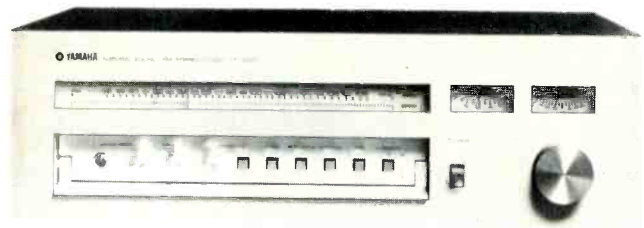
Yamaha Model CT-7000
Stereo FM Tuner



MANUFACTURER'S SPECIFICATIONS
IHF Sensitivity: Normal, 2.0 μV; Wide, 2.5 μV. **Selectivity:** Normal, 85 dB; Wide, 18 dB. **S/N:** Mono, 78 dB; Stereo, 75 dB. **Capture Ratio:** Normal, 0.7 dB; Wide, 0.6 dB. **THD:** Mono-normal, 0.06%; Mono-wide, 0.04%; Stereo-normal, 0.06%; Stereo-wide, 0.04% at 400 Hz. **THD** (from 50 Hz to 10 kHz): Mono-normal, 0.2%; Mono-wide, 0.08%; Stereo-normal, 0.3%; Stereo-wide, 0.15%. **Image, I.F. and Spurious Rejection:** Over 120 dB. **AM Suppression:** Over 60 dB. **Separation:** 400 Hz, normal or wide, 50 dB; from 50 Hz to 10 kHz; normal, 35 dB; Wide, 40 dB. **Frequency Response:** 30 - 15000 Hz ±0.3 dB. **Sub-carrier Suppression:** 70 dB. **Muting Override Signal Level:** From 3 to 30 μV. **Stereo Threshold:** From 3 to 30 μV. **Output Level:** 775 mV fixed; 2mV to 70mV variable.
GENERAL SPECIFICATIONS
Power Consumption: 23 watts, 13 watts with illumination off. **Dimensions:** 17 ¼ in. W x 5 ¾ in. H x 12 ½ in. D. **Weight:** 28.6 lbs. **Price:** $1200.00

The first thing that must be said about this magnificent FM tuner from Yamaha is that it succeeded in giving the test equipment in our laboratory a bad inferiority complex. Frankly, we thought we had upgraded our measurement equipment so that it was a couple of orders of magnitude better than anything we would be called upon to test. Yet the Yamaha CT-7000 specs are, in the main, better than we are able to measure. And rather than repeat that statement with each reported measurement that follows, we thought we'd better get that out of the way right at the beginning. We shall report our measurements as we read them, with the understanding that readers (and the people at Yamaha) will sympathize with our limitations.

As for the operating features, circuit highlights, and appearance of this tuner, those we can describe quite accurately, and the photo of the front panel helps. Many of the goodies are hidden under that sleek hinged door which runs across most of the lower portion of the front panel. But with it closed, all one sees when looking at the brushed silver panel of the Yamaha CT-7000 is a long, linear dial scale, calibrated at every MHz and augmented by a 100 point logging scale for easy station reference. Two small "windows" at the right frame the center-of-channel tuning meter and the signal-strength meter which also doubles as a multipath indicating meter. There's a lever-toggle power on-off switch and a massive tuning knob coupled to just about the smoothest acting flywheel arrangement we've ever spun across the 20 MHz's of the FM band. Ah, but when you tap the door base lightly to open it (it swings downwards), what additional wonders are disclosed! There's a built-in headphone jack powered by its own amplifier and a level control for that output, as well as one for the variable-level output jacks at the rear. There's a *Muting* on-off switch placed adjacent to a control which varies muting threshold, an *I.F. Mode* switch which varies the i.f. bandwidth (about which we'll have much more to say shortly), a button which selects mode of operation of the dual purpose signal-strength meter, an *Auto-Blend* switch which, when activated, introduces two degrees of high-blend when listening to weak-signal stereo, depending upon actual signal strength. Also hidden behind the hinged door are a mono/stereo *Mode* switch and a pushbutton which turns the dial *Illumination* on and off.

The rear panel, pictured in Fig. 1, contains antenna input terminals for 300-ohm connection and two types of connection facilities for 75-ohm transmission lines—one a coaxial connector, the other a combination of cable clamp (to retain the cable by gripping the outer shielded conductor) and screw terminal for the inner conductor. There are two pairs of output jacks, one for the fixed level outputs, the other for the variable level outputs controlled from the front panel. A pair of jacks designed for connection to an external oscilloscope comes next, followed by an i.f. output jack (which might perhaps been labelled more clearly as a detector or 4-channel output jack which is what it is). A chassis ground terminal, fuseholder and unswitched convenience a.c. outlet complete the rear panel layout.

Circuit Details

Details of the circuitry contained in the Yamaha CT-7000 would take longer to enumerate than the entire length of this report. We shall try to describe a few of the more important and innovative ones. (Refer to Fig. 2, an internal view.) Three dual-gate MOS FETs are used in the frontend, which contains two full stages of r.f. amplification and employs what Yamaha claims is the world's first *seven-gang* tuning capacitor. A seven-stage differential amplifier is used in the i.f. section together with ceramic and LC filter blocks. There are actually two separate i.f. stages; "normal" mode is for best selectivity in "crowded" signal areas, while "wideband" works best where station crowding is not a problem

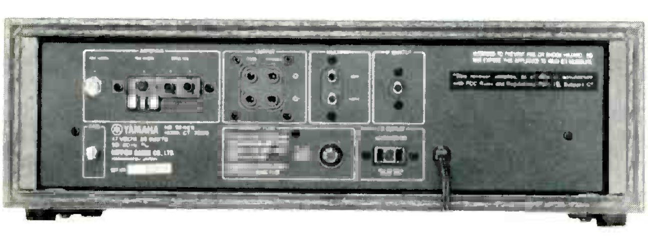


Fig. 1—Rear panel.

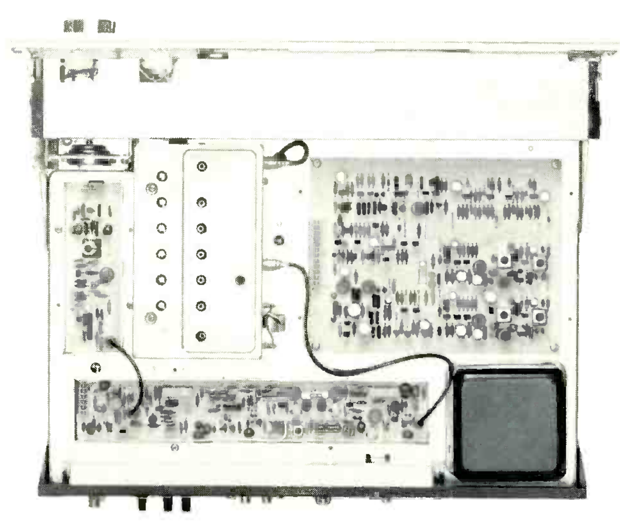


Fig. 2—Interior view.

and you want the ultimate in phase linearity, high-frequency stereo separation, and lowest possible distortion. A linear-phase wideband discriminator circuit serves as an FM detector and is, according to the maker, individually adjusted for minimum phase distortion. A dual-stage, constant-current circuit drives the detector for increased stability. The MPX decoder section employs a phase-lock loop circuit built up from discrete components (rather than from one of the available "packaged" IC PLLs now available). Negative feedback, unique to Yamaha, is used to reduce intermodulation distortion. As for the PLL circuitry, it is of the type that requires no tuned inductance circuits. Active low-pass filters are used to provide the necessary de-emphasis characteristic, and the outputs of these filters are direct coupled in a three-stage amplifying system, which includes a buffer amplifier. A separate three-stage direct-coupled audio amplifier is used to drive the headphone jack. Each circuit board used in the construction of the Yamaha CT-7000 is individually shielded in its own stainless steel cover. While we don't normally "count transistors" in evaluating the quality of a product, it may be of some interest to readers to know that this tuner contains 108 transistors, 12 FETs, 33 diodes, nine zener diodes, and seven IC circuits!

Laboratory Measurements

Again, with apologies for our test equipment, we measured an IHF sensitivity of 1.9 μV in the "normal" i.f. position and 2.4 μV in the "wide" position. Some 50 dB of quieting was attained at 2.8 μV and 3.4 μV for these two operating modes respectively. Figures 3 and 4 show our measured S/N results of 78 dB and 69 dB for mono and stereo in the "narrow" or "normal" position and 75 dB and 68 dB for the same measurements made in the "wide" setting. We suspect that our generator's limit in mono is therefore 78 dB in mono and somewhere around 70 dB in stereo, which accounts for our inability to measure claimed specs. Note too that our THD measurements at mid-frequencies (1 kHz) for both the "normal" and "wide" settings were the same, indicating clearly that we were limited by the built-in distortion of our signal generating equipment. We are therefore in no position to dispute Yamaha's claim of 0.04% in the wide position for both mono and stereo. It's very likely true!

We had previously suspected that the stereo separation capability of our stereo FM generator was about 50 dB, since that's the most we ever measured for any tuner or receiver we ever tested. Much to our surprise, we obtained a reading of 53 dB for mid-frequency separation when operating the CT-7000 in the narrow position, as plotted in Fig. 5. Interestingly, in the "wide" operating mode, although mid-band separation was slightly less, high frequency separation was clearly better, remaining above 40 dB all the way out to 15 kHz. These two plots clearly prove the importance of wide bandwidth for good high-frequency stereo separation, as well as the need for excellent phase linearity throughout the i.f. and detection system. Yamaha obviously has a good measure of both. Even within the limitations of the generator, the distortion curves shown at the bottom of Fig. 6 clearly show that increased bandwidth leads to lower high-frequency distortion in stereo. Note that when this setting was used, THD at 10 kHz in stereo was less than 0.3%—the lowest we have ever encountered. No low-pass filters were used in making this or any other measurements. There was just no evidence of the usual "beats" with the 19-kHz carrier we find with so many other FM tuners.

As for spurious, image, and i.f. rejection, our instrumentation permits us to read up to 100 dB for these specs. That's what we read with the Yamaha under test. The manufacturer claims better than 120 dB for each. What more can we say? Measuring capture ratio below 1.0 dB is tricky at best, but we managed to confirm at least one of Yamaha's claims for this important spec: 0.7 dB in the normal setting with a 100 μV input. Fantastic!

Muting level was found to be adjustable in our sample over the signal strength range from 4 μV, while stereo threshold was factory set at about 7.0 μV, a suitable point for this tuner. Transition from mono to stereo is noiseless and absolutely positive. At signals below 100 μV, blending of high frequencies as well as upper mid-frequencies takes place automatically, reducing separation substantially above 1 kHz, but reducing noise to listenable levels. Between 100 μV and 1 mV signal strengths, the auto-blend feature, if activated, blends highs to a lesser degree, retaining adequate separation while accomplishing noise reduction which makes medium-strength signals received from stereo broadcasters quite good, in terms of background noise.

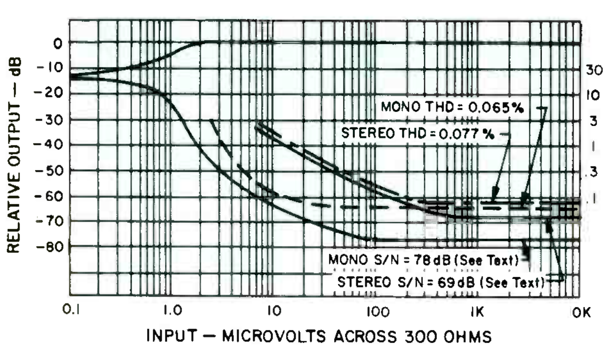


Fig. 3—FM quieting and distortion characteristics, "Normal" i.f. position.

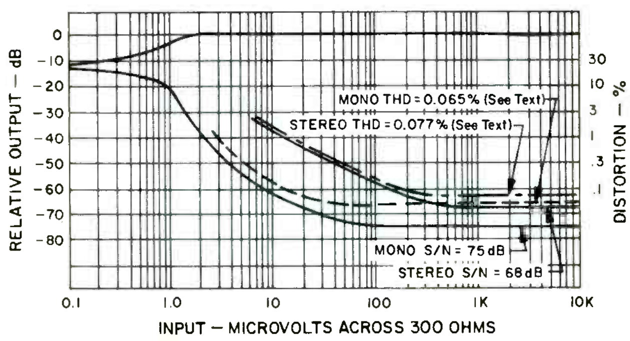


Fig. 4—FM quieting and distortion characteristics, "Wide" i.f. position.

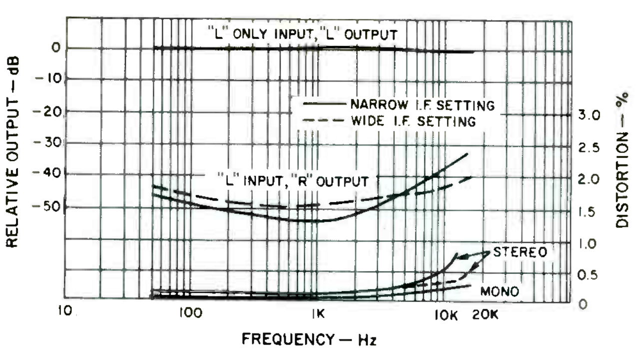


Fig. 5—Separation and distortion versus frequency.

Listening and Use Tests

We were in for a few more surprises when we got down to our listening tests and evaluations. It was then that we first began to appreciate what Yamaha calls its "auto-touch tuning" system. The large tuning knob is actually part of a capacitance switch which is activated by the user when he touches the knob. While tuning to a station, the built-in a.f.c. circuitry is turned off to enable the user to zero in precisely on the desired station signal. While this tuning is in process, the tuning indicator lamp is extinguished. As a station signal is approached, it glows softly and, when tuning has been completed and the user lets go of the tuning knob, the a.f.c. circuit comes back on and the tuning indicator is illuminated to full brightness. Normally, we have frowned upon a.f.c. circuits in general as it was our feeling that they usually contribute to distortion and non-linearities in a tuner circuit, and that given today's heat-free solid-state stable circuitry, there is no need for a.f.c. as a drift-prevention crutch. Well, this a.f.c. circuitry contributes absolutely *no* added distortion to the audio output of the Yamaha CT-7000. In fact, all our measurements in the lab were made with the a.f.c. active since we obviously didn't keep our fingers on the tuning knob while making the measurements. When we became aware of the a.f.c. feature and the way it works, we decided to check some of our measurements to see if we could fine-tune the tuner (with hands on tuning knob to deactivate the a.f.c.) for lower distortion than we had previously observed. If any improvement occurred under these conditions, it was too low for us to observe, since we were already below 0.065% in our previous mono THD readings. Nor could we detect even the slightest shift or change of appearance of the output waveform as observed on our monitoring oscilloscopes as a.f.c. was activated and deactivated. The only time the a.f.c. did anything was when we deliberately detuned the signal. Under those conditions, the a.f.c. was strong enough (when reactivated automatically) to pull the tuning close enough to optimum so that distortion was once again below 0.1%. Here is an a.f.c. circuit that doesn't "get in the way" of performance and can be of genuine assistance to the "sloppy" knob twirler who doesn't have the patience to tune as carefully as he or she should!

(Editor's Note: Yamaha says they designed the a.f.c. circuitry in their tuner sections as a final tuning compensator, to minimize distortion and increase separation.)

It goes without saying that the signals received were limited in quality entirely by the broadcast practices of the stations we received. Which, of course, brings us to the ultimate question. Should one spend $1200.00 on what is probably the best performing tuner presently available, when most stations are becoming worse and worse insofar as the quality of their broadcast signals are concerned? That depends largely upon where you are located and whether or not you have stations in your area that are careful about the kinds of signals they transmit. Is this the "best" tuner we have ever measured? Well, the previously tested Sequerra Model One does as well (we can't say "better" since in both cases our test equipment was not as good as the product demanded), but it costs more than twice as much as the Yamaha. Of course, the Sequerra has all kinds of nice things like digital frequency readout and panoramic oscilloscope displays, use of a scope instead of meters for tuning and signal analysis, and the like. There would therefore seem to be two choices for the absolute perfectionist who seeks the best in FM tuners. If you are fascinated by digital readout, scopes and the like and seek absolutely tops in specifications and performance, take home a Sequerra Model One. If you are strictly a purist, and want a super FM tuner, that's equally first-rate in its specifications and measured performance, that's flawlessly designed, simple to operate, and offers that important wide-band mode (we were able to take advantage of it even in the crowded New York metropolitan area down at the bottom of the dial), you're not likely to find a better performer than the Yamaha CT-7000.

Leonard Feldman

Check No. 92 on Reader Service Card

Koss HV/1A Headphone

MANUFACTURER'S SPECIFICATIONS

Elements: 50-mm dia. (2 in.) Decilite™ dynamic, velocity operated. **Source Impedance:** 157 ohms at 1 kHz, designed to operate with source impedances of 3.2 to 600 ohms. **Frequency Response:** 15 to 20,000 Hz. **Sensitivity for 100 dB SPL:** 0.9 V rms, sine wave, at 1 kHz; 0.5 V rms pink noise. **THD:** Less than 0.5% at 1 kHz, 100 dB SPL. **SPL at 1% THD, 1 kHz:** 108 dB. **Power Handling Capability:** 5 V continuous. **Cord:** 3-conductor, coiled, 3 meters (10 ft.) extended. **Earcushions:** Soft acoustical sponge. **Headband:** Extendable, with self adjusting, pivoting yokes and soft, padded vinyl cover. **Weight:** 285 grams (10 oz.) less cord. **Price:** $49.95.

The Manufacturer's Specifications section above just about tells the entire story, since there is very little more that can be said. However, some additional details may be interesting to the reader. For example, the foam pads are doughnut-shaped, 2¾ in. outside diameter with the "hole" opening 1-5/16 in. diameter, and the depth is ¾ in. The color is dark drown, matching the brown plastic cups which have a shiny chromed trim ring. The cups are vented both in back and around the sides, which accounts for the low isolation from external sounds. The cups themselves are 3 inches in diameter and are held by a double swivel, permitting full adjustment to the wearer's ears. The extendable portion of the band has detents to ensure that it remains in the selected position. The band itself is a sturdy steel strip, vinyl covered, with the detent mechanism encased in molded plastic terminations of the band. The cord is dark brown in color, miniature in dimension, and is coiled in a very compact form, with the three leads terminating in a detachable plug, rather than a molded one which is difficult to replace if ever one wants to change to some other form of plug. The d.c. resistance of each phone was measured at 148 ohms in the maximum-volume position and 869 ohms in the minimum-volume position, indicating that the level control is a series resistance, which would be a distinct advantage if the

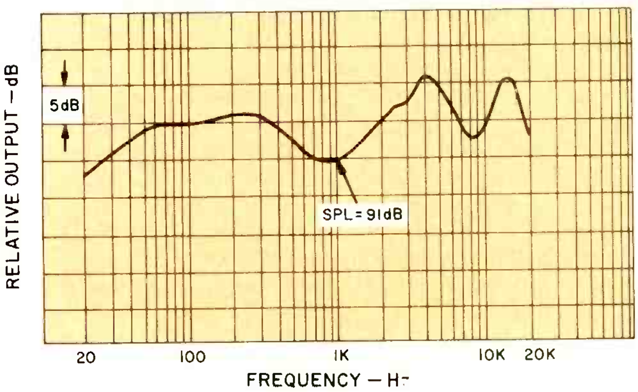


Fig. 1—Frequency response of Koss HV/1A stereophones.

phones were to be used on 600-ohm circuits. The impedance of the single phone was measured at 240 ohms. Weight is specified as 10 oz. for the phones, less cord, though the cord and plug add only another 3 oz. to the total.

Measurement Method

Determining the performance characteristics of headphones is a fairly complicated project. The phones must be coupled to a microphone by means of an artificial ear of specific characteristics—and here there is a difference of opinion. In general, the cavity between the microphone diaphragm and the headphone diaphragm is specified as 6 cu. cm for headphones and 2.5 cu. cm for hearing-aid phones. For some years, this observer has used a shop-built artificial ear which gets modified occasionally in striving for improved performance. The "ear" consists of a maple block 5 ½ inch in diameter and 6 ½ inches long—about the dimensions of the average head as far as spacing goes. Through the center is a 3/4-in. hole to accommodate an AKG C-451E condenser microphone, and on the front is a B & K metal adapter. The microphone is inserted just far enough to provide the 6-cu. cm cavity. The microphone—suitably modified for single-ended use rather than for the phantom powering normally used—has its output fed into the proper receptacle on a graphic recorder, the Justi-Meter III. For a frequency run, the source is the B & K QR-2009 test record, which has a sweep from 20 to 20,000 Hz, and it's equalized in such a manner that it can be reproduced "flat" with networks using R and C elements, as is the case with Justi-Meter III. The record is reproduced, and the output fed to an amplifier with the signal terminated by the amplifier's normal load resistance. The signal is adjusted to 3.0 volts, and the headphone driven from this 3-volt signal through 100 ohms—about normal for receiver headphone jacks.

Frequency runs are made for both phones, and a second measurement is made by reproducing a 400-Hz square wave through a loudspeaker and with the headphones off the "ear," the output of the microphone is measured. The phones are then placed on the ear, and the output again measured. This gives the isolation, in dB, with the phones on the ear, as compared with the signal without the phones. There are many uses where a high degree of isolation is desired, as where one is recording in the presence of a live source, and airborne sound from the source should be reduced as much as possible in order to hear the actual sound that is being recorded.

Performance

Figure 1 shows the frequency characteristic of the HV/1A phones, averaged between the two phones of the pair. While that may seem a little unscientific, it is actually almost the way we hear the phones. And besides, the two curves were never more than 3 dB apart over the entire range from 20 to 20,000 Hz—a remarkable feat. Furthermore, these phones offer the best—thus, flattest—response of any that we have measured with the exception of the Koss ESP-9.

Sound pressure level from these phones measured 91 dB with the 3-volt signal applied through 100 ohms, a value which is lower than the loudest ones tested, but still within the ball park. The HV/1A phones are still 3 dB louder than the Koss ESP-9, and to the average user, adequately loud for any normal purpose, though perhaps not loud enough for the rock buff. In any case, they were comfortable to wear for periods up to one hour, at least, and were not tested for more than that at one sitting.

Because of the openings in the cup and the softness of the foam pads, isolation from outside sounds was less than 2 dB, so the phones would not be suitable for recording in the vicinity of the live source, but for simply listening to music, they are excellent. *C. G. McProud*

Check No. 93 on Reader Service Card

Crown VFX2 Electronic Filter-Crossover

MANUFACTURER'S SPECIFICATIONS

Frequency Response: 18 Hz to 38 kHz ±0.5 dB with 600 ohm load. **Output:** 10 V maximum, 2.5 V rated with 600 ohm load. **Gain:** 0 to 15.5 dB. **Hum and Noise:** 100 dB below rated output, 20 Hz to 20 kHz. **IM Distortion:** Less than 0.01% at rated output. **Filters:** Separate 18-dB Butterworth high and low pass with adjustable corner frequencies. **Dimensions:** 19 in. rack mount with W.E. hole spacing, 3 ½ in. H, 5 ¾ in. D. **Weight:** 6 lbs. **Price:** $299.00

The Crown Model VFX2 is a dual-channel filter/crossover unit designed to provide continuously variable filters to perform either high-pass, low-pass, or bandpass functions in a professional, commercial or home high fidelity system. The unit is ruggedly constructed and fits the standard 19 in. rack mount, occupying only 3 ½ in. of vertical rack space and 5 ¾ inches in depth.

The front panel has four sets of range/vernier knobs for the high- and low-pass frequency setting of the filters in the two audio channels. A shadow mask pushbutton switch controls the power. The rear panel controls are a screwdriver-adjustable level (attenuator) control and a mode switch for selecting either the crossover (low pass) or filter (bandpass) outputs for each audio channel. Rear-panel connectors include a variable-gain bridging input for each channel in addition to the unity-gain unbalanced input. Output connectors (both high and low pass) are provided for inverted and noninverted (normal) modes.

For greater reliability, the VFX2 uses ¼-in. phone jacks; input is by 3-conductor jacks balanced/unbalanced or 2-conductor jacks unbalanced unity gain; output is by 2-conductor jacks. To prevent the accidental moving of the range/vernier settings, the VFX2 has a smoked plastic cover that can be attached over the front of the unit.

The VFX2 filter set is quite useful for reprocessing or playing all types of records, particularly the early 78-rpm acoustics and electrics as well as the more recent monophonic 78s and LPs, the current stereo and matrix recordings. Other equally important uses include equalizing deficient program material in a manner not possible with the usual tone controls, constructing one-third octave or other narrow-band pink-noise sources for use in acoustical measurements, etc.

Of particular interest to the professional and the audiophile is the use of the VFX2 as a high-quality dual-channel crossover with 18 dB/octave slopes and continuously variable crossover frequencies. Currently, there is renewed interest in electronic crossovers for use in bi- and tri-amplified speaker systems, with great interest in achieving true high fidelity bass performance by utilizing speakers specifically designed for low bass reproduction.

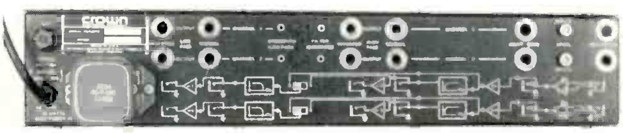


Fig. 1—Rear panel.

Fig. 2—Internal view.

The VFX2 utilizes five RC4558 dual operational amplifiers in each channel, in effect, 20 op-amps equivalent to 402 transistors, 44 diodes, and 2 zeners. The circuit appears to be a variant of a non-inverting voltage-controlled voltage-source (VCVS) design, and consists of a 6 dB/octave stage cascaded with a second 12 dB/octave section. The latter is implemented with two operational amplifiers. Results of this design approach are very high input impedances, low output impedances, and low sensitivities to drift arising from temperature changes and circuit aging.

The liberal use of operational amplifiers in the VFX2 is apparent throughout the unit. There are buffer amplifiers at the inputs to assure very high (1 megohm) input impedances which permits use of the crossover with long cables and any type of preamplifier. Use of high supply voltages for the op-amps, e.g., ± 15 volts, assures that the crossover can handle up to 10 volts without overloading. At unity gain, this type of output is far greater than is required by any power amplifier. The liberal use of op-amps is also apparent in the availability of inverting outputs. Each of these inverting outputs requires one op-amp so that four of these op-amps are used to supply the four inverted outputs. The inverting outputs, although not essential, are a very useful addition because phasing of the loudspeakers can be made at the crossover rather than changing connections at the speaker or power amplifier. Phasing becomes particularly important when the crossover is set at very low frequencies (e.g., 100 Hz) such as when used with a high quality extended-bass system. An out-of-phase condition between the low-frequency speaker and the rest of the system would appear as an attenuation in the upper bass region and would definitely be audible. By merely changing the outputs from *Normal* to *Inverted* for one of the sections (high pass or low pass), the listener can quickly determine which jack gives the correct phasing.

The filter characteristic is a third-order Butterworth, which has a maximally flat response in the pass band of the filter to a time dependent signal such as is encountered with musical material. The phase shifts for this type of filter are, however, not linear with frequency. A linear response would be an ideal case since this would lead to a constant group delay at all frequencies and, consequently, the time dependent signal would not be distorted. However, human hearing is relatively less sensitive to distortions of this kind than it is to amplitude variations with frequency, e.g., ripple in the pass band.

Measurements of harmonic distortion products using a wave analyzer with an 80-dB range showed that second harmonic distortion was always less than 0.1% at input voltages less than 9 volts and at frequencies below 10 kHz. Third harmonic and all higher order distortion components were nearly unmeasureable, i.e., less than 0.01%. The second harmonic distortion reflected the internal residual distortion of the oscillator and not of the VFX2. The frequency response measured was the same as given in the VFX2 specifications.

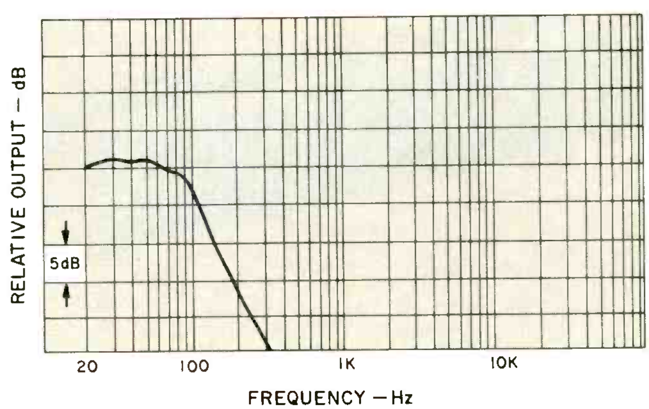


Fig. 3—Response with crossover frequency of 100 Hz.

The slopes of the filters were within 0.5 dB down to −60 dB relative to the bandpass output.

The corner frequencies as indicated on the front panel are reasonably accurate. However, for exact frequency settings, use a simple a.c. voltmeter, an audio oscillator and, if available, a frequency counter. Figure 3 shows the response curve for the VFX2 when set for a crossover frequency of exactly 100 Hz.

In using the Crown VFX2 as a crossover in our system, we noted two deficiencies. The first is a lack of level controls on the high- and low-pass sections, which means that power amplifiers used with the VFX2 should have a volume control to compensate for varying power amplifier sensitivities and loudspeaker efficiencies. The second deficiency is the omission of a summed left-plus-right output for the low-pass filter. Experience shows that for reproduction of low bass, a single monophonic bass speaker (woofer) is adequate in small rooms where the left- and right-channel speakers are not widely spaced, particularly since bass frequencies below about 100 Hz are not directional. A crossover such as the VFX2 cannot be used in this monophonic mode, but must be used with two subwoofers. However, a suitable op-amp mixer is very easy to add to the VFX2 because of its op-amp design, thus permitting the use of one subwoofer in the common mode.

As mentioned earlier, the VFX2 is very useful as an 18 dB/octave filter for record collectors, particularly collectors of early acoustic and electrical 78-rpm shellac records. The unit is inserted in the *Tape In* and *Out* circuits or between the preamplifier and amplifier when they are separate units. Figure 4 is a response curve for the playback of early acoustic 78-rpm records. The high-pass filter of the VFX2 is set at 160 Hz and the low pass filter at 3000 Hz. In practice, the low-pass filter (cutoff frequency) is usually adjusted for each recording as it is being played until the noise elements are removed or diminished appreciably. In Figure 4 the cutoff frequencies are at 160 and 3000 Hz with an attenuation rate of 18 dB/octave, while Figure 5 shows the same response, but at 36 dB/octave attenuation. Note that with such a steep attenuation, the cutoff point has shifted to about 2800 Hz at the high end. A shift towards a narrower bandpass has also occurred at the low end. In Figure 6 the high-pass filter is set at 20 Hz or turned off and low-pass cutoff frequency is set at 10 kHz, with an attenuation rate of 18 dB/octave. This is a useful setting when playing noisy LPs. As mentioned earlier, in practice the low-pass and high-pass filters are usually adjusted while the record is playing and the bandpass limits set for the least amount of noise.

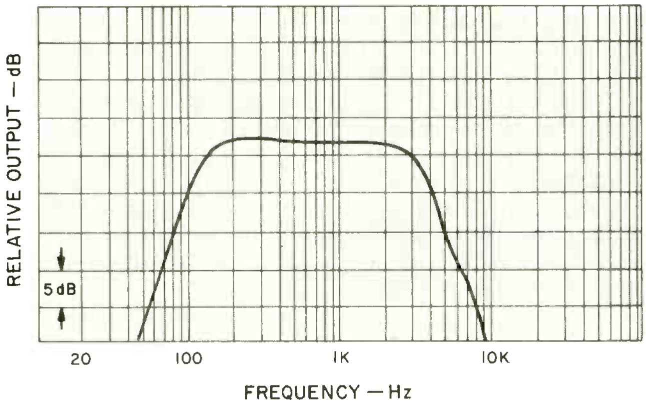


Fig. 4—Bandpass setting with cutoff frequencies of 160 and 3000 Hz, 18 dB/octave slope rate.

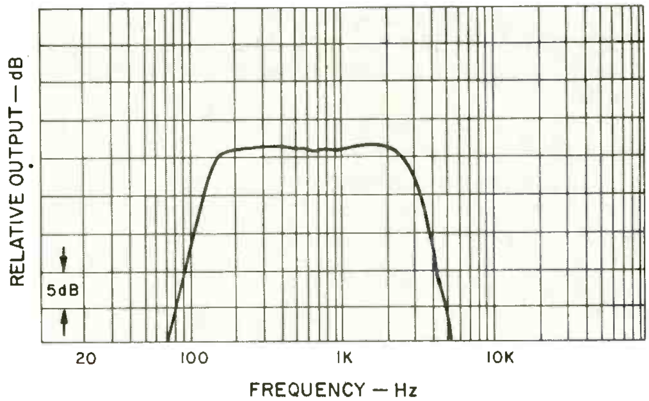


Fig. 5—Bandpass setting with cutoff frequencies of 160 and 3000 Hz, 36 dB/octave slope rate.

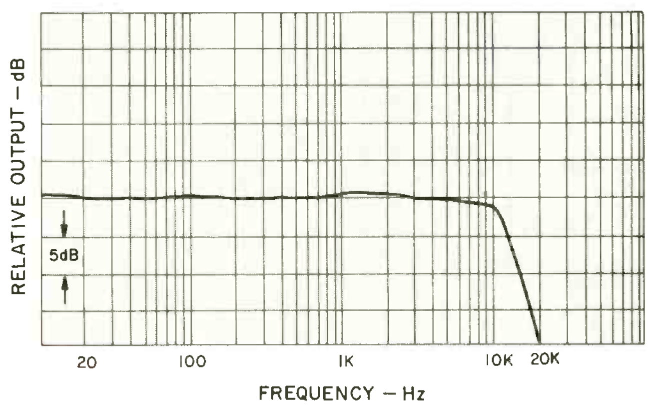


Fig. 6—Setting for noisy modern discs, high-pass filter at 20 Hz or off, low-pass at 10 kHz with 18 dB/octave slope rate.

During the past six months, the Crown VFX2, used as a crossover, has performed faultlessly in our music system. The unit is set for a low crossover frequency of exactly 100 Hz and the common mode output is fed to a 60-watt amplifier having a high damping factor. The output of the amplifier is coupled to our Janis Audio Associates W-1 subwoofer.

Use of the VFX2 crossover and the W-1 subwoofer permitted us to hear the bass frequencies below 100 Hz in an extremely clean manner and *in toto*. In the usual high quality music system, the very low bass frequencies are too often missing since very few speakers can adequately reproduce musical frequencies much below 40 Hz. This is particularly true of the low-bass pedal organ frequencies such as the 23 Hz note recorded on the Advent 5009 record (**Lemmen•Vierne•Dupre•Widor**) and the 27.5 Hz note on the ARK 10251-S (**Organ Music From Westminster**). There are many classical recordings in which the nine-foot concert bass drum, generally tuned to 31 Hz, is prominently featured, but the actual fundamental is rarely reproduced. On the Columbia MQ33172 (**Carmina Burana**), the Angel S-35430 (**Pictures at an Exhibition**), or the RCA QuadraDisc ARD1-0707 (**Citizen Kane**), to name a few, there is liberal use of this great drum, and this system reproduces it with awesome power and sonority. In the currently popular music the various bass instruments and, in particular, the ultra-low frequencies produced by synthesizers is superbly reproduced. The Crown VFX2 dual-channel filter/crossover is undoubtedly a very excellent crossover in this application.

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