FM Tuners: The Present State of the Art of FM Reception

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FM appears to be the orphan subject of audio reviewers, probably because most audio reviewers have no grounding in radio-frequency (RF) technology. This article is intended as a remedy.

Editor's Note:** Once again, Dr. Rich has come up with a tutorial type of survey article that may be a bit too technical in spots for some of our readers. Once again I say, don't worry about it. The consumer knowledge he imparts is clear and simple at all times. If you follow the circuit discussions, fine; if you don't, you'll still know what are the corner-cutting solutions, what a quality design entails, what to look for when you pay a lot of money, how to be an enlightened purchaser of FM tuners. It would be wrong to start running for cover as soon as you see an engineering-oriented paragraph. No other audio publication gives you comparable insights, so take advantage of them. Technical concepts have a way of sinking in, even when you find them bewildering at first. All it takes is the desire to know. Of course, as I've pointed out before,* ***The Audio Critic *is not "My First Book of Electricity." If you know the difference between a volt and an ohm, between ac and dc, you'll get something out of David Rich's techie marathons.*

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The importance of FM radio.

Why are we running a major article on FM tuners? FM tuners deliver a signal that is not as good as a CD, so who needs them? Music lovers need them because they offer a world of music not available from a personal CD and record collection, no mater how large. Each week, commercial and noncommercial FM stations present "live on tape" performances of this country's symphony orchestras. They also offer other live-on-tape performance series of opera, chamber music, and acoustic jazz not available from any other source but FM radio. Commercial recordings are made only of the best-known performing organizations, and they do not capture the remarkable quality and range of the musical groups throughout the nation.

Each weekday, NPR's *Performance Today* fills part of the gap left by commercial recording companies. Live performances recorded throughout this country are presented, along with insightful commentary. The quality of the programs' production and the music itself are unmatched by anything to come from the BBC or European radio. FM radio also brings educational programming, produced by NPR, PRI, and independent sources, that helps the young and not so young understand fine music. These programs include *Adventures in Good Music, Pipedreams, The Record Shelf, Saint Paul Sunday, Sunday Opera* (with Wayne Conner), and *Schickele Mix.*

Other examples of quality music and spoken-word programs that come through a tuner include *Jazz from Lincoln Center, New Sounds, A Prairie Home Companion, Selected Shorts, Car Talk, Wade in the Water, Music from the Hearts of Space,* and *Thistle and Shamrock.* Yes, a lot of trash is on FM, but the preset switches on your tuner take care of that problem

Unfortunately, this is not the best of times for radio. Commercial pressures are pushing fine classical and jazz stations off the air. Noncommercial radio stations have started to play rock music (they call it "world music" and "adult alternative") in an attempt to boost listenership and thus increase contributions. The problem here is not that noncommercial radio systems are starved for cash but that the present distribution method, where each town has its own NPR station, is so inefficient that little cash gets used to produce programs and broadcast them. The money instead flows into the pockets of those who run the local station.

All these upheavals on the FM dial may cause you to go in search of a new tuner. For example, in Philadelphia the NPR station switched to all-news to gain listeners. The music programs are now only available on a station 40 miles away, and to receive that station well Philadelphians suddenly need a good antenna and tuner.

FM has some fundamental limitations that could be overcome by newer technologies. For example, digital radio could be transmitted across the country using a satellite. It would be similar to the DSS television system. Hundreds of different channels would be available. This is a much more efficient method than the present system. Narrowcasting is possible because the signal goes to the whole country. The satellite dish could be as small as two inches and it could even fit on a car. The technology exists today, but commercial radio stations are blocking its implementation. They fear that this technology would supersede them. Since they have been given their licenses to the FM band for free, it makes no sense that they have any right to block a better technology. Unfortunately sense is not something reliably found in the halls of government.

The fundamentals.

Enough of this philosophical discourse on radio; let's get on with the business at hand. How does an FM receiver work, and how do you know if you have a good one?

The first thing to understand is that the tuner has to do two distinct jobs. One is to extract the desired signal from all the signals and background noise it is receiving. The other is to demodulate the signal so the two channels of audio information can be recovered. The circuitry should reject as much of the noise and undesired signals as possible, so that the tuner can perform its second function as well as possible.

Different signal conditions put different requirements on a tuner. If we are trying to receive a strong signal from a directional outdoor antenna which is free from multipath (reflected signals) as well as from strong adjacent (200 kHz away) and strong alternate (400 kHz away) signals, then the tuner's first job is easy. How good the station sounds will then depend on how well the tuner demodulates the signal (Job Two). Traditional audio measurements (signal-to-noise ratio, distortion, and frequency response) tell us how well the tuner is doing its second job. The only distinction with tuners is that the measurements should be across a range of antenna signal levels, since the tuner's performance degrades as the signal level decreases. Often specs are given only at unrealistically high antenna signal levels. Accuphase, a Japanese high-end manufacturer, is unique in presenting the complete information, and they even guarantee the specifications! Of course, the Accuphase tuner costs $2995.00, so it had better perform well, and the company should have no reason to hide anything by giving only minimal specifications.

Now consider the case where we are trying to receive a weak signal accompanied with strong adjacent and alternate signals. This signal is being received on an indoor antenna having a small gain and poor front-to-back ratio (see antenna article this issue), so the signal is noisy and corrupted with multipath. Now the ability of the tuner to do Job One is most important. As we shall see below, removing the adjacent- and alternate-channel signals is going to distort the signal. The low signal level will also cause the audio signal-to-noise ratio to be low, regardless of how well the tuner has been designed to do Job Two, because of the theoretical limits of FM reception. The job of the tuner under these conditions is to reproduce the signal with a performance level as close to the theoretical limits as possible.

As you would expect, some tuners do one job better than the other. For this reason, as we shall see, there is no "best" tuner. Further, it is important to understand that the tuner's specifications can tell you a lot about how well it will do the job of receiving the signal, but they are by no means complete. The signal conditions that are present at the input of a tuner are very complex, and there is no substitute for connecting the tuner into your signal environment to see how well it performs. (Just to make sure even the far-gone tweaks understand this, we are talking about noise and distortion levels here, not "slam" or "pace" or some other tweako thing.)

So why do you need to read this article? Well, if you have the need for a super tuner to do Job One well, the best thing would be to convince all the dealers in town to loan you all their best tuners. You could then pick the one that gave the best results at your home location. Unfortunately, we have not encountered such friendly dealers. So we'll try to give you some signposts regarding the small group of tuners that may be best for your signal conditions. That way you'll only have to try a couple of tuners at home. But please do not run down to the store on just our advice and purchase something you cannot return because what worked great for us may not work great for you.

You should also note that tuners have dozens of internal adjustments. If they are not set correctly, then all bets are off. We have encountered many tuners that were not adjusted properly. No doubt you may also end up with a misadjusted sample. Thus you might have the perfect tuner and never know it because it is not performing properly. Getting a tuner properly adjusted is not an easy job. Proper adjustment of tuners often requires equipment that is not in your average TV repair shop, such as a low-distortion stereo FM signal generator and a high-precision distortion analyzer. In addition, the adjustments often interact, and thus the process of adjusting a tuner properly requires a large amount of time. If you have a dealer who has the equipment to do the job (ask to see it!), you are better off purchasing the unit from him than saving 5% on mail order.

It is a lot easier to assess how well a tuner will do Job Two. As we shall see below, many manufacturers make big cost-cutting moves that are easily identifiable and measurable.

That said, we should also point out that designing a tuner is a much bigger problem than designing an amp or preamp. Most designers of tweako audio stuff would not know where to start.

The ins and outs of demodulation.

I am going to reverse things and look at Job Two before Job One. Before we discuss how a signal is decoded by the tuner, we need to know how the signal is encoded (modulated). Frequency Modulation involves assigning to the transmitted signal an instantaneous frequency (for techie folks this is the derivative of the signal's time-varying phase) that represents the amplitude of the information signal. As the amplitude of the incoming signal changes so does the instantaneous frequency. A voltage-controlled oscillator, as the name indicates, is a device that will do frequency modulation, since a voltage controls the frequency of oscillation. Now, the VCO must be very low-noise or the noise will also be transmitted. In addition, the VCO must be very linear in mapping the signal's amplitude to the instantaneous frequency. Any nonlinearity will result in distortion. You have no control of this, since the VCO is on the site of the FM transmitter. So, if the station you want to receive has poor equipment, it does not matter how good your tuner is.

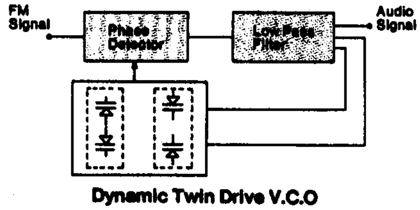
Now, it should be clear that the louder the signal gets, the further the frequency deviates from the zero-input-signal frequency (the carrier). The FCC restricts how far a signal can deviate, lest a signal from one station should wind up in the space of another. The FCC thus sets the maximum frequency deviation from the carrier frequency that a transmitted signal can have (75 kHz for FM). Since the process of Frequency Modulation is nonlinear, the spectrum of the signal at the output of the VCO can be much more than twice the frequency deviation. To understand this we need to use the Bessel function [Cook 1968].

[This section contains too much math and has been censored by the Editor.—DAR]

We can thus see that the typical FM spectrum occupies from 150 to 200 kHz. To prevent exceeding the FCC limits, stations put limiters in the audio signal path. Because loud stations attract more listeners, many stations compress the audio signal and then let the limiter come on often. Nothing can be done to fix the signal once it is mangled in this manner. Classical, jazz, and some NPR stations do try to produce a low-distortion, low-noise signal with a wide dynamic range, worthy of the tuners covered in this article. But it must be noted that even these stations will use some compression to prevent the signal from becoming inaudible on cheap equipment or in a car.

So how should we demodulate an FM signal? The optimal receiver in communication systems often involves placing the encoder in a feedback loop. For the case of FM, we would have the decoder's output connected to the input of a VCO (the modulator for FM signals). We then need a mechanism to compare the output of the VCO and the incoming signal, so that the VCO's instantaneous frequency matches the instantaneous frequency of the incoming signal. An appropriate mechanism is a phase detector which compares the phases of two signals. (Recall that the instantaneous frequency of a signal is related to the phase of the signal.) The error signal at the output of the phase detector is filtered to stabilize the loop and returned to the VCO input to close the loop. The circuit that has just been described is a phase-looked loop (Figure 1). It has been shown that the PLL is indeed close to the optimum demodulator for FM [Viterbi 1966]. As we shall see, many high-end FM tuners use a PLL demodulator. This a relatively recent development because, as stated above, the VCO must be very linear and have very low noise. Only recently have the Japanese designers been able to make such a high-performance VCO available at low cost. An interesting property of a PLL is that it can demodulate a signal with a lower signal-to-noise ratio at the antenna terminal than other types of demodulators [Panter 1965].

Figure 1: Phase-locked loop FM demodulator with high-performance VCO.



One advantage of a PLL is that the bandwidth of the demodulator can be easily changed. For the most faithful demodulation of the signal, the loop should have a wide bandwidth. This is the condition we want for a good, clean signal, but if the incoming signal is noisy, then we want to reduce the loop bandwidth to reduce the effect of the noise [Gardner 1979]. Another key advantage of the PLL is that the output signal level is independent of the amplitude of the incoming signal (the reduced signal level may reduce loop bandwidth, however; see [Gardner 1979]). Since noise and interfering signals will cause the incoming signal to have significant amplitude variation, we want the detector to ignore these variations. The specification that tells us how well a tuner rejects AM signals is the AM rejection. AM suppression is defined in terms of the relative disturbance caused by amplitude modulation when the carrier is simultaneously amplitude- and frequency-modulated [IEEE 1975]. This is a key specification to determine how well the tuner rejects interference such as fading, multipath, airplane flutter, lightning, electrical equipment noise, etc. Super tuners should be in the 80 dB range. Care must be taken with this specification because it gets better at higher RF signal levels, and most data sheets do not give the level at which the test was performed.

The pulse-count demodulator is another high-performance circuit (Figure 2 shows an implementation by Accuphase). Based on theoretical analysis this circuit

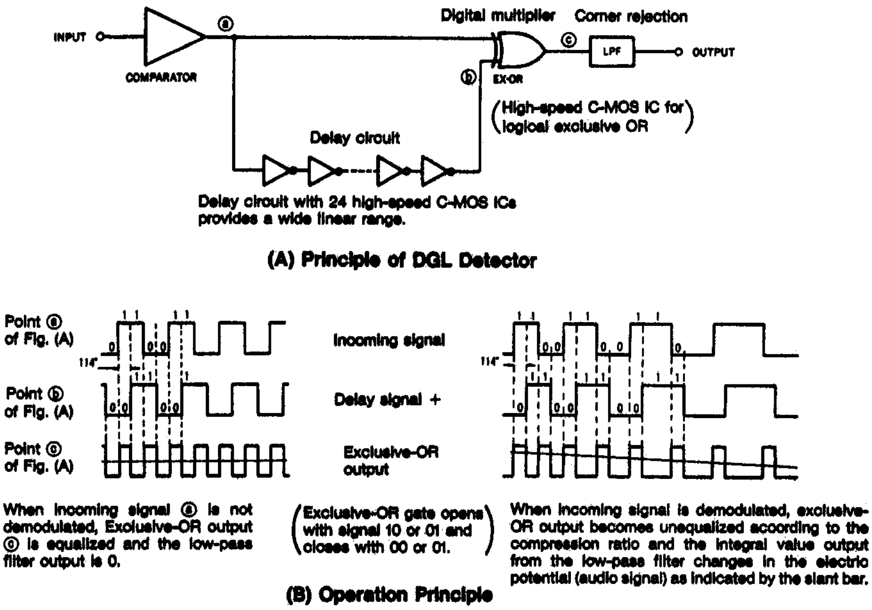


Figure 2: Pulse-count detector (or demodulator) in an advanced implemetation by Accuphase.

should not be as good as the PLL but in practice it is much simpler to implement, and thus the practical results may meet or exceed PLL performance. It is inherently a very linear and very low-noise design. It is also easy to see how it works. Every time the incoming signal crosses from a positive value to a negative value (a zero crossing), the circuit generates a pulse of fixed duration. (In the Accuphase implementation shown in Figure 2, a zero crossing from negative to positive also generates a pulse.) If the instantaneous frequency is high, the pulses bunch together. If the instantaneous frequency is low, then the pulses are farther apart. Filtering the pulse stream with a lowpass filter averages the pulse count over time, and this yields the demodulated FM signal. This works because closely spaced pulses have a high average value and pulses farther apart have a lower average value. Like the PLL, the pulse-count detectors have very good AM suppression because only the zero crossings of the incoming signals are used. The problem with this type of detector is that it requires a double-conversion IF stage that reduces the IF frequency from the typical 10.7 MHz to a lower frequency in the 2 MHz range. This adds significant complexity, although it has an additional advantage of improving selectivity in some cases, since it may be easier to build a selective bandpass filter if it has a lower center frequency [Carson 1990]. Double conversion is required so that the frequency deviation of the signal is a significant percentage of the carrier frequency. The higher this ratio, the lower the output of the detector. Note that you cannot directly convert to a 2 MHz IF because the image rejection of the tuner would become very poor (see below).

We have one more detector to look at (we are leaving out a bunch of older or less popular types). This detector is significant not because it is very good but because it is very common, being (yes, you guessed it) cheap to make. The detector is the quadrature detector. In the quadrature detector a two-step approach is used. First the signal is differentiated, then it is sent to an AM demodulator. Both stages of the process are difficult to realize with precision. In the quadrature detector a simple tuned circuit near resonance provides a nearly constant phase characteristic that is used to approximate a delay line. This delay line is then used to approximate a differentiator. Thus we have two levels of approximation to the differentiator [Clark 1971]. A balanced mixer is then used as a synchronous detector for the AM signal that comes from the differentiator.

What is a balanced mixer? It is a simplified form of a circuit called an analog multiplier. A true analog multiplier takes in two analog signals and multiplies them together. It can work in the voltage, current, or charge domain. Typically it is voltage-in-voltage-out, so if port A was at 2 volts and port B at –3 volts, the output of the analog multiplier would be –6 volts. The term *balanced* is used to indicate that none of the signals at the inputs of the mixer will appear directly at the output as distortion components. Harmonics of the multiplied signals may appear in a balanced mixer, and this differs from a true analog multiplier. These harmonics result because one input port of the mixer may significantly distort the incoming signal. Even-order harmonic distortion components from this port do not become involved in the mixing process in a balanced mixer, but the odd ones will. In an unbalanced mixer all harmonics are involved. The balanced mixer is easy to integrate, so the only external component is the tuned circuit. That is what makes it cheap. Unfortunately, given the two layers of approximation, distortion performance is never as good as with the PLL or pulse-count detector. Worse, the tuned circuit must be tuned to give the correct delay or the distortion is even higher (see the Denon review below). As many as three different adjustments may be on the tuned circuit, and these adjustments require the equipment your TV repairman does not have. If the frequency deviation becomes large, the approximations break down completely and significant distortion can result. Signals with interference on them require a detector with a wide bandwidth if the interfering signals are to be rejected [Panter 1965], so the limited performance of the quadrature detector presents a problem here. In addition, the quadrature detector cannot reject AM on its input, so circuits preceding it must limit the signal to provide any rejection of AM. Signals with large levels of interference will have a large AM component, so we are in trouble again.

Stereo FM explained.

Once we have demodulated the FM signal, we must now recover the two stereo channels. Several methods are used to modulate the stereo signal. I will describe the one that I think is easier to understand. A switch alternately samples the left and right channels at a 38 kHz rate. Of course, the signals must be bandlimited before sampling, just as in digital audio. The band limit is 15 kHz. Now, a mono receiver will reproduce both channels, since it will just average the alternate samples together. In a stereo receiver another switch is synchronized with the original switch, so when the switch at the transmitter is connected to the left channel the switch in the receiver will also be connected to the left channel. Then the switches at both the transmitter and the receiver go to the right channel (Figure 3). A filter follows the switches to remove out-of-band signals, again for the same reason this is done in digital audio. Often it's a pretty sloppy filter, so tuners often have frequency-response specs that allow a loss of 1 dB at 15 kHz. The "subcarrier product rejection" is the specification that indicates how well the filter is rejecting these out-of-band signals.

So far so simple, but how do we synchronize the switches? What is done at the transmitter is to send a 19 kHz pilot tone that is one half the switching frequency. At the receiver a PLL (this is another PLL, not to be confused with the PLL detector above) locks onto this signal. We want the output of the VCO to run at twice the speed of the 19 kHz pilot, since that is the switching frequency required. A digital divide-by-2 is placed between

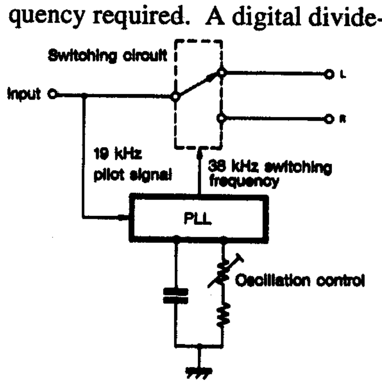


Figure 3: Basic configuration of a stereo demodulator.

the VCO and the phase detector, so the VCO runs at 38 kHz instead of 19 kHz. The 38 kHz signal from the PLL drives the switches. Note that a simple VCO that generates a square wave can be used to drive the switches. Also, this VCO does not have the linearity requirements of an FM detector because it has to run at just one frequency.

Often the VCO operates at a much higher frequency, with larger divider chains separating it from the switches. This allows the production of a signal with less phase noise. (*Jitter* is the term we would use if were examining the VCO's output in the time domain. Time-domain jitter and phase noise are mathematically related, as discussed in Bob Adams's article in Issue No. 21.) Phase noise on the pilot signal will lead to increased noise and distortion at the audio outputs, as well as reduced channel separation. As an example of very high-frequency VCO operation, most Sanyo chips run at 456 kHz. That allows them to use an external mechanical resonator, which will have even lower phase noise. This is a high-Q mechanical element similar to a quartz crystal but made of different material; it is usually used at frequencies where quartz crystals are uneconomic. Also, the resonator is superior to an RC-based VCO in three important areas: it is more accurate and does not need adjustment; it is not sensitive to temperature changes; and it has excellent long-term stability. Because of this the lock range of the PLL can be greatly reduced. That allows the loop bandwidth to be narrowed, making the PLL less sensitive to noise on the incoming signal. A design by Delco Electronics [Manlove 1992] uses a different architecture to accomplish a similar thing. The centerpiece of the Delco system is an analog/digital PLL that is too complex to discuss here. Delco reports they reduced the lock range of the PLL from a typical value of 800 Hz down 2 Hz.

Of course, some of us can hear the 19 kHz pilot, so it has to be removed. The el cheapo approach is to use a lowpass or notch filter at the output of the demultiplexer. Thus we have more stuff in the signal path to create frequency-response errors and add distortion. The better approach is to remove it by cancellation. Since we have made a replica of it with the PLL, this is relatively easy, but first the square-wave signal from the VCO must be filtered, since the transmitted pilot tone is a sine wave and the replica signal is a square wave. Then the signal level must be trimmed, using a pot to exactly cancel the pilot tone. The Delco folks have a clever way of doing this without external components to filter the signal or trim the amplitude or phase of the summer [Manlove 1992], but it is too complex to go into here.

We have a big problem with the multiplex decoder outlined above. The switches that are selecting the left or right channel will also demodulate any signal that corresponds to a harmonic of 38 kHz. Such signals will be present as a result of spurious frequencies created by adjacent and alternate channels and by RF intermodulation. Adding to the fun is the fact that subcarriers containing Muzak or digital data such as paging signals are also sent along with the FM signal you are listening to. *[The politically corrupt and cynically compromised FM stereo standard bulled through the FCC bureaucracy in 1961 is the reason for this.—Ed.]* These adjacent-channel frequencies, when demodulated, cause beat interference called "birdies." (If you listen to a weak stereo signal on your tuner during a quiet passage you will know why they call it birdies.) Removing the interfering signals with a sharp filter before the switches will clean up the birdies (they call it an antibirdie filter) but will also cause phase shifts that prevent synchronization of the switches to the transmitted left and right channels. That results in distortion and reduced channel separation. A specification called "SCA rejection" indicates how quiet the tuner is when the Muzak subcarrier is present.

A better approach is to use a sinusoidal 38 kHz signal and an analog multiplier, instead of the square-wave-driven switch. No demodulation of out-of-band signals can occur in such an arrangement. The problem is that you need to have a high-performance analog multiplier. These are open-loop devices; they have no feedback loop in the multiplier core to improve the accuracy of multiplication. Multiplication accuracy to 0.1% is thus very difficult to achieve. Obviously, any inaccuracy in the multiplication will give rise to distortion. It is difficult to design analog multipliers to have good noise performance. The Rotel RHT-10 tuner reviewed below is the first tuner to my knowledge to do this. (Something did not turn out quite right with the design because an antibirdie filter is still in the signal path.)

Pioneer also uses a sinusoidal 38 kHz signal. They combined this with a pulse-count detector (see explanation above), since the output of the pulse-count detector is binary. They connect the input of the switch to the 38 kHz signal and switch it with the binary signal from the pulse-count detector [Ishida 1984]. The problem with this approach is that it only works with a pulse-count demodulator. Pioneer appears to have extended the approach to the use of PLL and quadrature detectors in their current product, but how it works using these demodulators has not been disclosed. (I tried to get an answer from the engineers at Pioneer, but their response was to refer me back to the original patent they had filed, which discusses only the use of pulse-count detectors. Either my question did not get translated into Japanese correctly, or the engineers just do not want to disclose what they are up to.)

Sansui uses another approach that allows for full integration of the demultiplexer (another term commonly used for the stereo demodulator or stereo decoder). Sansui attempts to approximate a sine wave using Walsh functions [McGillem 1974]. The Walsh functions are a set of periodic rectangular pulses which, when combined at different amplitudes, can approximate an arbitrary periodic waveform. The Fourier series does the same thing but with a set of periodic sine waves. What makes the Walsh functions ideal for integrated systems is that summed scaled rectangular pulses are easy to generate on an IC. It turns out that two scaled Walsh functions are all that is needed to eliminate all the harmonics in the PLL output, up to the 8th harmonic [Takahashi 1985]. To generate the correct Walsh waveforms, the VCO in the Sansui chip runs at 304 kHz. Since Sansui has fallen on hard times, you cannot purchase this interesting tuner (TU-X701) with the above chip and a nice PLL FM detector anymore.

Chips by Allegro, Sanyo, and Sony use a similar scheme, although they have not published the detail that Sansui has. Antibirdie filters are typically not used in front of these chips. The team from Delco Electronics [Manlove 1992] also uses a similar idea, although the complete circuit implementation is significantly different. Delco uses more periodic rectangular functions to get the job done. This removes even more harmonics.

You may have noticed that this is the third time I am mentioning this Delco chip, so you might want one in your tuner. Unfortunately, to my knowledge, Delco does not sell chips, so this chip can be found only in GM car radios. Yes, the FM demultiplexer chip in your car is quite possibly better than the chip in your tuner.

Noise in an FM system rises with the audio frequency [Taub 1971]. For this reason the signal transmitted at the FM station is pre-emphasized, so that the signal rises 6 dB per octave above 2 kHz. A matching de-emphasis circuit is in the receiver. This de-emphasis circuit counteracts the rise in the noise. The problem with this scheme is that the signal levels at frequencies above 2 kHz must remain low, or else the station will overmodulate. Acoustic music and speech luckily have spectral densities that decrease with frequency above 2 kHz.

In a stereo transmission, the mono signal occupies the same spectral band as it would if a mono-only transmission were taking place (see explanation above). That was an FCC requirement in order to insure that a mono receiver could receive a stereo signal. The information about the stereo signal is contained higher up in the spectrum (Figure 4). These new sidebands come from the sampling action of the switches at the transmitter, which run at 38 kHz. It can be shown (but not here, says the Ed.) that these sidebands contain the signal L–R. When you add L–R to the mono signal, you get the left channel. Subtract it and you get the right channel. It can be shown that the sampling action of the switching circuit in the

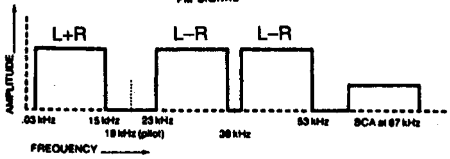


Figure 4: Demodulated spectrum of a stereo FM signal.

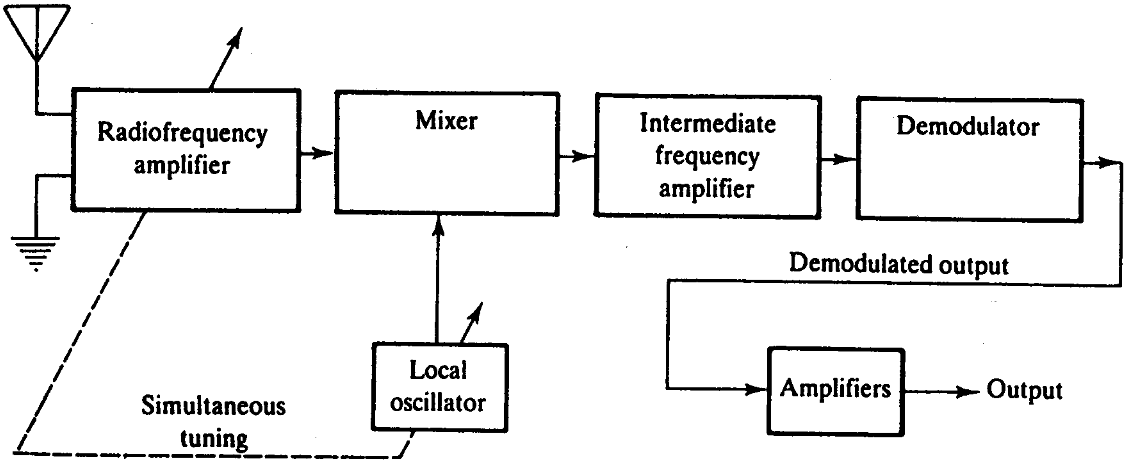


Figure 5: Simplified block diagram of a superheterodyne receiver.

multiplex demodulator does this function, but do not worry if this is not clear. What you need to know is that a mono FM signal has a bandwidth to 15 kHz, but a stereo FM signal goes out to 53 kHz. Now recall that noise increases with frequency in FM. So the L–R information in the 23 to 53 kHz portion of the composite signal has a lot more noise. How much poorer is the signal-to noise-ratio of the stereo information? An order of magnitude poorer, 22 dB worse to be exact [Taub 1971].

If the incoming signal is strong and clean, the theoretical signal-to-noise ratio is very high and this 22 dB performance deficit is not significant, since other noise sources dominate. The signal-to-noise ratio then should, ideally, become the value produced by the tuner when it is receiving a mono signal. If the signal is weak, then we can hear every bit of the 22 dB noise penalty when we switch from stereo to mono. It turns out that the ear tolerates fairly low channel separation at higher frequencies, so a blend circuit is often used to reduce the channel separation (and hence the noise) in the upper frequencies. Pioneer takes this one step further. They have a set of blend circuits, each in a specific passband. A circuit looks at the noise in each passband and adjusts the amount of blend up or down in response to the noise in that band. This dynamic adjustment gives a better stereo image, with less noise on weak signals. The downside of this system is that it is very complex.

Carver (the man and the company) varies the amount of L–R signal used, depending on how different in level and content L is from R [Feldman 1982]. Carver creates two signals, L/R and (L+R)/(L–R), and uses them to sense how different the levels of L and R are. Carver also uses more L–R when the leading edge (fast, short-term information) of a signal is detected. Carver says he uses a psychoacoustic phenomenon known as the precedence effect. When leading-edge information occurs, it is critical to the localization process. At other times, Carver uses a phony L–R signal concocted from the low-noise L+R. He does this with what he calls a phase randomizer and a spectral shaping circuit. It would appear that all this should not sound at all like the original stereo signal, but in practice it works very well, at least to my ears. Is the Pioneer approach better than the Carver approach? Stay tuned (no pun intended); we intend to tell you when we have both tuners in our laboratory back to back.

Pioneer has one more trick up their sleeve. They noticed that most of the noise and interference occurs above 38 kHz. They also determined that they only need the information from 23 kHz to 38 kHz (for the techie crowd, this is the lower sideband of the L–R as seen in Figure 4) because the upper and lower sidebands contain the same information. Now Pioneer uses only the information in this band (for the techie folks, they use a single-sideband demodulator). Unfortunately, this process introduces some distortion in actual practice, and the THD of the decoder is in the 2% range. That is very listenable, however, in the context of a weak FM radio signal.

The front-end circuitry.

OK, it is now time to look at the front end of the tuner, since we started in the middle of the signal path. The job of the front end is to provide the *sensitivity* to detect the presence of the desired signal and the *selectivity* to accept the selected station and reject all other stations. Figure 5 shows a superheterodyne receiver. The RF amplifier, the local oscillator, and the mixer are called the front end. In a superheterodyne receiver the incoming RF signal is mixed with a signal from the local oscillator, changing for each station, in such a way that the output has the same center frequency for all stations. This approach allows for excellent selectivity because the tuned circuits in the IF strip that reject interference are *fixed* tuned circuits. This results in much sharper cutoffs than if the filters had to be variable, as they do in the RF stages. In addition, it is possible to have higher-gain amplifiers at lower frequencies.

The RF stage amplifies the weak signals to a level at which the mixer can work properly; thus it increases sensitivity. If the signal were strong you would think you could bypass the RF stage, but you cannot for two important reasons. The first is local oscillator radiation. If the RF stage did not exist, the signals from the local oscillator could get back into the antenna through capacitive and inductive coupling. Since the local oscillator runs between 98.6 MHz and 118.6 MHz, this is a good way to cause interference right in the FM band.

The second thing the RF stage does is improving image rejection. In a superheterodyne receiver, a problem at the mixer occurs because the mixer can translate signals both above and below the local oscillator frequency, and it thus is possible for an undesired station to get translated to the IF frequency. In most FM receivers, the local oscillator is set above the incoming RF signal; thus the difference frequency contains the desired signal at the output of the mixer. The image frequency is above the local oscillator frequency. The image frequency is thus the desired RF signal's frequency plus twice the IF frequency. Now, if the IF frequency is 10.7 MHz, the image frequency can never be in a broadcast FM signal. That is not an accident; we do not want an FM broadcast signal to become an image signal. The FM band is 20 MHz wide (87.9 MHz to 107.9 MHz), and since the image signal is at the RF signal frequency plus *twice* the IF frequency, any IF frequency greater than half the width of the FM band will work. They put the IF frequency as close as possible to this limit, because the lower the IF frequency the easier it is to build a selective filter and create high-gain amplifiers. (For homework show why the IF frequency of an AM receiver is 455 kHz). So we are concerned with what is on the air between 109.3 MHz and 129.3MHz. What is there usually consists of not very large signals because these are frequencies for aviation activities. Note that if the local oscillator had been chosen to be below the RF frequency, the images—in this case the incoming signal frequency minus 21.4 Mhz—would come from TV channels 3 to 6, at 60 MHz to 88 MHz, each TV channel being 6 MHz wide [Cook 1968]. So now you know why that is not done. The specification for a broadcast tuner that tells you how well it rejects images is called the "image-response ratio" or "image-rejection ratio" [IEEE 1975].

Unfortunately, other image interference is possible, because the mixing signal often contains harmonics. These harmonics also generate sum and difference products that can move undesired signals to the IF strip. TV channels 9 to 13 (186 MHz to 216 MHz) are sources from which second-order images, caused by the second harmonics of the oscillator, can get into the IF strip. Filtering the RF signal beforehand and attenuating the undesired one will prevent these images from contaminating the desired signal. Third- or higher-order images are rarely a source of interference because they are very highly attenuated by the tuned RF amplifier [Cook 1968].

Another interference is caused by spurious response. Spurious response is caused by nonlinearity of the RF amplifier or mixer. This can cause intermodulation products which again can fall into the IF band. Under worst-case conditions, with a heavily overloaded RF front end, a station may appear at a number of places on the dial. A common mechanism producing a spurious signal is the second harmonic of the local oscillator beating against the second harmonic of an RF signal. The IEEE test for this [IEEE 1975] is called the "characteristic-frequency test." In this test the incoming frequency is set to 103.35 MHz and the receiver is tuned to 98 MHz. The tuned circuit before the first RF stage helps here also by filtering out undesired signals before they get to the tuner.

Other scenarios involve two incoming signals and the local oscillator together forming a spurious signal [Cook 1968]. The IEEE standard test for RF intermodulation involves testing for the condition where the second harmonic of one interfering signal at 98.8 MHz is mixed with another interfering signal at 99.6 MHz. The mixed signal is at 98 MHz. The test is reported as the "two-signal spurious-response ratio." The frequencies of the two signals are too close together for the RF stage to provide any filtering. The test thus checks for how well the RF and mixer stages are designed by checking to see if they generate intermodulation distortion.

The IEEE standard allows the worst-case result of the tests for characteristic frequency and two-signal spurious response ratio to be reported as a single number called the "spurious response ratio," representing the poorest of these measurements. Unfortunately, this is often not followed in specification sheets, and it appears that the characteristic-frequency spurious response ratio is then published as the spurious response ratio, period.

In the case of a very strong station, it is important that some method of signal attenuation be supplied at the

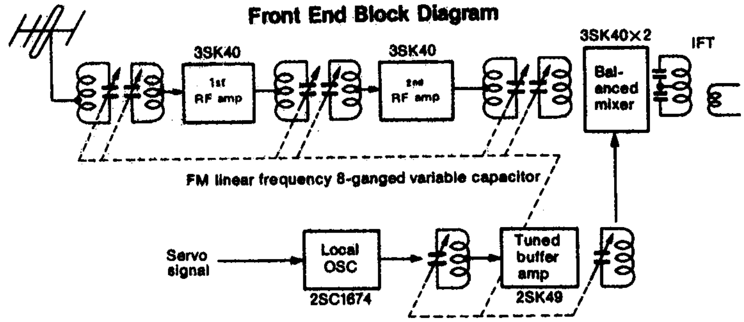


Figure 6: RF stage of the now defunct Technics "Professional Series" ST-9030 FM stereo tuner (from the good old days when front ends were front ends).

input of the RF stage to prevent overload. In some tuners this is automatically controlled by an automatic gain circuit. Other tuners have manual controls that enable the attenuation. The manual controls are nice because sometimes the AGC gets things wrong. Consider the case of a relatively weak-signal channel adjacent to a very strong signal. The AGC would turn off the attenuation to gain up the signal, but in doing so it would cause the strong adjacent-channel signal to overload the RF stage. Smart AGC circuits, such as used in the National Semiconductor LM1865, can be designed to look for strong adjacent-channel interferennce before increasing RF gain.

In the good old days, super tuners had super front ends, For example, the Technics ST-9030 had an eight-gang tuning capacitor and two RF amplifier stages (Figure 6). The ST-9030 had an image and spurious-response rejection of 135 dB because of this complex RF stage. The eight gangs added up the following way: One was for the local oscillator, and one was for a filter after the local oscillator to remove harmonics. Each of the RF stages had a double-tuned filter preceding it, as did the mixer. Each double-tuned section needed two gangs. A double-tuned section provides a flatter passband and a steeper rolloff rate than a single-tuned section. Today you get one RF amplifier and the equivalent of three or five gangs. The result is that some of the best tuners today have characteristic-frequency spurious-response and image-rejection ratios of only 80 dB. In addition, they may lack sufficient RF gain, especially when operated from an indoor antenna. The Magnum Dynalab "Signal Sleuth" is an extra stage of tuned RF gain which can be added to any tuner. This device can solve some difficult signal problems that some of the modern super tuners cannot handle. The Signal Sleuth is reviewed in this issue.

Another important parameter that the RF amplifier should satisfy is constant input impedance, independent of frequency. The need for this is explained in the indoor antenna reviews found elsewhere in this issue. As in the case of antennas, the deviation from the ideal input impedance can be given as a voltage standing-wave ratio (VSWR). This is how Accuphase gives it (as usual they have the most complete set of specifications). Most manufactures do not give the specification at all.

In modern receivers, all the tuned circuits are tuned with voltage-controlled capacitors that use a diode assembly. This is called a varactor. The voltage range to tune the varactor should be large, so that the time-varying information signals present at one end of the varactor cannot significantly change its capacitance. Signal-dependent changes in capacitance can give rise to modulation components. Early varactors had this problem, but voltage swings for tuning over the FM band are now 20 V for modern varactors, and the information-signal swing is much smaller than this.

The voltage for the varactors comes from the loop filter that forms yet another PLL. The local oscillator is the VCO of this PLL. A set of digital dividers after the VCO is used to tune the tuner. The output of the dividers is connected to a phase detector. The other input of the phase detector comes from another digital divider that has as its input a crystal reference oscillator. Under microprocessor control, the dividers are set so that the local oscillator runs at the correct frequency to move the desired signal to 10.7 MHz. The voltage at the output of the PLL loop filter also adjusts the tuned circuit's RF stage and mixer to the correct position to receive the desired station. Clearly, no noise should be on the voltage that is connected to the varactor in the VCO because this will give rise to phase noise in the local oscillator. Since the PLL is a closed-loop system, the loop filter cannot be designed to provide an arbitrary amount of filtering because this will cause the PLL to go unstable [Gardner 1979]. The best way to insure a quiet varactor input voltage is to run the phase comparator at as high a frequency as possible. The undesired signals at the output of the phase comparator can then be more easily removed by the loop filter because they are high in frequency.

Early digital tuners with slow IC technology had a problem running at high phase-comparator speeds and so they acquired a bad reputation. The phase comparators also had problems with dead bands that made things worse. (A dead band is a region where the phase detector gives no change in output even when the phase difference of the incoming signal changes.) Modern ICs run at higher frequencies and use high-performance phase comparators; thus modern tuners do not suffer a performance penalty when they use digital tuning. RF noise from the microprocessor and other digital circuits also caused noise problems in early frequency-synthesized tuners. These problems do not occur in modern designs because the problem is understood and techniques to minimize the interference are known. Linearity of the VCO is not a problem in digital tuning because only one frequency has to be synthesized to receive a given station.

The problem with digital tuning is that the station may not be at the exactly correct frequency. This is especially true of stations retranslated to other spots on the dial on FM cable systems. It is thus desirable that a fine-tuning mode be available to detune the receiver.

The mixer is a relatively simple circuit but very hard to optimize, according to Richard Modafferi, who has done state-of-the-art mixer designs. Obtaining sufficient dynamic range is the really tough design challenge. Two design parameters must be optimized to have good dynamic range: (1) noise levels in the mixer must be very low so that weak signals are received; (2) the mixer stage must be able to handle strong signals in a manner that does not allow intermodulation distortion to occur, which is a more difficult requirement here than in the RF stage because the mixer is not just an amplifier; it is also performing frequency translation. A lot of other design challenges also exist in mixer design, such as making sure the mixer does not load down the RF stage or interfere with the proper operation of the local oscillator. A balanced mixer will usually perform better than a single-ended one. In a balanced mixer, even-order harmonic distortion components from the local oscillator do not become involved in the mixing process. This reduces spurious interference. The downside of a balanced mixer is that it is more complex. Also, it must be designed very carefully if is to be truly balanced at RF frequencies.

The IF section amplifies the desired signal and removes the interfering signals before the FM detector. One thing the IF strip should *not* do is directly pick up any signals at its center frequency. This can happen because the IF strip has a lot of gain. 10.7 MHz happens to be kept relatively quiet by the FCC because of this problem. Correctly shielding the IF stage also helps. The IF rejection specification is a measure of how well the IF strip rejects this signal. In the IEEE test called the "IF response ratio" [IEEE 1975], an FM-modulated signal with a carrier at 10.7 MHz is applied to the antenna terminals.

The gain requirements for the IF strip are quite high. At the antenna input a signal may be as small as 1 μV. The FM detector may require a signal of 1 V. That adds up to 120 dB of gain. The RF section is going to have about 30 dB of gain, so the IF strip needs to have a gain of 90 dB. Fortunately the stage does not have to be linear, so feedback is not used and the gain is achieved by cascading individual stages of open-loop gain. This does not prevent the possibility of oscillation in the IF strip through parasitic coupling between the output and input of the strip. Because of all the gain in a tuner, the designer must always be on guard against such problems.

In between each stage is a tuned filter. This is important because, as we gain up the desired signal, we also gain up the undesired signals. The filters reduce the amplitude of the interference before it is sent on to the next stage of gain. In this manner the desired signal amplitude becomes larger than the interfering signal.

For the FM signal to pass through the IF section without picking up distortion, it is important that none of the sideband components of the FM signal be shifted in phase relative to one other. (No, all you tweaks, this has nothing to do with the ear's sensitivity to phase. In FM we have modulated the signal so that the instantaneous frequency, and thus the phase, contain the information about the baseband signal. That is why phase integrity is important in FM signals.) Now we have a problem, since a sharp IF filter will reject the interfering signals, but it is also going to affect the phase. This is because you cannot have a sharp transition band in a filter without significant phase delay [McGillem 1974]. Consequently, a good tuner offers selectable IF filters. Wide filters give low distortion in the demodulated audio signal. Narrow filters give good rejection of interfering signals but have poorer audio performance. Some tuners even have an extra narrow mode for bringing in very difficult signals, at the penalty of even more audio signal distortion.

Modern filters in a tuner are mechanical resonant elements whose input is driven by, and whose output is picked up by, an electromechanical transducer, such as a piezoelectric element. The quartz crystal is a simple mechanical filter. In the IF strip multipole elements, often called ceramic filters, are used to give flat passbands and sharp rolloffs. These filters perform much better than the old double-tuned circuits they replace. Another advantage is that these filters do not have to be tuned. In the wide mode two filters may be in the signal path; in the narrow mode four filters. In extra narrow we can see five or more filters, and the passband of each of the filters will be decreased at the cost of phase distortion. The Pioneer Elite F-93 wins the contest of the most mechanical filters in the IF, with eight filters. The old McIntosh MR-78 had a filter that placed transmission zeros at the alternate- and adjacent-signal positions. That appears to be a good idea. I cannot figure out why it is not used in other tuners. (The high cost of the filters is the likely reason.)

Pioneer came up with a very innovative solution to the "narrow filters distort the signal" problem. They used this solution in the F-91 tuner. It is such a good idea that it could have gotten someone a Ph.D. (Indeed, a related concept did get someone a Ph.D.—see [Rich 1991].) What the engineers at Pioneer observed was that, although the spectrum of an FM signal is wide when *averaged over time,* at any instant in time the signal has just one spectral line at the instantaneous frequency. So, if you build a *time-varying* filter with a very narrow bandwidth that moves with the instantaneous frequency of the desired signal, then you can eat your cake and have it too. The very narrow bandwidth removes the interferers, but the time-varying nature of the filter insures that the FM signal is not distorted. To move the filter around, a complete auxiliary IF strip designed in the conventional manner and a PLL FM detector are required. We now have a chicken and egg problem, since the conventional receiver has to place the passband of the time-varying filter in the correct place. This is what limits the performance of the system.

One very interesting (at least to this author) variation on this is to replace the bandpass filter with a notch filter. Then the stronger signal is removed, but a weaker cochannel interfering signal may now be revealed. Further information on this is in—you guessed it—[Rich 1991].

Surprisingly, the F-91's tracking filter was dropped from the newer Pioneer Elite F-93. The reason given by Pioneer was that the circuit needed very high-precision (read high-cost) parts and that performance nearly as good was achievable without them in the revised tuner. The F-93 reduces the distortion caused by the fixed narrowband IF filter by compensating for it in the FM demodulator.

Sony uses a variation on the tracking-filter theme that involves dynamically varying the position of the RF and mixer filters in response to the signal level at the output of the FM demodulator (which represents how far the instantaneous frequency has moved from the carrier frequency). This is easier to implement in the RF and mixer stages than in the IF because these filters are already tunable. One would think the large signal delay in the IF strip and detector would prevent this from working well, but Sony is shipping tuners with this thing, so it must do something positive.

One wonderful property of FM is that two cochannel signals can be very close in amplitude and yet only the stronger will emerge from the FM detector. This is the capture effect in FM systems. It does not exist in AM systems. A specification that tells you how much stronger a signal has to be to *capture* the receiver is the capture ratio. In a super tuner it is as small as 1 dB. The capture ratio is also important for a situation where multipath exists, since a multipath signal looks like a cochannel signal to the receiver [Panter 1965].

When multipath or a significant interferer is present, the amplitude of the IF signal can vary significantly [Panter 1965]. This does not affect the performance of the FM detector provided it has good AM rejection—strike one for the quadrature detector. When multipath or a significant interferer is present, the composite instantaneous frequency variation can be very wide; thus the FM detector must have a very wide linear range to accommodate these excursions—strike two for the quadrature detector. The widebanding works because the inband signal at the output of the FM demodulator turns out to be the desired signal. The interfering signal gives rise to distortion products that are not present inband. If the discriminator is not wideband enough, then inband distortion products may appear in addition to the out-of-band products [Panter 1965]. The bandwidth requirements rise steeply as the power ratio of the desired to the interfering signal closely approaches 1. Thus a detector which does not have wide bandwidth will have a poor capture ratio.

The quadrature detector does not strike out because of the limiter stage. The ideal limiter takes an input that is varying in both frequency and amplitude and provides an output of constant amplitude that retains the instantaneous frequency information of the transmitted signal [Cook 1968]. In essence it is a high-gain amplifier that is driven to clip. At some small signal level, the limiter will not have enough gain to clip, and the circuit will just amplify the signal and pass on the amplitude variations. Cascading multiple limiters reduces the signal level that is required to saturate the limiter. So the limiter removes the amplitude variations that the quadrature detector is sensitive to and gives the receiver its AM rejection, usually 50 to 60 dB. Of course, if the detector has some AM rejection, like the PLL or pulse-count detector, then the limiter is icing on the cake and the AM rejection goes to 80 dB.

The limiter can also help to reduce the bandwidth requirements at the detector in the presence of an interferer. This is done by the use of a cascaded narrowband limiter [Baghdady 1955]. What Baghdady found was that when the desired and interfering signal enter the limiter, the energy of the interfering signal spreads out in its spectrum. If you then send the signal through a narrowband filter, some of the interferer's energy will be removed. Continuing to do this a number of times will reduce the energy of the interfering signal relative to the stronger signal at the detector's input. This then eases the bandwidth requirements of the detector.

We have now arrived at the input of the FM detector, which is where we started at the beginning of this article. With a basic understanding of how an FM tuner operates, we can move on to the reviews. It should be noted that we did not hook all the fish in this first survey. Promising tuners not included are the **Pioneer Elite F-93** and **Sony ST-SA5ES** referred to above. Also promising are the very expensive **Accuphase T-109** that was also mentioned above and the bargain-priced **JVC FX1010TN** and **Yamaha TX-950**. Unfortunately, one of the great tuner manufactures of the past, Kenwood (great because they also design ham radio equipment and thus have a large and experienced design team) no longer brings any super tuners into this country.

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How I Evaluate FM Tuners

By Richard T. Modafferi
Technical Consultant to *The Audio Critic*

I've designed and built RF devices (transmitters, tuners, shortwave radios, cable TV amplifiers, etc.) for nearly 50 years, and thus one may assume in this regard that

(1) I know how to design and test RF devices, or

(2) I'm good at fooling people that I'm good at (1) above, or

(3) we're all ignorant and it doesn't matter.

Perhaps a combination of the three items above is closest to the truth. Regardless, the Editor has roped me into the job of testing, but not reviewing, tuners for ***The Audio Critic***. An explanation of my test procedures is thus necessary.

An FM tuner is really just a radio. By this I mean that a tuner should work like a radio and receive as well as possible all signals present at its antenna terminals. Reception problems and the solution thereof are the largest challenge facing an FM tuner designer. Radio circuits in an FM tuner must work over a very large dynamic range, up to 120 dB in the best designs, as for example a lower limit of 1 μV and an upper limit of 1 V. In addition, FM tuner circuits have an astonishing amount of voltage gain, up to 130–140 dB in some designs. My MR-78, for example, has a voltage gain in its RF-IF circuits of about 137 dB! One μV at the 75Ω antenna input produces 7 V at the input to the detector bridge, at an impedance of 52Ω. This is gain, folks. I'm not impressed by the bragging that goes on about about the dynamic range in the latest digital stuff. Tuner designers play with bigger numbers! The challenge of making tuner circuits work well over these big numbers involves a lot of study, experiment, and sleepless nights. The reward at the end, however, is a super tuner and good FM reception.

I share a mountaintop with two broadcast stations, an AM on 1430 kHz 600 feet away, and an FM on 92.1 MHz only 138 feet away. Big brother on 92.1 measures 1 V at my tuner antenna terminals. Another station on 105.7, across the valley, comes in at 0.25 V. Weak signals exist on 91.3, 91.7, 105.3, and 106.3 MHz. I try reception of the aforementioned weak signals in one of my tests for a tuner's reception capability.*

Tuner RF Performance Tests

(1) Outdoor tower-mounted high-gain antenna system. The tuner is connected to this antenna, and its ability to receive weak signals on 91.3, 91.7, 105.3, and 106.3 MHz is noted. The tuner must cope with a 1 V signal on 92.1 and a 0.25 V signal on 105.7.

(2) Selectivity test. I attempt reception of a weak NPR station on 91.3 (75 miles away), adjacent-channel to a local NPR station on 91.5 MHz (4 miles away). So far, I've found only three tuners that can receive 91.3, and these are the Onkyo T-9090II, my MR-78, and the Accuphase T-109.

(3) Indoor antenna reception test. The tuner is connected to a three-quarter wavelength "gamma-matched" vertical antenna, and reception quality on local stations is noted. Reception should be free of spurious responses. In some cases reception of fairly weak distant signals is possible.

(4) FM generator strong-signal test. The tuner is connected to an FM generator, and 1 kHz monophonic harmonic distortion is observed as signal is increased from 1 μV to 1 V. Distortion plus noise should drop from the initial value of 3–5% (–30.5 dB to –26 dB) at 1 μV to below 0.25% (–52 dB) at 10–15 μV and remain low up to the 1 V antenna input level.

(5) *$2f_1 \pm f_2$ test.* Two stations close in frequency will produce a pair of spurious signals whose frequencies are given by this formula. For example, 105.1 and 105.7 MHz combine to produce a "spurious" at 106.3. I look for it. This is the classic "stations coming in all over the dial" or "at wrong places on the dial" syndrome. The $2f_1 \pm f_2$ test is by far the toughest of the spurious-response tests. The front end in Figure 6 may do well on the other spurious-response tests and yet do very poorly on this one because the extra RF stage could cause intermodulation to get worse.

Baseband IM Distortion Test

The tuner is connected to a Sound Technology 1020A FM signal generator. A stereo signal of 1000 μV is applied to the left channel, modulated 100% at 10 kHz. Output from the right channel is observed. A spurious output tone of 1 kHz will appear, and in good tuners this tone should be 60 dB or more below 100% 1 kHz modulation (0 dB reference level). I devised this tough test during development of the MR-78 25 years ago. In 1968, only one tuner, the Marantz 10-B, could pass this test. The MR-78 passes. Today, almost any decent modern tuner passes also.

Residual-Junk Listening Test

This time I play music into the tuner from the Sound Technology 1020A. I modulate the left channel only and listen to the right. A perfect tuner would produce silence. Good tuners yield a clean low-level output. Even a small amount of distortion is audible because you listen to the "residual" channel, already down 40 to 50 dB because of separation. Here, 0.1% (–60 dB) distortion, as referred to the other channel's output level, will be only 10 to 20 dB below the residual and hence clearly audible! Surprisingly, some tuners do well on this test, implying that their stereo decoders have very low distortion, and also that the tuner's entire circuitry has very good linearity.

This ends my regular test regime. I also perform some additional distortion and separation measurements, generally in order to verify the manufacturer's specs and to check tuner alignment. I do a touch-up alignment if needed.

David Rich notes:

Here in Audio Critic country, in Eastern Pennsylvania, reception problems abound. The principal classical-music station is 40 to 60 miles away (depending on the test site). Unfortunately, a 50,000-watt rock station is alternate channel to this station and about 5 to 15 miles away. This is the type of condition that separates the real tuners from the pretenders.

Over at the commercial-free low end of the dial, adjacent-channel stations are coming from all around the area. Some come from local 100-watt stations or 10-watt translators. The local NPR and college stations are quite clean but more often than not they play rock. Only when they play what a government-supported station should, not Classic B Sides, can we find out how well the tuners do Job Two. The marvelous Mercer County Classical Network comes in on a new translator. Before that it was 70 miles away and one of my principal worst-case test signals for this survey.

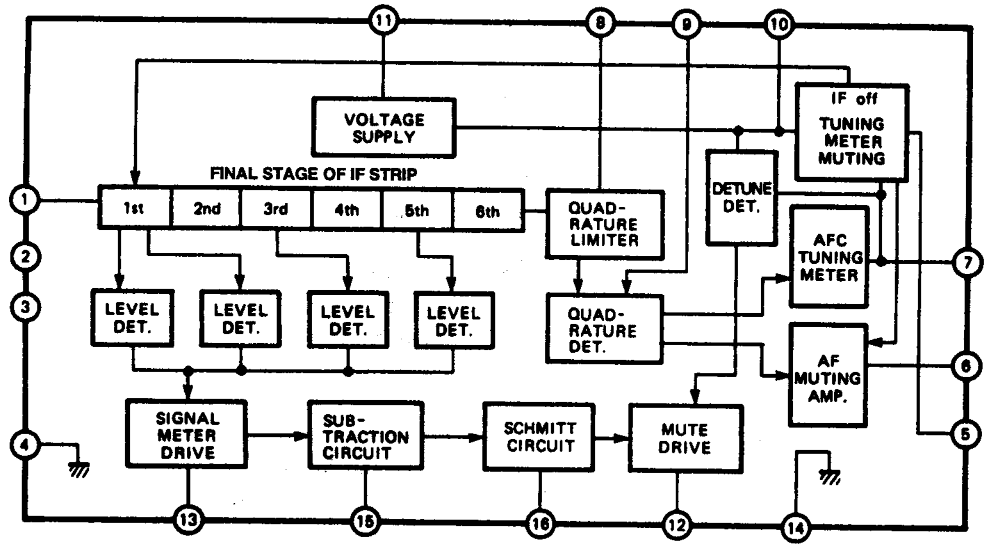

Figure 7: Block diagram of the ubiquitous Sanyo LA1235 chip (with the iniquitous quadrature detector).

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Denon TU-680NAB

Denon Electronics, a division of Denon Corporation (USA), 222 New Road, Parsippany, NJ 07054. Voice: (201) 575-7810. Fax: (201) 575-2532. TU-680NAB AM/FM stereo tuner with remote control, $600.00. Tested sample on loan from manufacturer.

The RF section of this tuner is reasonably well-done with the equivalent of five gangs, one at the RF input and two at the mixer section. The mixer is fully balanced—but to no avail, as we shall see below. Another pair of tuned stages is in the local oscillator. One is used to set the local oscillator's frequency; the other filters the signal from the local oscillator as part of a buffer stage.

The IF strip starts with a single discrete stage and uses two integrated amplifiers between the ceramic filters. Two filters are used in the wide mode. Four filters are used in the narrow mode, with an additional integrated IF amplifier also switched in when narrow mode is engaged.

The final stage of the IF strip and the FM demodulator are in the ubiquitous Sanyo LA1235. This chip is the most popular (it's real cheap!) IF detector. It uses a quadrature detector design, as shown in Figure 7. Now, Denon knows that a quadrature detector just won't cut it in a high-end tuner. They used an advanced PLL detector in the last generation of products that did not cost more. Two tank circuits in the quadrature detector must be adjusted for the tuning to be on channel center and for minimum distortion (try to get that one done by your local TV repairman), and the unit came to us incorrectly adjusted. Another adjustment involving a distortion measurement must be performed on the tank circuit connected between the mixer and the IF stage. The remaining adjustments should be within the scope of a competent repair person.

The multiplex decoder is another IC, the Allegro ULN3827. Allegro is a U.S. company that designed this chip primarily for car audio, and most of its specifications are not state-of-the-art for monolithic multiplexer chips. It uses a 608 kHz mechanical resonator in the VCO and has a 19 kHz pilot canceler that requires no adjustments. Walsh functions are used to improve rejection of spurious signals like the SCA carrier. Surprisingly, Denon still chooses to use an antibirdie filter. A dynamic blend circuit is included, which activates when the noise ratio becomes high. Most autoblend circuits (clearly a function you need more in a car than at home, where you can flick the blend switch) activate the blend on signal level, not noise level, and can blend a weak but clean signal accidentally. The PLL loop bandwidth in this chip is made wide for fast acquisition when no stereo signal is present and then is made narrow to increase noise immunity when the stereo signal is captured.

Another nice feature of the Denon is that the channel-separation adjustments are made individually for the narrow and wide modes to insure the best possible performance in the narrow mode. A pair of passive filters follows the multiplex decoder to remove the pilot tone and other higher-frequency products. The outputs are buffered with JRC NJM4558 devices.

The AM section is quite complex and performs much better than most AM tuners. It has a wider audio bandwidth and lower distortion than traditional AM tuners, as well as sophisticated noise-reduction circuits. It also has excellent sensitivity. (We can confirm that Rush Limbaugh sounded better on the Denon, but we performed no controlled listening tests.) If you have an AM station in your area that broadcasts something of interest, this may be the tuner for you.

A single 12 V supply powers the analog stages. Two 10kΩ resistors tied to the 12 V supply and a 100 μF filter capacitor form the analog ground path. You cannot get any cheaper than this. General construction quality is that of a mass-market product.

The FM RF front end has a very low noise figure. The 50 dB RF quieting occurred at 10.5 dBf. Most tuners are 20 dB noisier at this signal level. RF intermodulation performance was not so good, however. Strong stations prevented reception of nearby weaker stations that could be received by other tuners in this survey. This indicates that the RF stage (and/or mixer stage—remember that mixer stages are very hard to design) does not have adequate dynamic range, although the cause could not be identified by me from the schematic. Poor dynamic range can cause intermodulation of signals at the RF stage's input, since the stage becomes nonlinear.

Measured 1 kHz THD out of the box was –57dB for a 30,000 μV signal at 91.1 MHz. This fell short of the specified –60 dB. In addition, minimum distortion was not achieved when the unit was tuned dead on. The Modafferi 10 kHz stereo IM test gave a result of –56 dB on our test sample; on another sample tested by Modafferi the result was –69 dB. Clearly our sample was misaligned as delivered, and that includes the quadrature detector. Channel separation from 50 Hz to 15 kHz was ≥39 dB in wide mode and ≥33 dB in narrow mode. It never met the specified 50 dB at 1 kHz, measuring 47 dB in wide mode, but the full-band results are very good. Frequency response in stereo just made the +0.5 dB, –1.0 dB strip given in the manufacturer's specification sheet. That *could* be audible and should be tighter in a $600 tuner.

The Denon proved to be an average performer when presented with good signals and did poorly under difficult signal conditions. The AM performance is truly exemplary, and the circuitry to achieve this must have added to the cost of the unit. If AM performance is important to you, consider this unit. Others should pass it by, as I believe the less expensive Harman Kardon TU9600, JVC FX-1100BK, Sony ST-S550ES, and Yamaha TX-950 should at least match its FM performance at much less cost. The slightly more expensive the Rotel RT-990BX and Onkyo T-9090MKII will blow it away on FM. (They have no AM, so again if that is important, look into the Denon.)

Harman Kardon TU9600

Harman Kardon Incorporated, a Harman International Company, 80 Crossways Park West, Woodbury, NY 11797. Voice: (516) 496-3400. Fax: (516) 496-4868. TU-9600 "active tracking" AM/FM stereo tuner with remote control, $449.00. Tested sample on loan from manufacturer.

We have had this not very new but still current model in-house for some time, but for some mysterious "organizational" reason it never got to be measured by Rich Modafferi. *[Mea culpa.—Ed.]* I don't want to wait, however, to tell you what you can get for $449.

The front end and IF sections are pretty typical of units in this price range, but the FM demodulator and stereo decoder are anything but. The tuner has a state-of-the-art PLL FM demodulator, but to make it work at this price they could not design a high-linearity VCO. So they got real smart (yes, it is patented) and realized that the output of the VCO in a PLL must always track the incoming signal accurately, even if the transfer function from voltage to frequency of the VCO is not very linear. This is because of the phase detector used in the PLL.

What Harman Kardon does is to take the output of the VCO and send it through a Sanyo LA1235 with its quadrature detector. No amplitude-modulation problem occurs in the quadrature detector because the VCO output is a constant and it's big. This combination does not allow the low-distortion properties of the PLL FM detector to be exploited but it does allow its excellent performance under poor signal conditions (including large amounts of AM) to be taken advantage of.

The loop bandwidth of the PLL is made narrow to exploit the circuit's special signal-demodulation properties under poor signal conditions; as stated in the main article, this may cause some distortion. The VCO output is thus an FM-modulated version of the desired signal cleaned up by the PLL. The quadrature detector then demodulates the cleaned-up signal.

OK, what more could you want for $449? Well, how about the top-of-the-line Sanyo multiplex decoder called the LA3450? It looks like a second source for the Sony CXA1064S chip used in the top-of-the-line Sony tuners. No other multiplex decoder has better specs than the Sanyo LA3450; it some cases it is significantly better. Yes, it has a pilot-tone canceler. Yes, the VCO runs at 456 kHz and requires no adjustment because it uses a mechanical resonator. Yes, it has no antibirdie filter because of a birdie noise-reduction system that appears to be similar to the Sansui approach. The decoded signal then goes, not to some cheap op-amp, but to a discrete amplifier designed in the Krell style. Other high-end touches can be found, although there are big limits at this price.

Measurements of the TU9600, plus more design details, will be published in the next issue.

McIntosh MR7084

McIntosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. MR7084 AM/FM stereo tuner, $1500.00. Tested sample on loan from manufacturer.

This somewhat baffling tuner was sent to us much later than the other units reviewed here and will therefore have to wait until the next issue for full coverage; meanwhile the Editor discusses it briefly elsewhere in this issue as part of the feature article on McIntosh Laboratory.

Onkyo T-9090II

Onkyo USA Corporation, 200 Williams Drive, Ramsey, NJ 07446. Voice: (201) 825-7950. Fax: (201) 825-8150. T-9090II Quartz Synthesized FM Stereo Tuner, $790.00. Tested sample on loan from manufacturer.

This tuner has been with us for 11 years, with the Mark II revision appearing in 1988. When it first appeared it was the DX (distant reception) engine for the masses, with Richard Modafferi's McIntosh MR-78 being the only thing that could outperform it. Today the T-9090II may be the ultimate super tuner for bad signal conditions, as the MR-78 is history and McIntosh is unwilling to do an update on it. (Engineering has done a couple of designs but marketing has canceled full-scale development. The marketing folks say they would sell exactly six of them because the price would be in the middle five figures.) The only challenger in current production is the Accuphase T-109. On the difficult real-world signal tests presented to the tuner by Richard Modafferi, the Accuphase has slightly better spurious-response characteristics, but its selectivity in not quite as good. Since the Accuphase sells for almost four times the price of the Onkyo, we can definitively say that the Onkyo is the cost-effective DX engine.

The Onkyo has a number of nifty operational features. They include a variable gain stage that can even be operated by remote control. Two antenna inputs are switchable on the front panel. I have one on my FM cable; the other goes to the indoor antenna (cable FM does not carry many local stations in my area). Because crosstalk between the two antenna inputs is only 40 dB, this may not work in all cases if you are trying to receive a weak signal off the indoor antenna at a place on the dial where the cable company has placed a strong signal. Speaking of cable, the tuner shows center-tuned conditions with three LEDs. Since the tuning increment is 25 kHz, you can tune to the station's center even if it is not being transmitted correctly. (Richard Modafferi points out that he has seen frequency-synthesized tuners with 10 kHz tuning increments. That is the increment needed to solve in every case the problem of an off-center station.) Off-center stations often occur on FM cable, when stations are moved around the dial to a different place than the over-the-air broadcast position. Provisions for connecting an oscilloscope to the tuner to check for multipath are also included. If you want to get depressed, try using this to view the quality of your cable FM on a scope. The antenna inputs are some nonstandard things that will take push-on connectors but not the screw-on type. This is a royal pain in the *tochis*.

The tuner automatically selects the stronger signal of the two antenna inputs. Signal strength is displayed in dBf. The measurements do not appear to be to extraordinarily accurate; although the signal-strength circuit is quite sophisticated, as explained below, our sample expanded changes for high-level signals and compressed changes for low-level signals. Even so, this is a very good way to get a feel for the relative strength of the signals. The tuner will automatically decide the control parameter, such as IF bandwidth, RF attenuation, blend activation, etc. It appears to do a reasonably good job, but you can override the tuner's choices and store your own choices if you wish. One nice feature that is not present on this tuner would be a recording calibration tone indicating the 50% modulation level. The record companies have apparently put enough of a scare into the hardware companies for the latter to drop any feature that would make things easy for the home tapist.

The RF section is the equivalent of six gangs. One is at the RF input and two are in the mixer section. The mixer is fully balanced. Another pair of tuned stages is in the local oscillator; one is used to set the oscillator's frequency, the other filters the oscillator as part of a buffer stage. The RF section can be bypassed through a separate tuning element that is coupled directly to the mixer stage. Local oscillator reradiation would appear to be a problem with this arrangement, but FCC regulations test for such things, and this unit could not be offered for sale if it failed the test. The ability to bypass the RF stage when hot signals are present is nice because RF overload problems due to inadequate dynamic range cannot occur if the stage is not in the circuit.

The IF stage for the wide and narrow modes starts with a single discrete stage and uses an integrated NEC μPC1163H amplifier between the ceramic filters. The final IF amplifier and limiter stage is the Sanyo LA1235, but the FM quadrature detector stage of the chip is used only for muting circuits. Two filters are used in the wide mode and four filters in the narrow mode. When the narrow mode is switched in, another μPC1163H IF amplifier is put in the signal path. The supernarrow mode switches in a completely separate IF strip. One double-tuned LC stage, five ceramic filters, two discrete amplifiers, four μPC1163H integrated amplifiers, plus another LA1235 form the supernarrow IF strip. A diode switch selects which IF strip's output will be sent to the FM detector. Another μPC1163H is used after this switch.

The signal-strength circuitry is quite complex. The Sanyo LA1235 has a signal-strength indicator output that is derived by measuring when its internal IF sections limit (it has six stages of amplification—see Figure 7). The earlier the stages limit, the stronger the signal. Unfortunately, even a relatively weak signal may limit the first stage because it has been through a significant amount of amplification in the IF strip that precedes the LA1235. For that reason, signal-strength meters using the LA1235 meter output are of limited usefulness. In the Onkyo, additional circuitry is included to test if earlier stages in the IF strip have gone into limiting. Additional places where limiting is tested for are at the output of the first amplifier in the IF strip, at an intermediate point in the supernarrow IF strip, and at the end of the supernarrow IF strip. The meter circuit has four—count 'em, 4—trim pots. Other tuners give you one. These are set for the correct indication at 5 dBf, 45 dBf, 85 dBf, and 105 dBf. Levels in between are less accurately displayed. A total of 50 active or passive components are used in signal-strength meter circuit. The muting circuit is also quite complex to prevent false mutes. This circuit monitors outputs from the LA1235 chips in both the standard and supernarrow IF strips, as well from the PLL detector. A total of four adjustment pots are used in the circuit to insure it works reliably.

The FM detector is a PLL. This is the most nearly optimum FM demodulator, and that's what Onkyo needs to build a super tuner. The phase detector is the standard double balanced mixer topology made from a four-diode bridge and a pair of broadband transformers. Dual varactors and two MOSFETs form the VCO. The loop filter uses an NJM4560 op-amp. The other section of the op-amp also buffers the loop filter output and does the de-emphasis equalization. This composite audio signal goes directly to the output stage in mono! It never sees the multiplex decoder. It is a sign that this tuner is intended for DXing deep fringe stuff, where the chance of getting anything usable in stereo is very unlikely. When stereo decoding is used, the loop filter output goes through a separate buffer circuit that includes the antibirdie filter.

Stereo decoding does not get as much attention as the rest of the design. An NEC μPC1223C multiplex decoder is used. The chip requires an antibirdie filter, as we saw above, and uses a low-end VCO design, namely an RC oscillator at 76 kHz that must be adjusted. An open-loop pilot-tone canceler is included, but no adjustments are provided to insure optimal cancellation. The age of the design really shows here, as much better multiplex decoder chips are available now. Individual channel-separation adjustments are provided for all three IF modes. Passive lowpass filters follow the decoder chip. The output buffer is an NJM4560. Two electrolytic capacitors are in the composite audio signal path in mono and four are used in stereo.

Construction quality of the unit is at the level of Japanese mass-market equipment; given all the stuff in the unit and the reasonable price, you would not expect anything else. The power supply to the audio section is ±15 V. A total of 8 voltage regulators is used. Adjustment of this tuner should be doable by any competent service technician because no distortion measurements are needed. The PLL detector does not need such adjustments.

The first unit sent to us by Onkyo performed very badly because it was either defective or totally misaligned. We sent it back. A second unit appeared to be working correctly and yielded the following test results: THD at 1 kHz in stereo was –62 dB in wide mode. That is very good, but it misses Onkyo's spec by 12 dB. The THD at 1 kHz was –36 dB in supernarrow mode. The Modafferi 10 kHz stereo IM test came out at –60 dB in wide mode, –64 dB in narrow mode, and –41 dB in supernarrow mode. Channel separation was ≥37 dB across the band in the wide mode. At 1 kHz it was 50 dB. That is 5 dB short of the specification given by Onkyo; the more important across-the-band separation figure is, on the other hand, 4 dB better than the specification. With the blend function enabled, channel separation is reduced to 18dB at 15 kHz. Frequency response in stereo fits into a ±0.4 dB window, which is better than the spec.

If you don't want to spend more than $1000 on a tuner and have difficult signal conditions, the Onkyo T-9090II is more likely to get you a listenable signal than any other tuner we have tested.

Rotel RHT-10

Rotel of America, P.O. Box 8, North Reading, MA 01864-0008. Voice; (800) 370-3741. Fax: (508) 664-4109. RHT-10 FM stereo tuner with remote control, $1499.90. Tested sample on loan from manufacturer.

Rotel and Harman Kardon are the only companies I know of that are using audiophile circuit-design attributes in the design of tuners. The Rotel RHT-10 takes things further than Harman Kardon has attempted. As we shall see, Rotel dropped the ball a couple of times, but I know of no other tuner that attempts to do what Rotel has done here. The RHT-10 is not designed to be a super tuner. Instead, the design is for signal conditions that are at least good.

In contrast to a spaceship like the Onkyo tuner, the Rotel has very few control buttons on the panel. Unfortunately, the presets are available only on the remote. Because of this I found this tuner to be a real pain in the neck to use (where did I put the remote??!!). Audio output is very high. It is fixed and must have been set here so the tuner would work with the Rotel RHC-10 passive control unit. It is a pain because the thing is 6 dB higher than CD standard level. You switch to tuner and then dive for the volume control on the preamp.

The RF section is the equivalent of six gangs, one at the RF input and two in the mixer section. The mixer is, surprisingly, not fully balanced. Another pair of tuned stages is in the local oscillator. One is used to set the oscillator's frequency, the other filters the oscillator as part of a buffer stage. As in the Onkyo, the RF section can be bypassed through a separate tuning element. How local oscillator reradiation is suppressed is unclear, but the FCC would not allow the unit to be sold if this problem existed. The IF strip starts with a single discrete stage and uses two integrated amplifiers between the ceramic filters. Two filters are used in the wide mode and four filters in the narrow mode. No super narrow mode is included. As I said above, this is not designed to be a super tuner. No instructions on how to adjust the tank circuit at the output of the mixer are included in the service manual. This adjustment typically involves a distortion measurement. Adjustment of components in the RF and IF stages can be done by any competent technician.

The last stage of the IF strip and the FM decoder is the el cheapo Sanyo LA1235. For $1500 I thought I was going to get something other than a quadrature detector. Two adjustments are required for the detector, one involving a distortion measurement. The presence of the quadrature detector in this design is like a large zit on a pretty face because everything else is done so well for the reception of good-to-excellent incoming signals. The output of the detector passes through an antibirdie filter, and then the high-end fun begins.

At first blush things look normal, with a Sanyo LA3433 multiplex decoder chip on the board. It uses a 456 kHz ceramic filter in the VCO for narrow PLL lock range and low phase noise. But this chip is used only to generate the 19 kHz pilot tone for the pilot-tone canceler and the 38 kHz square-wave signal for the multiplex decoder. The rest of the circuits on the chip are not used at all. Instead, high-quality op-amps and discrete circuits are used wherever the audio signals actually travel.

First the 38 kHz output of the LA3433 is converted to a sine wave by means of a tuned circuit. This circuit is outside the PLL feedback loop, so adjustment of the tank is critical if the phase relationship between the composite FM signal and the 38 kHz sine wave are to be maintained. Two inductor adjustments do this. Pioneer puts the tuned circuit in the PLL loop when they generate a 38 kHz sine wave for their proprietary decoder. That eliminates the need to trim, but a custom IC design is required to do this. Rotel decided to do it with standard parts to save the cost of designing a special IC. Luckily, a simple level-peaking adjustment sets the inductors accurately, but they can drift after adjustment.

An NE5532AN chip (which is a lot better than what you will usually find in tuners' audio sections) is the active circuitry used to filter the 38 kHz sine wave. Another NE5532AN sums the incoming composite signal and the 19 kHz pilot tone together for pilot-tone cancellation. An inductor forming part of a bandpass filter that filters the pilot-tone canceler signal is adjustable The inductor sets the phase of the pilot tone, and a pot sets its amplitude. These are simple adjustments that can be made with a voltmeter, but again they can drift.

So now we have a composite signal with a pilot tone canceled and a 38 kHz sine wave. Multiply them together using a pair of analog multipliers and you would have birdie-free stereo. Unfortunately, low-distortion analog multiplier chips are expensive, so nobody has used this approach until now. Rotel does not use the analog multiplier chips, but they do take the plunge by making a discrete version of an analog multiplier instead. First the composite signal is converted into a pair of balanced currents. This is done with a clever two-transistor discrete circuit that uses feedback. These currents (I_{ee}) each enter the tails of two differential pairs. The 38 kHz sine wave signal (V_{id}) is applied to the bases of each differential pair. Now, the differential output current of a bipolar differential pair can be written as

$$I_{od} = k_1 I_{ee} \tanh(k_2 V_{id})$$

where k_1 and k_2 are constants. (I'll buy Bob Harley dinner at CES if he can tell me what k_1 and k_2 are.) For small values of V_{id} the hyperbolic tangent operation can be dropped (remember from senior math in high school something called the Taylor series), and now we have

$$I_{od} = k_1 k_2 I_{ee} V_{id}$$

Just what we want—an analog multiplier. Rotel uses an NE5532AN as a differential-to-single-ended converter at the multiplier core's output. The 5532 also acts as the current-to-voltage converter for the analog multiplier. Rotel uses the fully balanced structure to take care of even-order distortion terms in the multiplier core, like the errors caused by the presence of the hyperbolic tangent function. (OK, I hear you. This is a lot harder to understand than "the midrange is woolly," but you've got to admit it is more interesting *[...get a life, Dave!—Ed.]*, and it could actually affect the sound quality!) Channel-separation adjustment pots make sure the correct amount of L+R gets added to the correct amount of L–R to reproduce the left channel (or subtracted, for the right channel). Separate pots are supplied for the wide and narrow modes. The modes are switched by relays, not cheap transistors! The blend circuit that follows uses a CMOS switch, but the output muting is also done with a relay. It all sounds so very good, but something must be wrong under the surface because no antibirdie filter should be required in this setup, but it is in the signal path.

To conclude the signal path, audiophile-design-style AD847 op-amps are the output buffers in the Rotel. Electrolytics are used for the dc blocks at the front and rear of the buffer. The antibirdie filter circuitry adds three more electrolytics into the composite FM signal path. A second-order lowpass filter with a 15 kHz passband is formed around the AD847. Normally a higher-order filter would be used, and more high-frequency energy was observed at the output of this tuner than usual.

High-end design practice continues in the power supply. Separate secondaries of the large transformer go to separate bridges for the analog and digital sections. The analog side uses 4700 µF capacitors on the unregulated rails. A two-transistor open-loop regulator is driven by a reference formed with a current source and a zener diode to generate the regulated ±14 V rails for the stereo decoder. The LA1235 gets a separate two-transistor discrete regulator that uses feedback. The other five regulators for the tuner are integrated units.

A very nice feature of this tuner is the inclusion of a separate IF strip (two ceramic filters and a discrete amplifier) plus an LA1235 chip for the tuning meter. This is done because the main IF filter strip is designed to limit even on weak signals. Once this happens, the strength of the signal cannot be determined. That's why most signal-strength meters pin even when the tuner is receiving a relatively week signal. With this specially designed low-gain IF strip made just for the tuning meter, the relative signal strength of strong signals can be determined.

Construction and parts quality of the Rotel RHT-10, both inside and out, are a step above the Japanese mass market, but they are not quite up to the level of the McIntosh MR7084 at the same price. For example, although a double-sided board is used, the top side is only a ground plane. The board has no through holes, and jumpers need to be used instead. Despite the build quality, some of the internal adjustments were way off. Channel separation was running 20 dB as the unit was delivered. The pots were noisy and had to be cleaned before they could be adjusted.

Although it is not designed as a super tuner—recall it has no supernarrow mode in the IF and that el cheapo quadrature detector causes the AM suppression to be no better than 60 dB—it sure performed like one in many ways. Real-world $2f_1 \pm f_2$ spurious signals were present only under the most challenging test conditions that Richard Modafferi could throw at it. Only the Accuphase beat it here, and performance was similar to that of the Onkyo T-9090II.

Selectivity was not as good as on tuners with a supernarrow IF strip but it comes very close. The Rotel can bring in things in narrow mode that require suppernarrow mode on the Onkyo. That means the signal will be cleaner on the Rotel, since the narrow filter distorts the phase less. But it must be noted that under worst-case conditions (40 dB differences in signal level between adjacent channels) the Onkyo could cleanly reproduce signals that were barely listenable on the Rotel.

THD at 1 kHz in stereo was –60 dB in wide mode and –44 dB in narrow mode. Better results in wide mode would have been possible with something a little better than the quadrature detector because the stereo decoder is designed to have very low distortion, as explained above. The Modafferi 10 kHz stereo IM test result was a superb –75 dB in the wide mode (no surprise, given the sophistication of the MPX stage). The IM result was –44 dB in the narrow mode. Channel separation after readjustment was ≥41 dB across the band in wide mode and ≥30 dB in narrow mode. Frequency response just made it inside the ±0.5 dB window given in the manufacturer's specification sheet.

At $1500 this is a good tuner. But at $750 it would be a runaway bargain for those of you who do not have worst-case signal conditions. But it is not $750—or is it? Rotel has a tuner called the RT-990BX that sells for just that price and is said to be almost identical to the RHT-10. We would like to tell you for sure, but Rotel of America appears to be no longer willing to lend equipment or even send a service manual to ***The Audio Critic***. It would seem that any criticism of the company's products, no matter how slight and how well documented, is unacceptable to them. This once again proves that great engineering and production people can work for companies that have marketing employees with small-minded agendas and petty resentments. So if you have the cash and decent signal conditions, try the Rotel RHT-10. If you do not have money coming out of your ears but want to take a chance, try the RT-990BX.

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AES Technical Paper by Rich and Aczel

On October 6, 1995, at the 99th Convention of the Audio Engineering Society in New York, David Rich presented to the Analog Electronics session his paper, coauthored by Peter Aczel, ***"Topological Analysis of Consumer Audio Electronics: Another Approach to Show that Modern Audio Electronics Are Acoustically Transparent."*** *The preprint number of the paper is* ***4053****. Copies are obtainable from the Audio Engineering Society, 60 East 42nd Street, Room 2520, New York, NY 10165-2520. Voice: (212) 661-8528. Fax: (212) 682-0477.*