of multiple tracks of music, voice tracks, and sound effects, producing complex spots and promos. Special effects and equalization, not needed in the main studios, remain standard fare in production.

Production directors are enthusiastic about digital workstations because they allow editing without razor blades and splicing tape. Digital workstations bring to audio production the speed and versatility that word processing brought to typing. Workstations allow editing of individual tracks, a feat impossible with multitrack reel-to-reel tape editing.

Because all material produced in this studio will eventually be played on the air, the quality of the equipment should be equal to, if not better than, that used in the main studio. Smaller-market stations erroneously tend to scrimp on equipment for the production room. Using hand-me-down and cast-off equipment in the production studio hampers the potential success of the station.

TYPICAL TELEVISION STUDIO LAYOUT

Audio for television stations can be challenging because of the need for many types of audio mixes. Multichannel TV sound (MTS) requires a stereo program feed to the transmitter, on-camera talent needs a monitor mix, programs with a studio audience require a mono public address (PA) mix, and talk and news programs must have a mix-minus for telephone hybrids and possibly a mix in a different language for a second audio program (SAP) channel.

The Grand Alliance advanced television (ATV) threatens to make life even more interesting. As the motion picture format 5.1-channel comes to television, the audio engineer must deal with left channel, center channel, right channel, two surround channels, and a low-frequency effects (LFE) channel. The first five audio channels offer full 20 kHz bandwidth. The LFE channel provides response from 3 to 125 Hz. All this arrives as digitally compressed audio in the Dolby AC-3 format.

Audio monitoring of the preview channel is accomplished through the mixer solo function. If a remote truck is involved, an interruptible foldback (IFB) system requires a program audio feed plus audio cues to the talent at the truck. The console to handle all this requires a design specifically for these complex tasks. The TV audio operator mixes audio and nothing else.

An audio routing switcher tied to the video switcher assumes some of the work load. The audio switcher operates in sync with the video source, leaving level correction adjustments to the station's audio processing chain.

STUDIO PLANNING

Technical Basics

Analog audio signals consist of complex mixtures of alternating currents of different frequencies at different powers. In broadcasting, the decibel (dB) serves as one unit for measuring audio power. We measure the power with VU (volume units) meters, marked in decibels, on our equipment. What is a decibel? The decibel is one-tenth of the logarithm of the ratio of two powers.

The VU meter shows a "0" reading, but it is not on the left end of the scale; it's to the right of center. On a VU meter, "0" does not mean zero or none. These volume units are referenced to the 1 mW of power produced by a 1 kHz audio tone across a 600 Ω load. This combination of audio power (1 mW) and load value (600 Ω) creates 0.775 V of audio and is represented by the "0" on the VU meter.

The designation dBm identifies all readings referenced above or below this power level. A signal of −3 dBm contains half of the energy of a signal measured at 0 dBm. A signal of +3 dBm is twice as loud as the 0 dBm signal. A signal of +10 dBm is 100 times as powerful.

Decibel values identified as dBu (decibels-unterminated) indicate an audio voltage of 0.775 V across an open circuit. Decibels measured in dBv (decibels-volts) are also measured across an open circuit but are referenced to 1.0 V. Because the circuit lacks the traditional 600 Ω load, these are simple measurements of an audio voltage, not power. Modern analog equipment no longer uses 600 Ω impedance-matching circuits; most references to decibels now carry the dBu or dBv notation.

Occasionally the 0 dBm output level on a console drives the input meter on a recorder to +4 dBm. 0 dBm still equals 0 dBm, but equipment manufacturers often calibrate their equipment to provide an output level of +4 dBm, or +8 dBm, when the output VU meter indicates 0 dBm. Engineers always standardize operating levels of all equipment in the station. This allows them to patch any output into any input without operating level problems.

Signal levels of 0, +4, and +8 dBm are all valid operating levels. Each station may use a different level, but an engineer will calibrate all the equipment in the station to one of these three "standard" operating levels. Someone bringing in equipment from outside the facility may find it necessary to recalibrate to the signal level used throughout the facility. Digital audio equipment presents another level-matching challenge. Headroom meters, not VU meters, monitor digital inputs. A headroom meter indicates how close the input signal comes to clipping the analog-to-digital converter circuit. A 0 dBm signal fed to a digital recorder with 18 dB of headroom will appear on the headroom meter at −18 dB. You will find headroom meters referencing a decibel value identified as dBfs (decibels referenced to full-scale).

Analog Equipment

An analog VU meter operates like the speedometer in a car, which directly shows the speed at which the car is traveling as a stated value. A headroom speedometer would show how much faster the car can go in relation to the speed limit. In this example, we will assume a speed limit of 60 mph (37 km/h). When the headroom speedometer shows −10 mph (−6 km/h), a conventional speedometer would show a speed of 50 mph (31 km/h). The headroom speedometer displays that there are 10 mph (6 km/h) to go before reaching the 60 mph (37 km/h) speed limit.

Analog circuits tolerate operation with the levels driven in the red above 0 dBm on their VU meters. 0 dBm is not the clip point. An analog circuit clips when the audio signal exceeds the voltage potential of the power supply. A sine wave then flattens on the peaks when the input signal reaches a level higher than the power supply voltage. This condition results in audio distortion.

Typically analog audio equipment clips around +24 dBu. If the manufacturer calibrated the VU meter to an output of +4 dBu ("0" on the meters equals an output of +4 dBu), the equipment is said to have 20 dB of headroom. Input audio reading an average value of 0 dB could contain audio peaks 20 dB higher without clipping and distorting as it passes through the equipment. It is easy to understand why analog equipment with 20 dB of headroom forgives trespasses into the red above 0 dB on the VU meter.

Digital Equipment

Digital equipment inputs demand closer attention. When a digital recorder, with a headroom meter, is driven above "0", the digital audio clips and the recording will contain irreparable distortion, clicks, or pops. Digital audio clips at the analog-to-digital converter (ADC). Digital clipping occurs when the analog input signal drives the ADC past its maximum output capability. The converter is outputting all 1s and can no longer digitally reproduce the rising analog input. Good digital recording practice maintains peaks of −6 dBfs.

Balancing Input and Output

Professional analog broadcast equipment features balanced inputs and outputs. A balanced audio output consists of two wires that carry the analog audio voltage to the next device. Neither of these wires connect to ground. Only the cable shield, which protects the audio from electrical noise and hum, is grounded. A balanced circuit can be run over properly shielded audio cables several hundred feet long.

The 230 V electrical circuit to an electric stove is similar. Two wires carry 230 V between them. Anyone who connects either wire to ground, may not live to tell about the resulting sparks. Although the low voltages associated with balanced audio do not represent any danger, grounding either side because will short-circuit half the audio voltage to ground. Balanced outputs should not be directly connected to unbalanced audio inputs.

When wiring balanced analog audio, care should be taken to connect the high, or plus (+), terminals. Connect the low, or negative (−), terminals only to similarly marked terminals. If 3-pin XLR connectors are used, pin #2 is always high (+). Pin #3 is always low (−). Pin #1 of an XLR connector is ground. When $\frac{1}{4}$-inch tip-ring-sleeve (TRS) plugs are used, the tip is high (+), the ring is low (−), and the sleeve is ground. Failure to follow the rules of polarity will result in out-of-phase audio. Stereo audio, wired out of phase, results in an audio dead spot centered in front of the speakers. Listeners with monaural radios hear only the difference between left channel and right channel if out-of-phase audio is fed to the transmitter.

In 1984, a Baltimore, Maryland, FM station once operated for 2 days with its audio out of phase. It went unnoticed by those listening in stereo; no one at the station detected anything wrong. But a bedside monaural clock radio reproduced nothing but a left minus right signal. Mono listeners assumed that the station was off the air; they heard only high-frequency spitting noises and muffled mumbling.

An unbalanced audio circuit consists of a single conductor and grounded shield. Consumer electronic equipment uses unbalanced audio circuits easily identified by the single-pin, RCA phono plugs found on the connecting cables. An unbalanced circuit carries half the power of a balanced circuit. It is more subject to hum and noise. Unbalanced circuits cannot support long runs of cable for this reason.

An unbalanced circuit is similar to the 115 V wiring in a house. One wire delivers 115 V to the lights, the second wire in the lamp cord is ground. It can carry only half the voltage of a balanced, 230 V circuit. Connecting an unbalanced output directly to a balanced input will not damage the equipment, but the input level will be too low. A matching interface box is needed to convert the unbalanced output to a balanced one and boost the signal level.

Bridging audio input circuits present no load to the source audio. Bridging inputs abandon the traditional 600 Ω, power-matching input circuits found on older equipment. Without a load, no power transfer takes place. These bridging circuits simply transfer an audio voltage from output to input. A bridging input handles a wider variety of input sources than possible when everything terminated with a 600 Ω load. If an output requires a 600 Ω termination, a 620 Ω resistor tied across the input terminals provides a perfect match.

In the days of tube equipment, both inputs and outputs used transformers. Tubes could not drive 600 Ω loads directly; solid state equipment could. Audio equipment manufacturers slandered the transformer with rumors of poor performance as they removed them during the transition to solid state design. Good transformers are expensive; manufacturers looked for ways to cut their costs and the audio transformer became a casualty.

Some applications still require the physical isolation that only a transformer can provide. Any time that audio equipment connects to a phone line, a transformer blocks the 48 V "telco battery" from entering the equipment. Unusually long audio cable runs operate best, with lower noise, when a transformer isolates the equipment on each end. The audio transformer provides best common mode rejection (CMMR) of electrical noise induced into long cable runs. Transformers still provide input termination on many high-end microphone preamplifiers. The transformer provides the required 150 Ω load for the microphone and isolates the preamp from the phantom voltage required to power condenser microphones. Common mode rejection of noise becomes even more important when dealing with the extremely low-level output signals of dynamic microphones.

Step One: Where to Begin

Planning a new studio, or rebuilding an old one, begins with a layout on paper or computer screen of all the required audio sources and feeds. The console inputs offer a good place to start. Working from a list of all possible sources, the engineer assigns them priorities according to how often and how quickly the operator must put them on the air. This determines the number of mixing channels needed and how many switched inputs each mixer requires.

All frequently used audio sources should be assigned to individual console mixers. Keeping input switching and patching required of the operator to a minimum avoids errors and dead air. A console with two or three more inputs and mixers than absolutely necessary provides insurance against obsolescence and frequent studio rewiring jobs.

Accepted engineering practice runs all line-level inputs through patch panels on their way to console inputs. This allows the engineer to reroute special program audio and patching around any problems that may develop. The exception to the rule is microphone-level audio. Directly wiring microphone outputs to the console input terminals remains the best option. Microphone input positions seldom change, and the extra wiring through patch panel jacks invites noise problems. A possible exception is the television studio where the program must allow transfer from set to set.

Part of proper planning for a new console ensures that levels from all sources will be compatible with the input levels required by the console. If not properly matched, the operating positions of the potentiometers will be different for each mixer, making it difficult for the operator to run the board properly. The operator may open the pot a fraction of a turn and drive the meters to the pin or may not get enough gain even with the control fully open. Either situation results in a poor audio mix with possible distortion and noise problems. Normal operation sets rotary attenuators at the 2 o'clock position and slider attenuators at a 70% position.

All studio sources should be adjusted to operate within their normal output range. Interface amplifiers (matching boxes) should be used to boost low-level audio sources to the +4 dB levels required for most consoles. If the source signal overloads the console input, an H-pad will drop it to gain a proper match.

An H-pad is a simple network of five resistors arranged in the shape of an H laid on its side. The resistors convert some of the audio energy into heat, which drops the audio level by a predictable amount. At the same time, the H-pad maintains the impedance of the circuit. Pads of 10 dB or greater also correct impedance mismatches. Figure 1 illustrates the schematic of the resistive H-pad.

Pads are essential when connecting two audio transformers directly together. If an output transformer directly feeds an input transformer, the output trasformer sees a changing inductive load. The frequency response of the system suffers.

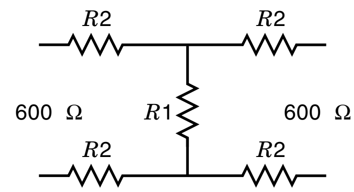


Figure 1. The H-pad resistive network reduces levels and matches impedances in balanced analog audio circuits.

With an H-pad between them, the output transformer sees a linear resistive load. The H-pad may be mounted inside the equipment or at the studio punch blocks. Table 1 shows resistor values for commonly needed 600 Ω H-pads.

The worksheets in Figs. 2 and 3 offer an example of the starting point for designing a studio. Figure 2 illustrates wiring to the console inputs; Fig. 3 designates the wiring path of the console outputs. Preparing customized worksheets for each studio saves false starts and wiring errors. Any computer spreadsheet program accommodates the task easily. Computer drafting programs offer another option for perfecting studio wiring prior to pulling cable.

The studio outlined in these worksheets is one of two in a station. A second studio serves as the production room. Patch panels route all audio. Outputs of both studios appear at the patch panel in the main studio as well as the production studio. If technical problems take the control room off-line, the station can originate the program from the production console.

Buying Equipment

Preparation of a shopping list and cost estimate for new equipment follows the design of the studio on paper. The major components (console, cart players, professional CD players, and furniture) should be chosen early in the planning stage so that budget cuts will not compromise their quality. Price increases, sales tax, and shipping costs should be al-

Table 1. Typical H-Pad Values

Loss (dB)	R1 (Ω)	R2 (Ω)
1	5100	18
3	1800	51
6	820	100
8	560	130
10	430	160
12	330	180
14	240	200
16	200	220
18	150	240
20	120	240
22	100	270
24	75	270
26	62	270
28	47	270
30	39	270
32	30	300
34	24	300
36	18	300
38	15	300
40	12	300

Studio Wiring Plan			
Audio Source	Input Jacks, Patch Panel #1	Output Jacks, Patch Panel #1	Console Inputs
Microphone #1			Mixer 1
Microphone #2			Mixer 2
CD #1	Jacks 1 & 2	Jacks 25 & 26	Mixer 3
CD #2	Jacks 3 & 4	Jacks 27 & 28	Mixer 4
Cart #1	Jacks 5 & 6	Jacks 29 & 30	Mixer 5
Cart #2	Jacks 7 & 8	Jacks 31 & 32	Mixer 6
Cart #3	Jacks 9 & 10	Jacks 33 & 34	Mixer 7
Satellite #1	Jacks 11 & 12	Jacks 35 & 36	Mixer 8
Satellite #2	Jacks 13 & 14	Jacks 37 & 38	Mixer 9
Reel-to-reel	Jacks 15 & 16	Jacks 39 & 40	Mixer 10
Phone hybrid	Jacks 17 & 18	Jacks 41 & 42	Mixer 11
EAS receiver	Jacks 19 & 20	Jacks 43 & 44	Mixer 12
Production studio	Jacks 21 & 22	Jacks 45 & 46	Mixer 13
Spare	Jacks 23 & 24	Jacks 47 & 48	Mixer 14

Figure 2. Careful planning of the console inputs prevents delays and rework during the installation process.

Studio Wiring Plan			
	Input Jack, Patch Panel #2	Output Jack, Patch Panel #2	
Program output	Jacks 1 & 2	Jacks 25 & 26	AGC amplifier input
AGC amplifier out	Jacks 3 & 4	Jacks 27 & 28	Limiter input
Limiter output	Jacks 5 & 6	Jacks 29 ¢ 30	STL transmitter
Spare	Jacks 7 & 8	Jacks 31 & 32	Spare
Audition output	Jacks 9 & 10	Jacks 33 & 34	Reel-to-reel
Spare	Jacks 11 & 12	Jacks 35 & 36	Prod. console in
Mono output	Jacks 13 & 14	Jacks 37 & 38	Office monitor
Mix-minus output	Jacks 15 & 16	Jacks 39 & 40	Telephone hybrid
Spare	Jacks 17 & 18	Jacks 41 & 42	Spare
Prod. console out	Jacks 19 & 20	Jacks 43 & 44	Spare
Spare	Jacks 31 & 22	Jacks 45 & 46	Spare
Spare	Jacks 23 & 24	Jacks 47 & 48	Spare

Figure 3. This example of studio wiring of the console shows the versatility afforded by patch panels. Should the main console fail, patch cords can feed the production studio to the on-air processing and the transmitter.

lowed for, and hidden "handling" or drop-shipment charges should be scrutinized.

Each engineer should maintain a working relationship with a reputable broadcast equipment dealer. Absolute bottom-dollar may not be the best deal. There is no saving in paying $15 less for a CD player that fails to arrive in time to make the on-air date for the new studio. Experienced broadcast equipment salespeople offer their best deals and service to customers with whom they do regular business.

The shopping list should arrive at the dealer at least a week before the cost estimate is needed in order to allow the dealer time to research and work up a quotation. Engineers who demand quotations on short notice seldom get serious attention by dealers. Competition in the broadcast supply business ensures that pricing between reputable dealers will vary by only a few percentage points.

Competitive bids should be limited to two. Time is more valuable than chasing nickels and dimes. If the regular dealer does not offer an item required for the project, ask for a recommendation for a source. A salesperson will know all good suppliers and sometimes offer to get equipment not in the suppliers' normal line for the best customers. This extra service can be worth a lot more than a few dollars when considering the big picture.

AUDIO CONSOLES

Radio Consoles

Centered in the radio studio, in front of the disk jockey, sits the on-air audio console. An 8 to 12 mixer analog console typifies this unit. Smaller mixing consoles find their way into news editing rooms and production studios.

The number of mixing channels limits the number of audio events that can occur simultaneously or in rapid succession. The station format dictates its requirements. Although an operator-assisted easy listening or satellite-based format may be able to use a four or five channel console, it would be out of the question for a fast-paced contemporary or rock program. These smaller consoles may not offer an audition bus or switchable inputs. The lack of multiple switched inputs requires one mixer for each audio source. External switchers can provide extra flexibility when required.

The console found in most control rooms offers ten or more mixers. The most common variety features rotary attenuators. Heavy on-air use begs for step attenuators because they require only occasional cleaning in order to maintain silent operation. These attenuators use make-before-break switches to move through a series of resistive pads. The step between the contacts results in uniform 2 or 3 dB steps throughout the entire range of 20 or more steps. Their rugged construction offers smooth, quiet, dependable operation, but with large size and rotary design as limiting factors. Program audio passes directly through the attenuator, but a switch routes the audio into the console cue bus as the attenuator reaches its fully counterclockwise position.

Wear and build-up of dirt plague attenuators that depend on sliding a contact over a resistive element. The resistive element consists of either carbon, conductive plastic, or metal film. Normal wear changes the element's resistance, and build-up of worn-off carbon may cause erratic resistive changes in the contact between the slider and the element. Noise and uneven tracking between stereo channels results with age.

Noisy pots present a major problem when the program audio routes directly through the potentiometer. In the case of a voltage controlled amplifier (VCA) design, audio does not pass through the control. With the VCA design, only a sample dc control voltage passes through the potentiometer. This voltage controls the gain of an amplifier, which carries the program audio. Figure 4 offers a schematic of a VCA control circuit.

VCA console design solves the problem of audible noise from defective attenuators, but the noisy attenuator will affect the accuracy of the control voltage. The audio passing through the VCA-controlled circuit will become erratic and nonlinear as the noisy control moves through its range. Replacement of the defective attenuator prevents clumsy-sounding crossfades and stereo channel dropouts. This console design allows for the use of cheaper rotary or straight-line (slider) attenuators. As with any amplifier with VCA introduces some thermal noise and distortion. A good console has an overall distortion figure of 0.05% or less. The noise floor should be less than −90 dBm.

Modular design consoles offer the engineer major advantages. Removing and swapping modules make troubleshooting easy. The layout of the console allows easy changes to keep up with station format undulations. The positioning of blank panels created dividers, neatly grouping sources together. Custom panels accommodate special functions such as reel-to-reel recorder control or telephone line selection. Extra space left in the main rack offers expansion with additional mixers as the station's needs grow. These benefits justify the extra investment in a modular console.

For on-air use, a cue channel allows the operator to receive cue audio from remotes and networks and to preview program material. Even if carts deliver all program material, a cue channel is essential. It serves as a valuable troubleshooting tool for the station engineer.

In the early days of radio, all program material was live, and the audition channel served as the cue channel. Levels were set, and program material was previewed using the audition channel. Today the audition channel records network

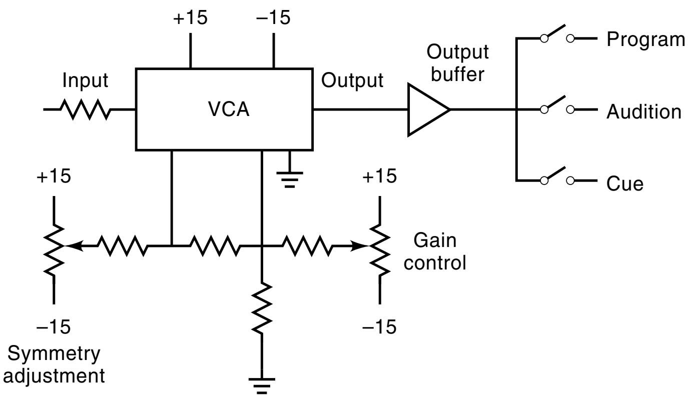


Figure 4. The VCA allows gain control via an adjustable dc voltage. This circuit is typical of those found in VCA-controlled audio mixers.

feeds for later use while the program audio travels the program channels. The audition channel can monitor the audio quality and set levels of remote feeds using the studio monitor system.

Some consoles allow audio from the mixers to feed the program and audition channels simultaneously. This allows recording of on-air programming using the audition channel. In talk formats using fixed-time delay systems, the audition channel mixes the real-time program audio and feeds it to the delay. The output of the delay is brought back in to the console on the program channel, which then feeds the delayed program to the transmitter.

Television Audio Consoles

The requirements of TV audio are compounded because more live audio sources come into play (as opposed to radio, where most program material is recorded). In television installations, additional monitoring requirements mean multiple output mixes. Television consoles include mono input modules for microphones, telephone hybrids, and other mono sources. The inputs offer a gain control or switch to allow stepping between microphone and line-level input. These modules include a pan pot to allow left-right positioning of the apparent audio source. A mode switch may replace the pan pot, enabling selection of normal stereo, left channel only, right channel only, a mono mix, or reversed channels. A cue channel feed plus a solo button permit stereo monitoring of a single audio source in the control room monitors. Audio sweetening requirements make equalization on each module a popular option.

The ability to create multiple audio feeds by using submaster mixing buses represents the major departure from radio consoles. The operator assigns mixer outputs to submaster buses, and these submasters in turn create the master mix. This allows us to create several mix-minus feeds for special monitoring requirements.

Console Features and Options

A growing list of optional equipment available on audio consoles serve both radio and television audio mixing. These make them more user-friendly. Clocks and timers put timing functions in the immediate field of vision of the board operator. The timer resets to zero anytime that a new channel is selected so the operator will know how long a CD or cart has been running.

A mix-minus, program audio minus the caller's voice, feeds console to the telephone hybrid and then down the line to the caller. If program audio, including the caller's voice, were fed to the phone hybrid, a feedback path would exist. Figure 5 illustrates a block diagram of a mix-minus circuit for telephone hybrid. This example demonstrates that not all console inputs must route to the mix-minus bus. Note that only input #1 and input #2 (announcer microphones) connect to program, audition, and mix-minus circuits.

If the studio contains multiple phone hybrids, a mix-minus feed for each hybrid is required so that callers will be able to hear each other's comments. In a television studio, a mix-minus feed provides on-air monitoring for talent on a live set. This feed includes program audio minus the talent's microphone preventing acoustic feedback while allowing the talent to hear program material and cues.

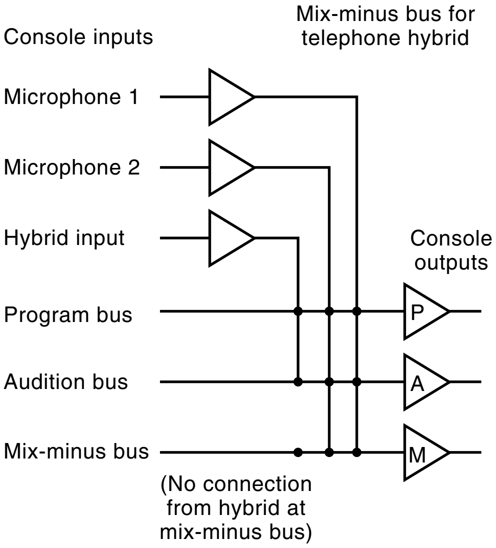


Figure 5. Broadcast consoles used with telephone hybrids require a mix-minus bus to feed the send input of the telephone hybrid. The mix-minus allows the caller to hear the talk show host's voice (Microphone #1 and Microphone #2) but prevents the caller's audio from being fed back into the hybrid.

Consoles designed for radio production and television contain prefader processing patch points. These route the audio source compressors, equalizers, or other signal-processing devices before arriving at the mixing bus. These processing loops provide convenient connection points for microphone processing units.

Monitor amplifiers may be external or built into the console. Power and space limitations restrict built-in monitor amps to less than 10 W. Many engineers prefer to drive studio monitors with external, higher-power amplifiers. The stereo monitor system should contain a stereo/mono switch. This allows the operator to check out-of-phase program material and misaligned tapes. A single-pole, single-throw switch wired between left and right channels at the monitor gain control accomplishes this function. At some stations, the control room monitors routinely operate in the mono mode. This immediately alerts station personnel to out-of-phase conditions.

A well-designed console offers switchable input levels and impedance matching on each input module. This handy feature allows easy transfer of input sources to different mixers, allowing for future changes in the studio. An input amplifier could then be used for either microphone or line-level audio. A second best system would have interchangeable mixer input amplifiers, which an engineer could shuffle between positions in the console mainframe.

Going one step further, some consoles offer programmable presets for input configuration. The engineer first stores preprogrammed console input settings in memory, then tells the console which program is planned for use. All the input sources automatically switch to the proper mixer. Fast and accurate setups result.

Professional consoles feature balanced, bridging inputs. Bridging audio input circuits used in modern consoles handle a wider variety of input sources than possible when everything was designed with 600 Ω, +4 dBm terminations. A bridging input provides a 10 kΩ or greater termination impedance, which provides no load to the source equipment. If an output requires a 600 Ω termination, a 620 Ω resistor tied across the input provides proper matching.

Remote start contacts for cart machines and other program sources became standard equipment in the 1980s. They allow the operator to start the equipment by simply turning on the appropriate channel. Some console manufacturers provide more flexibility by using logic circuits, which allow the mixer to be turned on by pushing the Start button on the cart player. When the cart machine recues, the mixer automatically turns off. Automatic disabling of this logic when the input selector is switched to another input source eliminates the annoyance of having a cart machine start when the mixer is turned on for an auxiliary function.

Console manufacturers offer consoles with a choice of conventional analog VU meters or light-emitting diode (LED) bar-graph metering. LED metering provides multiple color visual monitoring of root-mean-square (rms) audio voltage plus peak values. One model shows left, right, and peaks on a single display. LED displays may make operators less likely to run a board with the meters buried in the red.

Multitrack recorders and digital workstations in the broadcast production environment require consoles with more than a single pair of left and right outputs. Four and eight channel consoles assist in producing award-winning production. Channel assignment switches route the audio to the proper bus. Pan pots then shift it between left and right. Equalizers on each mixer allow adjustments to each audio source. Such production consoles resemble those once found only in recording studios.

Alert engineers realize that digital technology arrived to the audio console industry in the mid-1990s. The basic operating rules still apply to these new boards. The only exception is that they pass along a digital signal rather than analog. As of this writing, many digital consoles remain in the prototype stage.

AUDIO DISTRIBUTION AND ROUTING

Patch Panels

There are three basic types of patch panels, or jack fields. The tip/sleeve $\frac{1}{4}$-inch jack size is the oldest type, consisting of one conductor and one shield. This obsolete design dates back to the early days of radio. The tip/sleeve patch panel requires four single-plug cords to patch a balanced stereo connection.

The tip/ring/sleeve $\frac{1}{4}$-inch panel remains the most popular patch panel for radio. It offers two shielded conductors per cable. A pair of single-plug cords will complete a stereo circuit. Dual-plug cable assemblies allow the convenience of patching a stereo source with a single cable.

The most useful $\frac{1}{4}$-inch patch panel design includes dual rows of 24 jacks. This configuration allows stereo pair spacing. These panels usually group their jacks in pairs with wider spacing between stereo pairs. This spacing technique, when used with a dual-plug patch cord, makes it impossible to cross-patch an audio source. Cross-patching occurs when the user inserts the first single patch cord in one audio feed and the second patch cord in the adjacent audio feed on the patch panel. The dual plug will align only in paired jacks; cross-patching becomes impossible with these types of patch panels and dual plug cords.

Patch panel jack numbering uses the following convention: the jack in the top, left corner is identified as jack #1. Counting across and to the right, the last jack on the top row becomes jack #24. Jack #25 falls below jack #1 and is the first jack on the bottom row, starting on the left. Jack #48 is located on the bottom row at the right end and below jack #24. Although patch panels may contain more, or fewer, than 48 jacks, this numbering system remains the standard for identifying individual jacks.

The $\frac{1}{4}$-inch patch panels also come in single rows of 26 jacks or dual rows of 52 jacks. These panels have standard spacing between all jacks and allow an additional stereo circuit on the 26 jack version and two additional stereo circuits on the 52 jack, dual-row model.

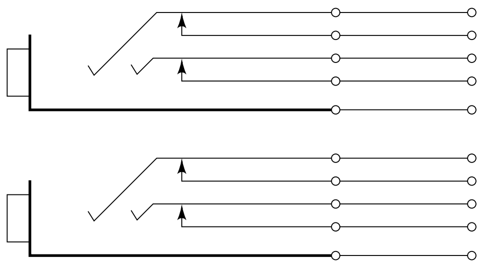
Special configurations of the $\frac{1}{4}$-inch jack field can create very useful designs. One version offers three rows of 26 jacks for a total of 78 jacks on the panel. The wiring scheme of the two lower rows create a conventional dual-row patch panel. Wiring the top row directly to the circuits of the middle row of jacks allows monitoring these equipment output circuits by patching between the top row jacks and the monitor amplifier inputs. Inserting a plug into the top row jacks does not interrupt the normal audio path through the patch panel.

Another custom item is a patch panel with special jacks that not only switch the conductors but also the shield when a patch cord is inserted. The patching of microphone circuits requires this seldom-used configuration.

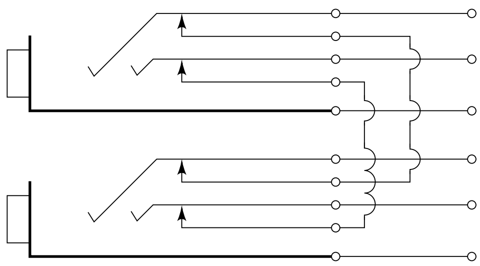
One arrangement features patch panels built into a 19-inch (48 cm) rack mount chassis. The entire assembly mounts into the equipment rack just like the equipment that it connects. The jacks appear on the front of the rack, and the rear termination points on the rear offer easy access to equipment wiring from the back of the rack. There remains a misconception that this design provides protection from radio frequency interference (RFI). However, the phenolic bay fronts provide no shielding of the jacks, and some manufacturers even wire these designs with unshielded wire.

Rapidly making its way out of the recording studio and into broadcasting is the bantam or tiny-telephone jack field. These 0.175-inch diameter plugs and jacks feature the tip/ring/sleeve configuration. The bantam patch panel consumes about half the space that a similar $\frac{1}{4}$-inch jack panel would require in an equipment rack. The 96 jacks fit in a $1\frac{3}{4} \times 19$ inch (4.45×48 cm) rack space. Television facilities discovered bantam jack fields years ago.

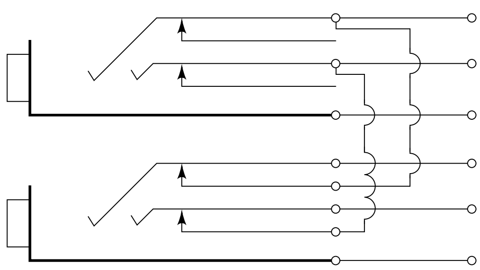
Patch Panel Wiring and Termination. The jacks used in an audio patch panel have, for each circuit, a set of contacts that make contact when no plug is inserted in the jack. The circuit connection opens when the user inserts a plug. This allows for "normaling." An audio source wired to a pair of these jacks passes automatically to the pair of jacks associated with an input that it normally feeds. Proper procedure wires all outputs to the top row of jacks in a dual-row patch panel, and the "normals" connect them to the bottom row of jacks. When not interrupted by the insertion of a patch cord, outputs feed to the proper inputs directly below, which are their normal



Patch panel normals brought out to termination.



Patch panel normals wired at the jacks with a jumper.



Top row bridging, which allows monitoring on the top row of jacks.

Figure 6. Audio patch panels are wired with either (1) all connections brought out to punch blocks, (2) the normals connected by jumpers at the jacks, or (3) the top jacks bridged to allow monitoring of the audio outputs wired to the top row of jacks.

connections. Figure 6 illustrates several methods of wiring patch panel normals.

Short jumpers between the rows usually make the connection for the normals between upper and lower rows of jacks. The jumpers can be brought through rear terminations. If the normals route through a termination, the engineer can determine whether the circuit is normaled or not. Changes in normaling then can be made without removing the patch panel and unsoldering the jumpers.

When situations require the board operator to reroute the output of a cart player normaled to console input #3 to input #6, the operator first inserts a pair of cords in the top row of jacks associated with the cart machine's output. This interrupts the audio path going to mixer input #3 by breaking the normal circuit. The operator then inserts the other ends of the patch cords in the jacks on the lower row associated with inputs for mixer #6. This breaks the normal circuit from the audio device normally feeding mixer #6 and puts the output audio from the cart player into the console input for mixer #6. Inserting a plug into a patch panel with conventional normal wiring breaks the circuit.

If the engineer wants the ability to monitor equipment outputs at the patch panel, half-normaling or top-row bridging wiring design meets the need. Half-normaling wiring connects the jumpers of the top row of jacks directly to their jack arms. The normaling contacts of the top row jacks are not connected. This means that the circuit between the top row jacks and the bottom row jacks is broken only by inserting a plug into the lower row jacks. This allows high-impedance monitoring, or metering, across the circuits without interrupting the audio connection.

Experienced engineers never connect patch panel jacks directly to equipment inputs and outputs. Termination blocks speed patch panel installation and offer flexibility when making wiring changes. Accepted practice mounts patch panel termination blocks in individual equipment racks, in a group in each studio, or in a central point in the engineering area. Short jumpers between connections on the termination blocks complete the links between inputs and outputs. This allows wiring changes at a convenient, easy-to-reach location without pulling new wire between equipment. Terminations used including solder-type "Christmas trees," wire wrap, Type 66 telephone punch blocks or the newer types of punch blocks designed for stranded copper wire.

Christmas trees remain popular with engineers who trust only solder connections. They still serve well in remote trucks because of vibration concerns. Otherwise, they are obsolete.

Most engineers have switched to punch-type terminations. A specifically designed tool "punches" insulated wire into a slotted connector. The wire insulation strips away as the wire pinches into the connection. The process eliminates the soldering task associated with Christmas tree blocks. The "66" block accommodates #22 gauge solid wire used in telephone service. Using only solid #22 wire ensures dependable connections. Stranded wire generally works although the strands may flatten out, preventing dependable removal of the insulation. Some strands may also break in the punch-down process. Solid conductor wire should be used for best results.

Audio Routing Switchers

The audio routing switcher offers an alternative to patch panels. This unit accomplishes the same function by switching the audio with relays or solid state switches rather than by plugs and jacks.

The system eliminates patch cords, can often be operated by remote control, and may often route audio to more than one feed at a time. Some of the more elaborate systems feature computer control. The increased flexibility of a routing switcher provides the only practical solution when many audio sources must be switched frequently such as in a busy TV control room.

Failure of the relays or cross-point switches in routing switchers adds some cause for concern. What happens when there is a power interruption? Latching-type relays hold their connections if power fails. Powering the routing switcher through an uninterruptable power supply presents another choice for fail-safe operation.

Size determines the cost of a routing switcher. A stereo switcher with 12 inputs and 12 outputs contains 288 crosspoints (12 inputs × 12 outputs × 2 audio channels = 288). If the switcher is visualized as two side-by-side matrices of 12 horizontal lines (inputs) intersected by 12 vertical lines (outputs), each intersection becomes a possible connection point. One matrix represents left-channel audio; the other represents the right channel.

In a television station, engineers must decide whether individual switchers handle left, right, mono, SAP, or if one large system routes all signals and audio. With the larger system approach, mono sources connect directly into left and right channels. The switcher may correct channel reversals and create mono mixes. A switcher with sufficient cross points to handle all switching tasks requires a larger investment than several smaller ones assigned to individual channels. The station considering the new ATV format is looking at a major investment in routing for the system's six channels of audio.

Distribution Amplifiers

When distributing audio to a number of locations on a continuous basis (without switching), a distribution amplifier (DA) proves invaluable. Sending a console's output to several recorders and other studios or routing a satellite receiver's feed to all studios ensure the distribution amplifier an important role in audio routing. A DA eliminates the need for constant patching and switching of various pieces of equipment. A DA becomes the only practical solution when audio must be fed in multiple directions on a constant basis.

The typical DA provides six to eight stereo outputs for each channel. Although there may be no input level adjustment, economical units should provide individual output trim pots. Modular distribution systems offer more versatility and avoid wasted, unused outputs. One model offers four stereo inputs, which can be assignable to any of its 14 stereo outputs by the use of jumpers. Popular DA options include metering, input level adjustment, audio compression, loss-of-signal alarms, and redundant power supplies.

STUDIO MONITORS

The control room audio monitoring system provides the first line of defense in spotting equipment failures and problems. For that reason, professional monitor speakers should be selected.

In choosing monitors, room size dictates cabinet size. In a large studio, invest in monitors with 12-inch woofers, 5-inch (13-cm) mid-range cones and horn or dome tweeters. Size limits small studios to a model with 5- or 6-inch (13- or 15-cm) woofers. Current speaker technology offers amazingly good sound quality from small cabinets. The designer should look for low distortion and flat response.

A meticulous studio designer will consider background noise sources, reverberation time of the room, interaction from walls and ceiling, and room equalization. Doing this properly means testing the control room with a real-time analyzer and positioning the monitors for best results. This is seldom practical.

When mounting the monitors on walls, suspension mounts, preferably with vibration-isolating components, should be used. Each monitor should be positioned an equal distance from the operator's normal position. Sound-proofing material should be used on as much of the flat wall surfaces in the room as possible.

In small studios, "near-field" monitoring provides the best solution. The monitors should be positioned in a triangular arrangement with equal distances between the monitors and the ears of the operator. Near-field monitoring ensures that the monitors will be close enough to the listener that the direct audio from the speakers will overpower reflections and any undesirable acoustics of the room. Mounting solutions include a shelf, or wall brackets, above the console, suspension from the ceiling, or floor stands behind the console but directly in front of the operator. Near-field monitors should be positioned at, or just above, ear level.

The power amplifier becomes another vital consideration. Space and power requirements limit audio console internal monitor amplifiers to 10 W or less. Noise and distortion specifications may not be as good as those of stand-alone amplifiers.

Matching the power amplifier with the requirements of the monitors means another task in studio design. Pushing a low-power amplifier to provide adequate listening levels can cause audio waveform clipping with distortion on peaks. Operating in this manner could damage the speakers. A better choice would be to operate a more powerful amplifier in a conservative manner.

To prevent DJs from blowing the speaker voice coils with too much power, fast-blowing fuses should be installed in the lines. The engineer needs to experiment with fuse values and listening levels to find the proper combination.

Just as important as amplifier power is the wiring between amplifier and monitors. At least #16 AWG should be used for low-power amplifier and speaker combinations. Heavier wire, up to #12 AWG, should be used for combinations above 100 W or long runs of speaker wire. The cables from the amplifier, out of the rack, across the ceiling, and then down to the monitors may eat up 50 ft (15 m) of wire even in a small studio. Audio purists insist that the wire length for both speakers remain equal.

AUDIO SOURCES

Compact Disk Players

The CD player commonly provides the audio source of choice in radio. CD technology encodes audio as digital bits recorded as etched holes on the surface of the disk. A transparent plastic coating protects the surface so that only an accumulation of dirt or scratches affect the playback quality. The bits are read by a laser beam focused on the spinning disk. Because nothing but the laser beam touches the disk, there is no wear.

Selecting the best equipment that the station can afford represents the most cost-effective choice. Several manufacturers build CD players designed specifically for broadcast and professional use. If the station that must use semiprofessional players should remember that they were designed to be used in a living room a few hours a week. These consumer-grade machines will not last indefinitely when run in a radio station 24 hours a day.

A station using semiprofessional players should keep two spare (meaning new, unopened, in the box) players in the station for quick replacement of a failed machine. No attempt should be made to repair a failed consumer-grade machine. They can be replaced with less trouble and expense than making repairs.

The output level and impedance of semiprofessional CD players are not the same as broadcast quality ones. If the console inputs require +4 dB levels and present 600 Ω loads, a matching interface should be used. These matching boxes convert the −20 dB, high-impedance, unbalanced output of the consumer-grade CD player to a +4 dB, 600 Ω, balanced source. Consumer-grade equipment can be difficult to cue and slow to start; the engineer should evaluate units carefully before committing to purchasing a quantity.

Turntables

Turntables still find work in some radio stations, but their importance has greatly diminished. Some unique source material remains available only on vinyl; the owner of one broadcast equipment manufacturing firm still sells 40 to 50 phono preamplifiers a month.

Turntables come in two flavors: the idler wheel design and direct drive. The once-common broadcast turntable used a motor which turned at 1800 rpm driving an idler wheel. The idler wheel in turn drove a large hub at the center of the platter. This design minimized wow and flutter caused by fluctuations in motor speed. The use of a heavy platter achieved further speed stability. Rapid starts necessary for tight cueing required a heavy, powerful motor.

The direct-drive turntable became more popular because of its reduced noise, wow, and flutter. There is no idler wheel to replace or bearings to lubricate; the platter is the rotor of these slow-turning electronic motors. The speed control circuits of direct-drive turntables constantly monitor and adjust their speed, keeping it more accurate than if left to line voltage and frequency. Further contributing to the demise of the rim-drive tables, the speed control function made precise speed enhancement of music possible. Circuit repairs may be a problem because of the minimal documentation provided with most direct-drive turntables.

No real broadcast tone arms remain; all current models are designed for consumer use. They track well and adjust easily but some prove difficult to cue and are not very rugged. A professional tone arm can be adjusted once and then left alone except for occasional testing. When installing a tone arm, the template from the turntable manufacturer should be used and the instructions supplied with the arm followed. The tracking weight is specified by the cartridge manufacturer.

The choice of the phono cartridge depends on the audio quality required. Rugged, less-expensive models give the longest life in on-air use. Moving up to more expensive but less rugged models gains better separation and high-frequency response. Consumer-grade phono cartridges should be avoided.

The turntable's preamplifier is easily neglected because it is never seen after installation. The important specifications of noise, frequency response, and separation need consideration. More expensive models offer filtering, high-frequency cut or boost, and adjustable cartridge loading.

Another pressing concern for the engineer is the preamp's resistance to radio frequency interference if the studio is co-located with the transmitter. The RF easily makes itself known in these high-gain amplifiers.

Microphones

No serious program director ever leaves the choice of the studio microphone to chance. Dynamic microphones remain the most popular for studio use. They are rugged, dependable, and affordable.

Condenser microphones crept out of the recording studios into FM stations during the 1970s and 1980s. Condensers yield flatter frequency response but cost more. Ribbon microphones were the industry standard 30 years ago but are traded only by collectors today.

Wireless microphones provide the advantage of mobility at radio remotes and for television use. The wireless systems use either a miniature lavaliere mike and belt-pack transmitter or a hand-held design with the transmitter in the microphone case. Television news crews value the extra directional characteristics of shotgun microphones in situations when the sound professional cannot get close to the on-camera person. TV studio sets use the shotgun mike on a boom to keep the microphone off-camera.

Audio Cart Machines

During the past 40 years, continuous loop tape cartridge (cart machines) proved invaluable for playing commercials, jingles, and music. Even stations that rely on digital audio storage system keep a few cart machines around as a backup system.

A mono machine uses two tracks, the upper track for program material and the lower for cueing. Stereo versions use three audio tracks on the endless loop of tape. Two of the tracks record stereo audio: the third carries cue tones. Trading recorded carts between mono and stereo players will not work because the tracks do not line up.

A brief 1000 Hz tone is recorded on the cue track at the beginning of the cart recording process. When the cart recorder is in the record mode, pushing the Start button generates the 1000 Hz "stop" tone and begins the recording process. After the tape loop cycles through the cart and returns to the starting point, the playback head detects the 1000 Hz stop tone and stops the tape at the beginning of the recorded program material.

Deluxe machines offer secondary and tertiary tones for cueing and starting the next tape. The secondary, or aux tone, is at 150 Hz tone and customarily triggers the next event in the program sequence of automation systems. The tertiary tone is a 8 kHz tone and triggers a cue light to warn air talent as the program material nears its end. The operator manually inserts the secondary and tertiary tones while recording the cart.

Although still a dependable and reasonably good storage medium for commercials and music, digital audio storage systems bulldozed the cart machine out of the radio business. In addition, while cart machines ruled the studio, their manufacturers failed to standardize on a single type of motor. As a result, when sales volumes fell, the price of all those custom motors rose at a logarithmic rate. Cart machine prices rocketed as the price of hard drives fell.

Reel-to-Reel Tape Recorders

The reel-to-reel recorder still remains a workhorse in some stations because of its simplicity and durability. Tape also provides an economical means for storing longer program material without filling the hard drive of the station's digital audio storage system.

Reel-to-reel recorders operate by mixing the incoming audio with a high-frequency ac bias signal of fixed level and frequency. This combined signal magnetizes the tiny ferric oxide particles attached to the plastic tape as it moves past the record head. During playback, the play head converts the magnetic fields stored on the recorded tape to an audio voltage sonically equal to the signal originally recorded.

The bias signal ensures that the record head creates a magnetic field sufficient to penetrate the ferric oxide portion of the audio tape fully. The frequency of the bias signal must be supersonic and typically is at least five times the frequency of the highest audio frequency recorded on the tape. The bias signal may be optimized for a particular audio tape by adjusting the bias level to produce minimum harmonic distortion when recording and reproducing a sine wave at a frequency in the range of 2 kHz to 3 kHz. The operator's manual for the reel-to-reel recorder will contain instruction for optimization of the bias signal.

During playback, an equalization curve applied in the playback preamplifier ensures that the reproduced audio produces a mirror image of the audio previously recorded. The equalization curve corrects inaccuracies related to the electrical characteristics of the record head, playback head, and speed of the tape as it moves past the heads. Both low-frequency and high-frequency compensation perfect the playback process.

Alignment of both the record and playback heads provide job security for the station engineer. Both heads must remain exactly perpendicular to the tape as it moves past the heads. Incorrect azimuth (side-to-side) alignment causes poor high-frequency reproduction. Stereo recorders exhibit a loss of stereo separation when allowed to drift out of perfect 90° azimuth alignment. Improper zenith (front-to-back tilt) also contributes to the high-frequency reproduction problem.

Professional machines include a third head in the tape recording process. An erase head uses the bias signal to clear any previously recorded audio from the tape during the recording process. If damaged, or not properly aligned, the erase head will leave remnants of audio beneath the new recording. In extreme cases, the old recording makes itself heard during silent, or low-level, portions of the new recording.

In the radio station control room, the reel-to-reel records news feeds from networks and reporters in the field. FM stations record music requests and contest winners on a reel-to-reel for delayed playback. Some program material arrives in the station on reel-to-reel tape. A two-track stereo deck with speeds of $7\frac{1}{2}$ IPS (inches per second) and 15 IPS fills the requirements of most control rooms.

Tape reels constantly turn in the typical production studio. Tape edit points, marked with a grease pencil, are cut with a razor blade and edited in a splicing block. Tape containing unwanted audio is discarded, and the two edit points are then spliced together. Words can be cut out and loose cues tightened using this method. This works well with mono or two-track stereo formats. When editing with a multitrack machine, the operator should remember that all tracks are cut on the tape the audio is edited.

When producing commercials with multitrack machines, the operator records elements of the production on different tracks and then mixes all tracks to a single stereo mix as the finished product records to cart. A multitrack recorder makes adding tags or reading copy into "doughnut" tapes much easier. The stereo music bed or agency tape is recorded on two tracks, and a third track contains the local copy. If the announcer makes a mistake, only the voice track must be re-recorded.

Multitrack recorders have a selective synchronization feature that switches the record heads of the tracks not in the record mode to the playback amplifier in order to synchronize playback with recording. Without this feature the timing between playback and the recorded audio on different tracks will be off by an amount equal to the distance between the record head and the playback head.

The favorite tape speeds for production work are 15 IPS = 38.1 cm/s and 30 IPS = 76.2 cm/s. Faster tape speeds generate the widest possible bandwidth and best audio quality. Faster speeds make cut and splice editing easier because the audio spreads over a greater distance on the tape.

Telephone Hybrids

During the 1990s, talk radio and TV talk shows moved to a position as a dominant format leader. This challenged station engineers to get the caller's voice from the telephone to the transmitter. The clash between old and new technologies made the job difficult.

The telephone system between the telephone company's central office and the home, or business, remains largely the same as it was at the turn of the century. We still depend on a pair of copper wires to transfer voices from one place to another. This part of the dial-up telephone network still operates as a two-wire system.

Both the voice being transmitted and the voice being received mingle back and forth on the same pair of wires. A telephone hybrid converts the two-wire system into a four-wire system which separates the caller's voice from that of the talk show host. Figure 7 shows the theory behind a telephone hybrid.

In this example, the core of the hybrid consists of two transformers, each having a single primary winding and two secondary windings. The talk show host's voice, the transmit audio, feeds to the phone line from the primary of T1 and through secondary #1 of T1. The caller's voice, the receive audio, travels from the phone line through secondary #1 of T2, then to the primary winding of T2. Note that the transmit

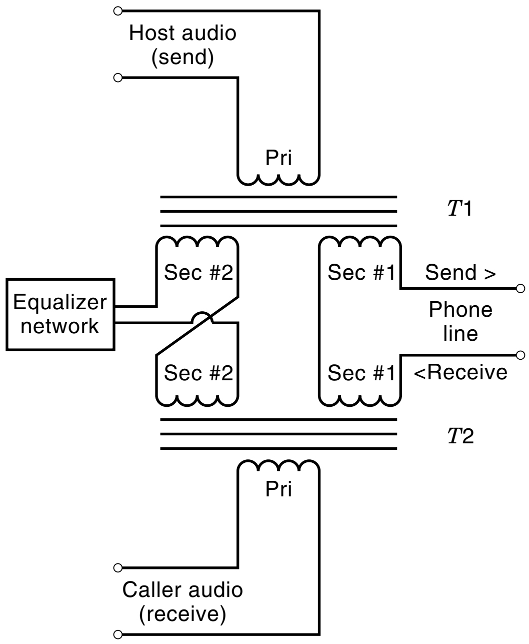


Figure 7. The telephone hybrid circuit converts the two-wire telephone line into a four-wire circuit. The hybrid creates individual send and receive audio connections separating the caller's voice from that of the host.

audio also passes, in series, through secondary #1 of T2. This means that the transmit audio appears in T2.

Note what is happening with secondary #2 on both T1 and T2; they are wired out-of-phase in relation to the first secondary windings. Although the transmit audio is introduced into T2 because secondaries #1 of both transformers are wired in series, the transmit audio is canceled out by the out-of-phase wiring of secondaries #2. This leaves, in theory, only the receive audio, the caller's voice, on the primary of T2.

If the phone line exhibited a perfect 900 Ω resistive load, a 900 Ω resistor at the location of the equalization network would produce a perfect match. Telephone lines do not represent a perfect world. Capacitance creeps into the mix because the phone company uses miles of #22 gauge twisted pair to connect to its central office. That much twisted wire forms a nice capacitor. The resistance of the circuit varies with the length of all that #22 gauge wire running back to the phone company. Loading coils, used in the telephone circuits to flatten frequency response, add the inductance.

We connect our hybrid to what amounts to an RCL network when we plug into the phone line. The null created by the out-of-phase circuit created by the #2 secondaries no longer matches the amount of transmit audio induced into T2 through its #1 secondary. A tunable equalization network wired between the #2 secondaries compensates for the electrical characteristics of the phone line circuit. When improperly tuned, audible amounts of out-of-phase host audio mixes with the caller's voice making the host sound "hollow," as if the show were taking place at the bottom of a large metal trash can. Our hybrid circuit may cascade into feedback in extreme cases of mismatch.

An analog phone hybrid requires tuning for best performance on each phone line to which it will connect. The process involves transmitting pink noise during a call to an outside phone number and then tuning the equalization network for a minimum level of pink noise at the hybrid's caller output. Digital phone hybrids accomplish this task with a short pink noise burst at the beginning of each phone call. The result is perfect separation of host and caller audio with excellent-sounding talk show audio.

Integrated hybrid systems include multiple hybrids and call switching functions in a single unit. This approach removes the challenge of building a talk show phone system from scratch with equipment from different manufacturers. The multiple hybrid system also handles the requirement of multiple mix-minus feeds. The installation time saved justifies any additional cost of the combined phone hybrid and call handling system.

Broadcast Delay Units

Radio talk shows can be hazardous to a station's liability insurance. The spontaneity of a good talk show ensure that callers can, and will, say anything. This prompts the need for a system that allows time to "pull the plug" before certain words, or accusations, pass through the transmitter.

Before digital technology accomplished this task with no moving parts, there was tape. Tape delay systems employed a special cart machine with an erase head. A 10 s cart was inserted in the recorder. Real-time audio was recorded on the tape. It took 10 s for the tape to loop through the cart before reaching the playback head. The output audio emerged from the recorder 10 s after it was recorded, allowing time for the talk show producer to interrupt the delayed audio containing profanity.

Simple digital delays perform this task without the worries of moving parts and broken tape. The device simply converts audio to a digital signal, records it in memory, and then plays it back 10 s later. The talk show producer mixes the program in the console's audition channel, which feeds the input of the digital delay. The output of the delay routes to the program channel of the console and to the transmitter.

If things go wrong, the producer turns off the mixer carrying the delayed audio to the program channel and inserts fill music and has the option of switching the show host from the audition channel to the program channel after dropping the caller. The host then resumes the program in real time.

Both tape delay and fixed-time digital delays pose the problem of transition in and out of delay. If the show host simply starts talking at the beginning of the program, words won't exit the delay system until 10 s later. Meanwhile, the audience is treated to 10 s of silence. Most stations overcome this by playing a 10 s recorded introduction to the show on the program channel as the show host begins talking on the audition channel. The host's voice exits the delay precisely as the recorded message ends. If timed properly, the transition is seamless.

A better approach to the problem of keeping talk show audio respectable is the digital delay unit that gradually builds the delay at the start of the program. This delay digitally records the real-time audio into memory. While it is building the delay time, it plays back the program audio slightly more slowly than it is being recorded. This process gradually fills the memory until the delay time reaches the maximum. After the memory fills, this delay operates just like the fixed-time delay; the audio appears at the output 10 s after it is received at the input.

If a caller says something offensive, the host pushes a dump button that erases all, or part, of the audio in memory. Because the profanity is stored in memory, it disappears when the dump button is pushed. The program now is on the air in real-time and the delay begins the process of rebuilding the delay time again. After about a minute the host has enough delay in memory to begin taking callers on the air again.

This type of delay allows easy return to real time at the end of the talk program. A few minutes before the show ends, the producer puts the delay in the exit mode. Now the delay records the program, but it plays back slightly faster than it is recording. This eventually depletes the audio stored in memory and the program returns to real time. The best of these delays features a relay bypass that takes the delay off-line when it is not in delay, or if it fails.

Audio Remote Systems

One of the most profitable activities for a radio station is the commercial remote. This consists of packing off one or two of the on-air staff with a van full of prizes, amps, speakers, and a microphone to a remote location. The primary concern is to transmit the voices of the talent from that site back to the radio station with reasonable quality. The dial-up telephone network provides a cheap and easy solution.

The output of a simple microphone mixer, or small audio console, connected to a telephone line coupler provides the basis for the most elementary remote system. The person doing the remote calls the station, is connected to the control room console via the station's telephone hybrid, and monitors the off-air signal for his cues. This system is easy to set up and operate, but the dial-up phone system limits the audio quality. Our telephone system, designed only to transmit voice from one telephone to another, limits the bandwidth to a range of 300 to 3200 Hz. The talent at the remote site *sounds* as if they are on the telephone.

Telephone Frequency Extenders

The problem with using the dial-up telephone system for delivering broadcast audio is the limited frequency response. The fact that the telephone system rolls off all audio below 300 Hz costs 2.5 octaves of audio on the low end (50 to 300 Hz). This tends to product the "tinny" characteristic that makes unprocessed, dial-up telephone remotes sound bad when compared with the full-spectum audio of regular programming. There is a solution.

Analog telephone frequency extenders trick the telephone system into passing audio with a bandwidth of 50 to 2900 Hz. An encoder-decoder process shifts audio frequencies upward by 250 Hz, sends the up-shifted audio over the phone line, and then returns the audio to its normal frequencies on the receive end. The encoder travels to the location of the remote broadcast and serves the double duty of shifting the audio up by 250 Hz plus connecting the remote mixer to the dial-up telephone network. Most include telephone touch-pads for dialing.

The encoder converts 50 Hz audio upward by 250 Hz to a frequency of 300 Hz. Audio at its natural frequency of 2950 Hz exits the encoder at a frequency of 3200 Hz, which barely squeezes through the limited bandwidth of the phone system. The output of the frequency extender encoder sounds quite strange. Even the lowest baritone voice sounds very high-pitched. Obviously, reverse treatment is required on the receiving end.

The frequency extender system's decoder shifts the audio which it receives down by 250 Hz, restoring it to the original frequencies. The analog frequency extension process delivers an audio bandwidth of 50 to 2950 Hz. The result is very pleasing voice transmission over a dial-up telephone circuit. The process sacrifices one-seventh of an octave between 2950 and 3200 Hz, but it restores 2.5 octaves between 50 and 300 Hz.

The system suffers from two disadvantages. Analog frequency extension is a one-way system. When the remote is out of the range of the broadcast station's signal, a second telephone line is required for talk-back and cueing. Compatible equipment is required on each end. A station in New York cannot send frequency-extended audio to a station in Los Angeles unless both have identical equipment.

Digital Audio Codecs

The computer age brings another means of transferring high-fidelity audio from one point to another. We can now digitize the source audio, apply data compression techniques, and send the audio over telephone lines via modems. Because computer modems operate bidirectionally, we also pick up the benefit of two-way audio communication over a single phone line.

In a nutshell, these digital audio codecs (coder/decoder) consist of a computer sound card, a modem, and data compression software bundled into a single package. The modem is designed for Switched 56, integrated services digital network (ISDN), or dial-up lines. Switched 56 service provides data transfer at a rate of 56 kilobits per second (kb/s). ISDN doubles the speed to 128 kb/s.

Dial-up lines limit data transfer to less than 53 kb/s and will vary from line to line with weather and with telephone traffic conditions. A 28.8 kb/s computer modem does not always connect at a speed of 28.8 kb/s. Switched 56 and ISDN service are consistent. The trade-off for less audio bandwidth brings the ease of connection to any existing telephone line without extra line charges and construction delays. For a 4 h commercial remote, the use of a dial-up line is a moot point. Coverage of a week-long special event may justify the expense of an ISDN line.

Systems designed for Switched 56 lines provide a 7.0 kHz, bidirectional audio circuit. ISDN service doubles the bandwidth to 15 kHz or allows stereo 7.5 kHz audio transmission. Improvements in modem speed and technology now challenge these premium services with bidirectional audio bandwidth of up to 10 kHz over dial-up telephone lines. The program audio bandwidth capability depends not only on the bandwidth of the telephone circuit, but on the data compression algorithm used in the codec.

The audio compression (data reduction) algorithms most frequently employed with Switched 56 and ISDN lines include ISO MPEG Layer II [International Standards Organization (ISO), Moving Pictures Experts Group (MPEG)], ISO

MPEG Layer III, apt-X, Musicam USA, or the international telephone standard ITU G.722 (formerly CCITT G.722). The rules of compatibility dictate that the codec on each end of the telephone line use the same algorithm. MPEG Layer III will operate with a Layer II device, but performance is limited to Layer II levels. MPEG is not compatible with ITU G.722. Likewise, apt-X talks only with apt-X. Musicam USA will converse with MPEG Layer II.

G.722 introduces a minimal delay, making it the most popular algorithm for talk shows and live remotes. Stations often find themselves using their top-of-the-line codecs in the "plain-vanilla" G.722 mode. A further discussion on audio compression algorithms is contained in the "Digital Audio Systems" section of this article.

POTS Codecs

The compression algorithms discussed above were designed for use on Switched 56 or ISDN lines, which have a guaranteed data rate of 56 kb/s or 64/128 kb/s. Standard analog telephone lines (POTS, or plain old telephone service) provide varying data rates depending on line quality. Telephone modems, such as the one in your computer, are designed to scale the transmitted and received data rate accordingly. This sliding data rate makes it very difficult to use compression algorithms, which were designed for specific transmission speeds.

In the past couple of years, new technologies have emerged that permit the transmission of compressed audio over POTS, thus delivering a bandwidth much higher than the normal 3 kHz signal normally available. POTS codecs use high-speed modems (33.6 kb/s as of this writing) and compression rates in the 12 : 1 range to deliver audio bandwidths varying from 5 kHz to 10 kHz, depending on the connection speed of the modems.

There is an important difference between audio sent by POTS codecs and audio sent via computer or the Internet. Computers use asynchronous modems, which means they send a packet of information, wait for confirmation that the packet was received, then go on to the next packet. For one-way delivery of audio that has been stored at a site, this method of transmission is not a problem. However, asynchronous modems must use a buffer to reconstruct the audio. This creates a delay large enough that two-way communication with the studio is simply not possible over the single telephone line used for the remote. For "real-time" bidirectional remotes, most POTS codecs use synchronous modems. With a synchronous modem, a bit is transmitted, and without waiting for verification of receipt, the modem goes on to the next bit. This constant data stream in both directions results in audio that is delayed only by the compression algorithm used. The modems go through a complex "handshaking" process to determine the best data rate for transmission in order to minimize errors caused by lost bits.

There are four companies producing POTS codecs, and each company uses a different approach to audio compression. One company uses ITU G.728 compression, also known as code excited linear prediction (CELP). Two other manufacturers use modifications of the MPEG Layer III algorithm; however, their systems are not compatible with one another. The fourth uses a proprietary compression algorithm designed specifically for the slower connect speed of POTS. All four companies' codecs are designed to scale transmit speeds and bandwidths according to line quality; however, the range of connection speeds varies with the manufacturer. Before buying a POTS codec, the designer should try to get a pair of demonstration units to determine whether they will work for the station's application. Remember, two codecs are needed, both from the same manufacturer.

DIGITAL AUDIO SYSTEMS

The Standards

The Audio Engineering Society and European Broadcasting Union (AES/EBU) standards provide a benchmark for digital inputs and outputs. Fortunately, these standards provide a common denominator found on all professional equipment. AES/EBU, also known as AES3, created a balanced system that can transmit digital stereo audio up to 100 m over a single shielded, twisted-pair wire.

AES/EBU calls for internal transformer coupling, with dc blocking capacitors, on both input and output circuits. The circuit is designed for shielded audio cable with an impedance of 110 Ω. The digital input protocol must match the digital output of the source. AES3 ensures that any two pieces of equipment using this standard will work together. Copies of AES3—1992 are available, for a fee, from the Audio Engineering Society in New York; the phone number is (212) 661-8528.

The AES3—1992 standard specifies a nominal signal voltage between 2 and 7 V measured across a 110 Ω terminating resistor. An earlier standard (1985) allowed a 2 to 10 V range. No compatibility problems exist between the 1985 and 1992 standards.

AES3 specifies connections using the familiar XLR three-pin audio connectors. Pins #2 and #3 carry the digital signal. Pin #1 is ground. This convention is exactly the same as used in wiring XLR connectors for microphone or balanced line-level audio. The connectors remain the same but the wire does not.

Miniature broadcast audio cable (Belden 8451 or West Penn 291) does not meet the impedance specification. Digital audio is actually computer data that run at a rate of 64 Hz times the sample rate. Digital audio, sampling at a rate of 48 kHz, becomes a data stream running at 3.072 Mb/s (64 Hz $\times$ 48 kHz = 3.072 MHz). Cable capacitance can rapidly degrade the 3 MHz signal. Low-capacitance digital audio cable will do a satisfactory job of transferring the digital audio signal for the AES3-specified distance of 100 m (328 ft).

The need to transport digital audio farther that 100 m (328 ft) resulted in a new standard using unbalanced coaxial cable. AES-3id—1995 allows transmission of a 1 V digital audio signal up to 1000 m (3280 ft). Considering the 3 MHz frequency of digital audio, 75 Ω coaxial cable make a lot of sense. No knowledgeable engineer would suggest conducting a 3 MHz RF signal over twisted-pair audio cable. AES-3id calls for RG6A/U, or RG59B/U, cable and BNC connectors.

Television and video production facilities rapidly embraced this new standard. AES-3id not only allows for longer cable lengths, but also permits them to use 75 Ω cable, terminated with BNC connectors, for both video and digital audio signals. Copies of AES-3id—1995 are available for a fee from the

Audio Engineering Society in New York. The Society's phone number is (212) 661-8528.

Sony and Phillips developed a standard for consumer-grade equipment. S/P DIF (Sony/Phillips Digital Interface Format) defines an unbalanced digital connection. S/P DIF and AES3 signals do not mix in all cases. An AES/EBU input will accept a S/P DIF output, but a S/P DIF input will not accept an AES/EBU output without an interface to correct the differences in the data formats and wiring. Only equipment with AES3 inputs and outputs should be used.

S/P DIF specifies the old RCA phono pin connector; 75 Ω coax works well as the conductor. The problem of yet another connector and cable type in the studio proves reason enough to standardize on AES3 digital equipment in the station.

The AES/EBU (AES3) standard supports 16, 20, or 24 bit quantization formats. The 20 bit audio sounds better than 16 bit; 24 bit offers improved performance over a 20 bit digital signal. When audio is digitized, sine waves are changed into vertical samples based on time. They also get sampled horizontally based on amplitude. Amplitude resolution is determined by the quantization format, the bit rate. Each bit communicates 6 dB of amplitude information.

In theory, the 16 bit audio of a CD can communicate up to 96 dB of amplitude change (6 dB × 16 bits = 96 dB). The 24 bit digital audio can reproduce a theoretical dynamic range of 144 dB (6 dB × 24 bits = 144 dB). In the real world, digital audio equipment achieves performance less than theoretically possible. The 16 bit audio typically renders 90 dB of dynamic range due to the limitations of the analog-to-digital converters and digital-to-analog converters.

An increase in dynamic range means a lower noise floor. Note the 48 dB reduction in the noise floor when comparing 16 bit digital audio with a 24 bit signal. Assuming a clip level of +24 dBu, 16 bit digital audio equipment can theoretically reproduce a minimum signal level of −72 dBu. Twenty bit digital audio has the potential of reproducing a minimum signal of −120 dBu. Figure 8 illustrates the difference in possible dynamic range between 24, 20, and 16 bit digital audio.

AES3 standardizes sample rates of 32.0, 44.1, and 48.0 kHz. The digital recording process limits the maximum recordable audio frequency to one-half the sample rate. Each cycle of a sine wave must be sampled at least twice during a cycle. If not sampled once during its positive peak and once on its negative peak, the sound cannot be accurately sampled and converted to digital audio.

A sample rate of 44.1 kHz permits audio bandwidth up to 22 kHz. The sample rate of 48.0 kHz pushes the upper limit to 24 kHz. Equipment using a 32 kHz sample rate chokes off the audio at 16 kHz but consumes less bandwidth and/or hard disk space.

Today the existence of three sample rates hinders digital broadcasting. Compact disks came out of the gate with the sample rate of 44.1 kHz. This single-standard sample rate crossed all brand names and entrenched itself as *the* sample rate for CD work. Recording studios adopted the sample rate of 48.0 kHz in an effort to achieve better fidelity. Digital audio workstation and digital audio storage equipment manufacturers shifted downward to 32.0 kHz to conserve disk space because FM radio passes only 15 kHz of audio bandwidth.

No manufacturers of digital broadcast equipment currently exhibit any interest in trying to develop a standard sample rate frequency. They support all three sample rates, driving up the cost of digital equipment. The issue of sample rate frequency remains as another roadblock slowing the progress of the digital output of a CD player from directly reaching the FM digital exciter.

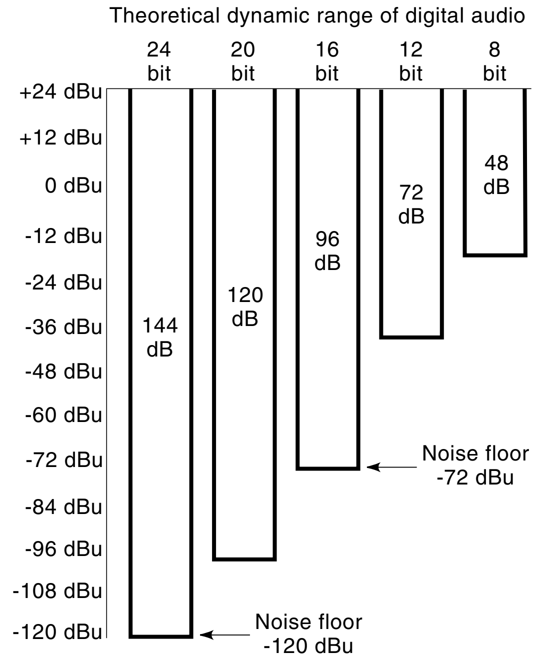


Figure 8. Higher quantization rates (bit rates) result in a wider dynamic range for digital audio. 24 bit audio provides a theoretical dynamic range of 144 dB for a 48 dB improvement over 16 bit audio. This also means a 48 dB lower noise floor.

As digital audio consoles approach the threshold of the control room, the engineer wishing to "go digital" in the studio has a problem. The digital audio of different sample rates cannot be combined even though all the sources adhere to the AES/EBU standard. Somewhere in the studio system sample rate conversion must take place. The 44.1 kHz sample rate of the CD must switch to the 48.0 kHz sample rate of the console, which must convert to the 32 kHz sample rate of the digital STL.

One console manufacturer offers input modules that match the sample rate of input audio to that of the console. Its solution to the problem is sample rate conversion on each input. A standard sample rate for all broadcast applications offers a better solution, but that has not yet happened. Careful planning during equipment selection remains the best defense for minimizing potential problems.

Even after the sample rate problem has been addressed, one more problem surfaces when a digital console arrives in the on-air studio. When mixing two digital audio sources of the same sample rate, their clocks must be synchronized. If one digital audio source runs at 48,000.00 Hz and is mixed with a second signal running at 48,000.02 Hz, and the internal clocks run separately, digital "train wrecks" occur.

The results pollute the combined audio with clicks and pops. A digital master clock provides a workable solution. A single timing signal connected to all digital audio equipment in the station synchronizes everything in frequency as well as phase.

This plan works when all digital equipment in the studio offers clock input connections. However, not all digital equipment is designed with this feature. Older digital equipment featured a “word clock,” which ran at the sample rate. Newer equipment locks internal clocks together with the AES standard digital audio reference signal (DARS), which is an AES3 digital signal without audio. The digital equipment manuals must be checked to verify which systems will operate in a facility.

Three more pieces of the AES/EBU puzzle remain. The first bit of the digital signal indicates whether the output signal originates from consumer-grade equipment or professional gear. If the first bit of the first 8 bit word is a 0, the source audio is from consumer equipment. Professional equipment identifies itself as such with a 1. Many professional recorders will not accept a signal from semiprofessional equipment.

Digital audio may also include a consumer copyright protection bit. The serial copy management system (SCMS) prevents illegal digital copying. Digital audio also includes a parity checksum for the channel status data. The cyclic redundancy check (CRC) error light on the recorder may indicate a problem with the recording configuration.

Unfortunately the engineer will need a digital audio analyzer to identify these situations correctly. The fixes require a means of real-time editing of the channel status bits or reprogramming of the equipment.

System designers greatly increase the storage capacity of hard disks by using the bit-rate reduction techniques of ISO MPEG, Audio Processing Technology’s apt-X, Musicam, the international telephone ITU G.722 standard, Dolby’s AC-2 or AC-3. Without bit-rate reduction, 1 minute of stereo audio consumes about 10 MB (megabytes) of hard drive space. With a 4 : 1 bit-rate reduction the same 10 MB could store 4 minutes of audio.

Early algorithms accomplished bit-rate reduction by transmitting only the difference between samples. These systems gained a 2 : 1 reduction in digital data, but were nondestructive. The algorithm restored the original audio signal when decoded.

Lossy bit-rate reduction discards bits not needed because the human ear would not hear the sound reproduced by the bit when played back. The sound would be either too low in volume to be heard, or covered up by a louder sound. The data-compressed information reproduces an audio signal almost indistinguishable from the original to the critical ear. This destructive process changes the source audio forever. After the data are compressed, the exact original audio can never be recovered. Huge gains in storage space and transmission bandwidth requirements justify the use of these algorithms.

The MPEG compression algorithm is the audio portion of a video compression system standardized by the ISO. The facts that (1) it is accepted worldwide and (2) it is attached to a video compression scheme ensure that it will be here for a long time. MPEG offers three levels of signal quality identified as Audio Layers I, II, and III. The complexity of the system, including hardware, rises and the quality of the audio improves with each advancing layer. Audio Layer I provides transparent 20 kHz audio quality at a compression ratio of 4 : 1. Audio Layer II achieves the same results but at a ratio of 6 : 1. Audio Layer III operates as high as 24 : 1.

Musicam USA’s compression system is compatible with MPEG Audio Layer II. Musicam chops the 20 kHz audio spectrum into bands of 750 kHz. It then discards any unnecessary bits in each band. The Musicam system represents a compromise between transparent re-creation of compressed audio and complexity of the processing algorithm. Processing delays increase with complexity of any system’s algorithm.

Bit-rate reduction serves to allow digitized audio transmission over limited bandwidth media. Early digital STL systems use bit-rate reduction to squeeze two channels of 15 kHz audio and remote control data into the 500 kHz bandwidths of the 950 mHz spectrum. Remember that bit-rate reduction algorithms throw away part of the digitized audio, preventing further audio processing. All audio compression and limiting must happen before the program audio passes through a digital STL that uses bit-rate reduction. Two digital STL systems that do not use audio compression algorithms were introduced in 1997. These solve the problem of bit-rate reduction and should become the systems of choice.

T1 digital telephone systems offer enough bandwidth to allow uncompressed audio to reach the transmitter. A T1 STL system offers flexibility in the location of audio processors. Multiband compression may be applied at the studio, and final limiting accomplished at the transmitter site. T1 systems also solve the problem of the congested 950 mHz spectrum.

Technology changes rapidly. An engineer should carefully study current technology as applied to the products available before making an investment in digital equipment.

Digital Audio Storage Systems

Digital audio storage offers the most rapidly developing technology in the broadcast audio field. Its acceptance by the industry became apparent in 1995; both Ampex and 3M announced their exit from the magnetic tape business. The magnitude of this development becomes evident after considering that Ampex pioneered magnetic tape recording in the United States. Working with development money provided by Bing Crosby, it perfected the process that was first used on Crosby’s radio show. Digital recording media now include magnetic tape, hard drives, high-density floppy disks and magnetooptical (MO) disks.

Designed for the consumer market, the rotating head digital audio tape (DAT) machine became widely accepted by broadcasters. This format uses a rotating head much like a videocassette recorder (VCR), and records on a cartridge tape. DAT cartridges offer recording times of 60, 90, and 120 minutes. The cartridge design prevents razor blade editing. Professional models feature fast cueing, instant starts, remote control, and time-code compatibility. DAT machines offer broadcasters the ability to record live events, concerts, and network feeds in the digital domain without increasing noise and sacrificing dynamic range.

Stations using multitrack consoles for production work may take advantage of eight-track DAT recorders. Like their smaller cousins, the eight-track models use a rotating head but record on VHS video tape cartridges. These DAT recorders excel in recording live concerts and storing multitrack production work.

Smaller facilities may opt for the digital cart machines coming on the market. These record to super-high-density floppy disks, minidisks, or MO disks. They offer the advantages of low-noise digital audio reproduction and the elimination of head alignment woes and maintenance associated with analog cart machines. But it leaves the disk jockey still sorting and slamming plastic devices into slots and pushing start buttons.

Note, however, that there is still a need for human intervention for trouble shooting and problem solving.

If a station is planning a transition to "digital," why handle the storage medium at all? Digital audio storage systems using hard drives as the storage medium threaten to oust cart machines and carts of all types from the control room. Hard-drive systems now offer the capability to store and play all the station's commercial library plus its program material. Multiple hard-drives provide the necessary "crash protection." The systems can automatically record network feeds for delayed broadcast, eliminating the need for reel-to-reel recorders. One large system offers simultaneous access to program audio by seven studios.

Digital Editing and Work Stations

The digital audio workstation benefits the broadcast station with faster and more creative production, using a process quite similar to the way word processors edit written text. Digital workstation systems store multiple audio tracks on hard disk and allow editing in random access memory (RAM). Two or more analog input channels (through A/D converters) and direct digital inputs receive the incoming audio. Options allow analog or digital outputs. They allow editing tracks individually and produce a finished product by using keyboard and scrub wheel rather than grease pencil and razor blade. The audio waveform of each track crawls across a screen allowing visual as well as audible cueing and editing.

To produce a spot with a digital audio workstation, the operator records the audio tracks and music beds in the system memory, commands the workstation to move component sounds, adjusts timing, edits tracks, and finally completes a stereo mix. Unlike razor blade editing with analog tape, the software process preserves the original material. Correcting mistakes and editing experiments become child's play with the Undo key found on most editors.

Workstation editors most closely emulating the operation of a reel-to-reel recorder get production work flowing quickly with minimal training time. An accurate scrub wheel operation speeds the work of a producer tightening loose voice tracks and other tricky edits. Scrub audio sounds exactly like that heard when manually rocking tape back and forth across the playback head of an analog tape deck.

A good display shows the audio waveform of all tracks on the screen simultaneously. Most editors show a vertical cursor line that moves across a stationary audio waveform as the editor reproduces the audio. This display mode consumes fewer computer resources than a system that shows the audio waveform crawling across a stationary cursor because the screen is not continually requiring refreshment. Some audio producers favor a display that depicts the waveform moving across a stationary cursor as the screen more accurately represents the operation of a reel-to-reel machine where the tape travels left to right across the stationary playback head.

Creative engineers can assemble a two-track audio workstation with a Pentium PC, a good sound card, and editing software. The system should include as large a hard drive as possible; the source material plus overhead consumes about 12 MB/minute for stereo audio. A 1 MB drive handles most two-track production tasks. The more elaborate systems include mixing capabilities providing an all-in-one approach for production work.

AUDIO PROCESSING EQUIPMENT

Compressors

All the hype whirling about concerning the on-air processing market simply clouds the central issue of reducing dynamic range of the source material. The difference between the loudest sound and the quietest equals the dynamic range. Audio compressors take large, rapidly rising audio voltages and make medium-sized, slowly rising audio voltages. The goal remains simple: reducing the dynamic range and increase the average modulation level of the transmitter.

The industry complicated matters when, in the 1960s, one engineer found a way to crank the average modulation level up 3 dB higher than the station across town. The resulting "modulation wars" in the AM band trashed audio quality and drove listeners to FM. Then Mike Dorrough showed us how to crunch FM audio without sounding quite as bad. Bob Orban next combined the stereo generator with Dorrough's multiband limiter and other than doing all this digitally, not much has happened since.

If the output of a CD player were connected directly to the audio inputs of a transmitter, the wide dynamic range would prevent the average modulation from reaching more than 50% to 60%. The station would disappear from the band among those processing their audio by today's aggressive standards. An AM station operating this way would sacrifice a part of its coverage area. An audio compressor reduces the dynamic range to a more practical spread and holds the modulation level at a much higher value.

Here is how it works. A threshold control defines the point where the compressor starts to attack a rising audio voltage. If the compressor's threshold control instructs it to attack a voltage at -20 dB, the compressor will attempt to hold down a rising audio signal after it rises above the -20 dB level. A signal below -20 dB will pass through the compressor unscathed.

After the audio passes the threshold (-20 dB), the ratio control instructs the compressor how serious it is about restricting the level of the audio voltage. A 5 : 1 ratio limits an audio signal that has risen 5 dB to an increase of only 1 dB at the compressor output. Using this example, an input signal of -20 dB would appear at the compressor output at -20 dB;

no compression would take place. An input signal of −15 dB (5 dB higher) would show up at the output at a level of −19 dB (1 dB higher).

Low ratios induce less compression for any given input above the threshold. When the user increases the ratio, the compressor aggressively attacks the dynamic range of the source material. The threshold presents more of a "brick wall" to the incoming audio as the operator dials in ratios of 10 : 1 and higher. Premium-quality compressors offer two thresholds with an independent ratio control attached to each threshold.

This type of compressor moderately squeezes a rising signal of low level as it crosses the first threshold level. Should the signal continue to rise above the second threshold, more aggressive compression attacks the signal. A compression ratio of 20 : 1 used above the second threshold provides the peak-limiting required to properly modulate a transmitter.

The compressor's "knee" is the point where the incoming audio signal rises above the threshold and the compression action begins. A "hard-knee" compressor engages the compression precisely at the exact threshold value with the exact ratio. A good ear hears the full amount of processing kick in as the signal rises above the threshold. If the compressor smoothes the transition with a gradual, rounded transition point at the threshold, the processing action engages slowly over about a 6 dB range. The transition into processing is less apparent as the compressor begins to act on the rising audio signal. This describes a "soft-knee" compressor. The old tube-type compressors, the UREI LA-2 for example, remain popular because of the smooth transition through their soft knee.

Release time determines how quickly the compressor releases the compression when a falling voltage drops below the threshold. A slow release time may measure 2.5 s or longer. A fast release time, measured in milliseconds, will release as each low-frequency waveform decays. A high compression ratio coupled with fast release time may result in too much of a good thing.

When more aggressive compression hammers the dynamic range, the normal soft passages in music disappear and the normal voice migrates to a stream of shouted commands. Overcompressed audio becomes irritating over a period of time. Some musical notes can actually be lost. If guitar notes and the beats of a kick drum arrive at the input of a compressor at the same time, the compression required to tame the kick drum will drop the level of the guitar so much that some notes will not be heard.

In the AM modulation wars of the 1960s, Mike Dorrough recognized this deficiency and split the audio spectrum into three bands. In the discriminate audio processor, the Dorrough DAP, Dorrough assigned each band its own compressor. The combined outputs of the compressors reassembled the complete audio spectrum while gaining independent compression for each band. Now the kick drum's compressor could be stomping 12 dB of compression while the guitar's band may only receive 3 dB of compression. The listeners no longer missed the rhythm guitar. Thus was born multiband compression.

This all sounds easy. Figuring out the frequencies included in each band made things more complicated. The proper crossover frequencies, the points on the audio spectrum where bass becomes mid-range and mid-range becomes treble were elusive and depended upon the program material. The crossover frequencies for the relatively narrow AM spectrum were not correct for the 15 kHz FM spectrum.

Also, that each band required different attack and release times. The much shorter waveforms of high frequencies require faster attack than the long waveforms of bass frequencies. Improper attack and release times for adjacent bands results in inconsistent processing in each of the three bands. Release times set too short for the low band will overcompress the bass notes. When the release time for the high-frequency band is set too long, the highs disappear and the station will sound "muddy."

Limiters

A peak limiter fits in the program chain just ahead of the transmitter, the stereo generator for FM. The limiter provides a brick wall for any overshoots that the compressor misses. Its sole purpose is to prevent overmodulation of the transmitter.

Limiters used for U.S. FM broadcast include the 75 μs preemphasis curve that boosts the high frequencies. Other countries may use different preemphasis curves; check the government's broadcast specifications. FM limiters operate symmetrically; positive and negative peaks receive equal limiting. AM limiters operate asymmetrically. They clamp their negative peaks at 100% modulation while allowing positive peaks to shoot upward to 125%. This maximizes the modulation and output power of the AM transmitter. AM limiters also include the National Radio Standards Committee (NRSC) preemphasis curve and 10 kHz low-pass filter. C-QUAM, AM stereo processing, usually requires an optional circuit board.

Most engineers locate the peak limiter at the transmitter site, feeding its input from the STL output. They leave the multiband compressor at the studio. Leaving the compressor at the studio end also enhances the modulation of a 950 mHz radio link.

AGC/Levelers

A novice may confuse an automatic gain control (AGC) amplifier with a compressor. The AGC/leveler is not an audio-processing device. It will not reduce the dynamic range or limit loud audio peaks. The AGC amplifier ensures that the compressor and limiter receive constant input levels.

The AGC operates similarly to a compressor when a signal above its target output range appears at the input. In this case, the AGC amplifier gently pulls the signal level down to the target output level. It is operating like a compressor set with a 2 : 1 ratio and a slow, 2 minute release time.

The AGC amplifier reacts differently when the input signal falls below the target output level. Now it turns up the gain of the low-level signal, bringing it slowly up to the target output range. Compressors cannot accomplish this level correction function.

Changing input levels to a compressor or limiter subjects the program audio to varying amounts of compression. Audio

Table 2. Audio Qualities and their Frequencies

Quality	Audio Frequency Range
Sub-bass	15–65 Hz
Bass	65–256 H
Voice	256–2,048 Hz
Upper-vocal	2,048–3,750 Hz
Presence	3,750–5,000 Hz
Sibilance	6,000–7,000 Hz
Brilliance	6,500–15,000 Hz

from a loud source may receive too much compression, creating distortion. If the output of a low-level source is not corrected, it receives little or no compression and sounds weak and thin. Sudden changes in the station's audio processing irritate listeners.

AGC/levelers provide a safety net for sloppy board operation, the times when the meters remain buried in the red for minutes at a time. Stations employing walkaway operations, where no one sits at the console correcting levels as the automation system switches between sources, benefit greatly from the processing consistency created by an AGC amplifier.

Equalization

Engineers work nights in anguish maintaining audio frequency response as "flat" or linear as possible. Program directors toil overtime in anguish producing a "signature" sound for the station. The two professionals butt heads at the equalizer. In the audio processing chain, creative use of an equalizer gives character to the station's audio. In production, an equalizer routinely cleans hum and hiss from noisy tapes. The production director also creates special effects and unique voice tracks with the help of his equalizer.

Low frequencies can be boosted to produce a heavy "thumping" bass and the upper mid-range can be boosted to add brightness. Enhancement of the highs and lows with an equalizer satisfies the desire to make a station sound better to the average nontechnical listener than the competition. Table 2 matches the audio ranges and qualities with the knobs on the equalizer.

Equalizers come in two types. The best known, the graphic equalizer, divides the audio spectrum into a series of bands represented by rotary or slider controls on the front panel. The operator adjusts the controls to affect gain in each particular band. The graphic equalizer with slider controls creates a visual picture of the frequency response curve that is being produced. Nontechnical people find it easy to use because of this design. Nine out of ten program directors prefer a graphic equalizer in the station's processing chain to tailor the on-air sound.

The parametric equalizer provides more versatility in the hands of a trained operator. A specific frequency can be dialed in with a parametric equalizer for elimination or boosting. A graphic equalizer does not allow this frequency-specific accuracy. The parametric equalizer also allows the adjustment of filter bandwidth or selectivity. A narrow bandwidth can "notch out" a hum or buzz. A wide bandwidth produces broad curve to boost or cut an entire band of frequencies. The typical parametric equalizer offers three or four sections covering the entire audio spectrum. Parametrics are considered valuable tools in the production studio.

Microphone Processors

Stations spend thousands of dollars for processing equipment to compress and equalize their music program material. The studio microphone often suffers from processing neglect although it deserves major attention. Compression, equalization and de-essing of the studio microphone give the on-air talent a chance of competing with the professionally produced program material played on the air. A good microphone processor elevates the local disk jockey closer to the voice quality level of professional voice talent.

Compression reduces the dynamic range of the natural voice, providing more power and punch by raising its average energy; it becomes louder. The added power prevents the voice from being buried by song introductions and music beds. The equalizer provides a means of boosting regions of the voice spectrum that lack natural presence. Male DJs always want a generous boost of low-frequency energy. A "muddy" voice benefits from a boost in "brightness" from the equalizer's upper mid-range (around 2.5 kHz). The de-esser reduces annoying, lisping, spitting "s" sounds. Sibilance problems show up at frequencies between 6 kHz and 7 kHz. The de-esser monitors the energy level in this range and kicks in additional compression to reduce sibilance problems. When adjusted properly, the de-esser will take the edge off a sibilance problem without punishment to the high-end frequency response. A de-esser circuit requires fast attack and release timing and a narrow (less than 0.5 octave) bandwidth.

The properly adjusted combination of 3 dB of compression, a one octave, 6 dB boost at 125 Hz, and a two octave, 3 dB boost at around 2.5 kHz from the equalizer can convert any voice to an acceptable quality. Table 3 provides some direction for equalizer adjustments.

Without voice processing on the studio microphone, the station may find itself overprocessing its music while stretching to achieve a suitable amount of processing for the on-air voices. The addition of a microphone processor provides a

Table 3. Vocal Qualities and Frequencies

Audio Frequency Range	Descriptive Quality
75–200 Hz	Rumble, heaviness
200–300 Hz	Bassiness, bigness
400–600 Hz	Warmth, chesty
600–1,000 Hz	Volume, loudness
2–4 kHz	Clarity
3 kHz	Presence, nasal
5–8 kHz	Enunciation, brightness
6–7 kHz	Sibilance
10 kHz +	Mouth and air noise

From 601 Digital Voice Processor User's Manual, courtesy of Symetrix, Inc.

more balanced processing mix from on-air voices and produced program material. Suddenly the local announcers no longer sound "wimpy" compared with network announcers and production studio voices.

Noise Reduction Systems

Noise reduction systems minimize source noise from analog audio tape, vinyl records, and remote program unit (RPU) radio links. Dolby and dual-ended systems record encoded audio on tape and decode it to reproduce the original audio with a lower level of system noise.

Audio "companding" (compression/expanding) systems provide another option. During the recording process, the dynamic range of the program material is severely compressed, keeping it further above the noise floor of the recording medium. The process reverses during playback when the recorded audio is expanded restoring the normal dynamic range. These dual-ended systems provide benefit in audio tape recording but require identical equipment on each end of the record/reproduce process.

Dolby and companding noise reduction systems hide noise. Single-ended systems provide freedom from the encode/decode process by "painting over" noise. These frequency-sensitive gating devices eliminate all audio in certain bands when the signal falls below a fixed threshold. The theory of operation assumes that anything below the threshold must be noise and should be eliminated. Single-ended systems offer a better solution than cutting the frequencies associated with tape hiss with an equalizer. The equalizer also eliminates desirable program material in that spectrum. The noise reduction unit attenuates a part of the audio spectrum only when no significant program material appears in that band.

Effects Generators

Digital effects generators enable production directors to produce the special effects of phasing, flanging, and echo for creative production. Several effects devices allow easy pitch changes of voice and music for special effects and to compensate for speed adjustments made to time tapes perfectly to 30 or 60 s. When tape speed varies more than 2% or 3%, pitch correction restores normal tone to the voice.

When using reverberation in program material, the amount mixed in should be 20 dB down from normal program level. Otherwise, the effect will be too distracting. When used with compression or limiting, it should be mixed back into the audio chain at the limiter output. This prevents the percentage of the mix from varying with the operation of the compressor/limiter.

SUMMARY

This article attempts to describe the state of the art in equipment and standards of engineering practice. A review of previous work on the subject of studio audio for the National Association of Broadcasters, starkly depicted the vast changes in the hardware realm in just five short years. Broadcasting technology now provides a digital signal path from the audio source to the transmitter.

The standards, which have served well since the infancy of broadcasting, now fall to new technology. Digital AES/EBU (AES-3) interface supersedes the "standard" 600 Ω, +4 dB connection between audio equipment. Long-familiar hardware—punch blocks, screw barrier strips, and XLR connectors—make way for DB25s, fiber optic cable, and ISDN.

The broadcast engineer can now build a studio using source material from digital audio storage devices mixed on a totally digital console. The console could become a touch-screen controlled computer. The all-digital processing system has arrived. Some stations have already replaced their 950 mHz STLs with T1 carrier systems. A digital STL that uses no data compression was introduced in 1997. Equalized program telephone lines are on the way out, and 111C repeat coils are obsolete. Digital stereo generators arrived several years ago; an engineer can now buy a digital FM exciter.

Retooling to provide the new technology in hardware challenges the equipment industry. New names are emerging and some of the industry leaders are fading from vogue as they fall behind the pace of changing technology. Pressure will be on broadcast engineers also to educate themselves in order to provide competitive product to their listeners and viewers.

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RADIO, CELLULAR. See CELLULAR RADIO; DIGITAL RADIO.

RADIO COMMUNICATIONS. See METEOR BURST COMMUNICATION; SKY WAVE PROPAGATION AT MEDIUM AND HIGH FREQUENCIES; TRANSCEIVERS.

RADIO, DIGITAL. See DIGITAL RADIO.