

Bandwidth and sampling rate in digitizing oscilloscopes

Bandwidth and sampling rate are both used as figures of merit in DMOs. Here are the practicalities on what each means, and the limitations

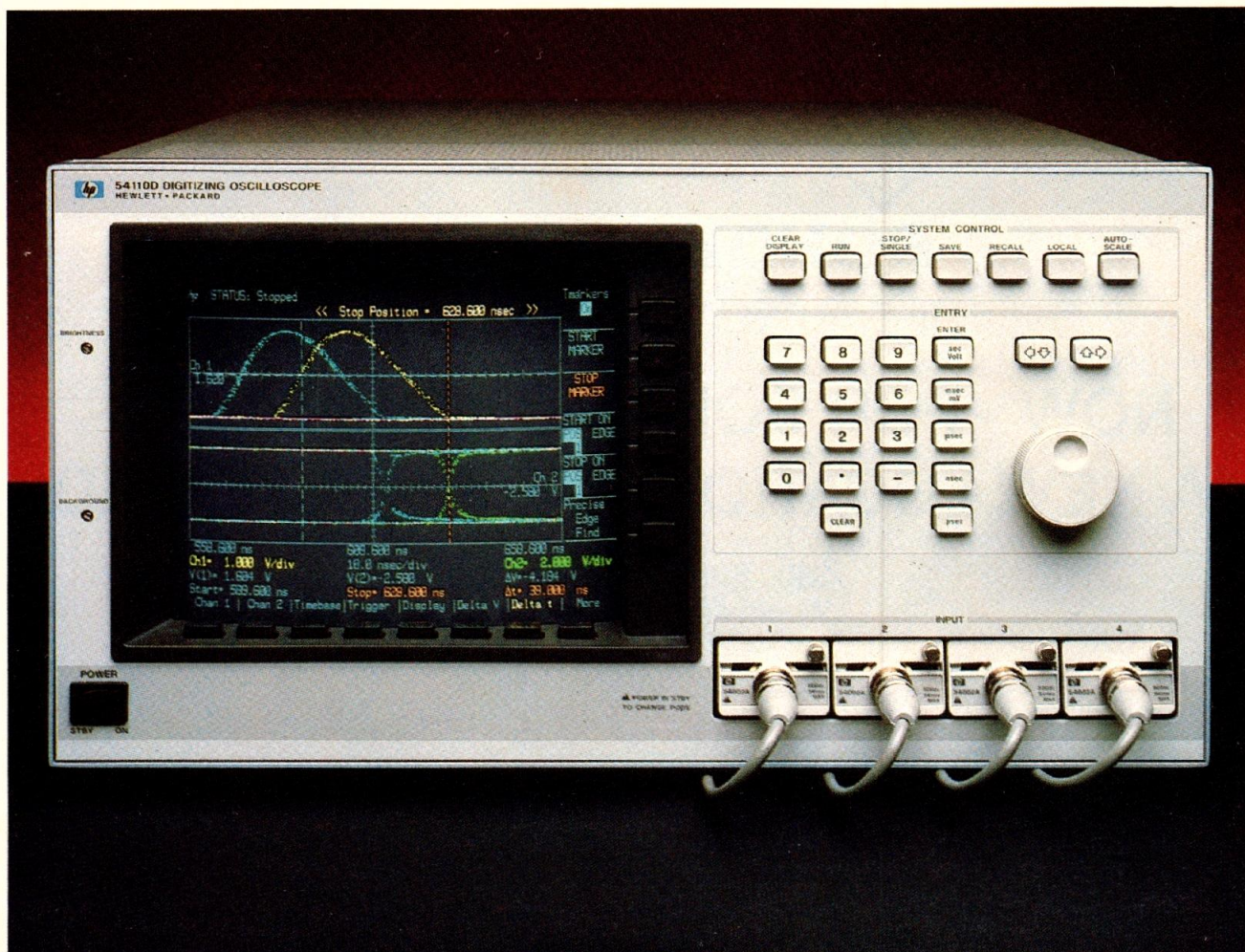
By Art Porter, Product Marketing Engineer, Hewlett-Packard Company

Bandwidth, along with its time-domain equivalent, rise time, has historically been the principal figure of merit for oscilloscopes. With the increasingly popularity of digitizing oscilloscopes, sampling rate, or digitizing rate, has attracted almost equal attention as a figure of merit. The relationship between bandwidth and sampling rate has given rise to a lot of confusion, so as a starting point, let's review why bandwidth and sampling rate are important. Oscilloscopes are not generally used to look at well-behaved, regular signals like sine waves, so who cares whether a sine wave is 3 dB down in amplitude at the specified frequency? What matters are the time and voltage relationships in irregular waveshapes, and how they are affected by bandwidth.

Most oscilloscope measurements are time-interval measurements. Consider a propagation delay measurement (*figure 1*). The question to be answered by this measurement is, "What is the time interval from the 50% point on the response transition?" Note that the time interval is defined in terms of a change in voltage. When the input voltage changes, the oscilloscope cannot exactly reproduce the input signal at every instant in time, due to the transient response of the oscilloscope.

In general, the greater the oscilloscope's bandwidth (conversely, the less its rise time), the smaller the errors in measuring time intervals. A good

Color in digital: the HP 54110D digitizing oscilloscope. See Product Profiles in this issue.



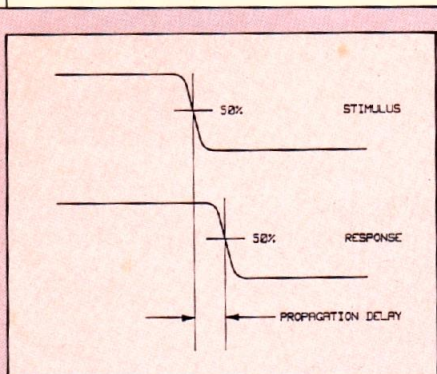


Figure 1

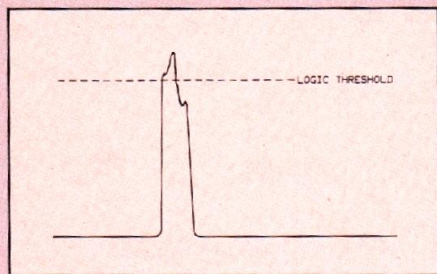


Figure 2

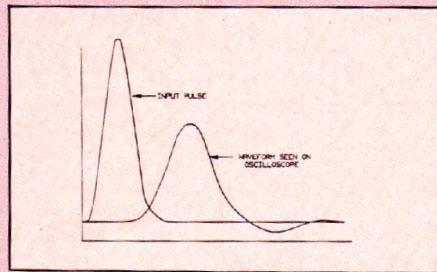


Figure 3

rule of thumb to use is that the oscilloscope's rise time should be less than one-third the time interval to be measured.

Bandwidth also affects amplitude measurements in the time domain. Consider measuring the height of a narrow glitch (figure 2). In a digital circuit, it may be important to know whether the glitch crosses the logic threshold. If the oscilloscope's bandwidth is insufficient, there may be a significant error in measuring the peak height of the glitch (figure 3). In general, the oscilloscope's rise time should be less than one-third the 50% width of the narrowest pulse to be measured.

Sampling

In a digitizing oscilloscope, the situa-

tion is further complicated because the waveform is quantized into discrete time and voltage samples. To add to the confusion, there are at least three methods of sampling the signal: real-time sampling, sequential sampling, and random repetitive sampling. Each of these sampling methods impacts measurements differently, so a brief description of each is in order.

In real-time sampling, the signal is digitized on the fly, in real time. There is a simple 1:1 correspondence between the samples and the times at which they were taken (figure 4). The advantage of real-time sampling is in single-shot measurements. All the data about the signal is acquired in one acquisition cycle.

In sequential sampling, only one sample of the signal is digitized on each occurrence of the trigger. With each successive trigger, the sampling point is delayed further from the trigger point. After many samples are acquired and digitized, the signal is reconstructed in the oscilloscope's digital memory. Sequential sampling requires that the signal be repetitive.

Random repetitive sampling is similar to sequential sampling, except that the signal is constantly sampled and digitized at a rate determined by the oscilloscope's sampling clock. To determine the time relationship of each sample to the trigger, the time between the sample clock and the trigger event is measured whenever a trigger event is detected. Again, as in sequential sampling, the signal data is reconstructed after many samples are acquired, so a repetitive signal is required.

Sampling rate affects measurements

Let's examine the simplest situation (i.e., real-time sampling) to determine the effect of sampling rate on measurements. We'll then compare that to the effect of bandwidth to illustrate the relationship.

Consider measuring a signal with a relatively fast transition (figure 5). When the samples are not spaced closely enough, we can't know the location of the edge very precisely, nor

can we ascertain its shape. The effect is the same as using an analog oscilloscope with insufficient bandwidth. If the oscilloscope has a higher sampling rate as in figure 6, the edge can be located correctly in time and its rise time can be measured.

Going back to the example of a narrow glitch (figure 7), if the sampling rate is insufficient, the glitch may be missed altogether, or its amplitude may be incorrectly represented. Again, the effect is the same as using an oscilloscope with insufficient bandwidth.

Nyquist Versus Bandwidth

Nyquist's theorem states that if a signal is sampled at a frequency $2F$, there is no information in the samples about the components of the signal at frequencies above F . The practical implication is that the effect on time-domain measurement uncertainty of a sample rate $2F$ is equivalent to a band-limit filter with a sharp cutoff at a frequency- F . Note: this should not be extrapolated to read, "The bandwidth of a real-time digitizing oscilloscope is half the sampling rate."

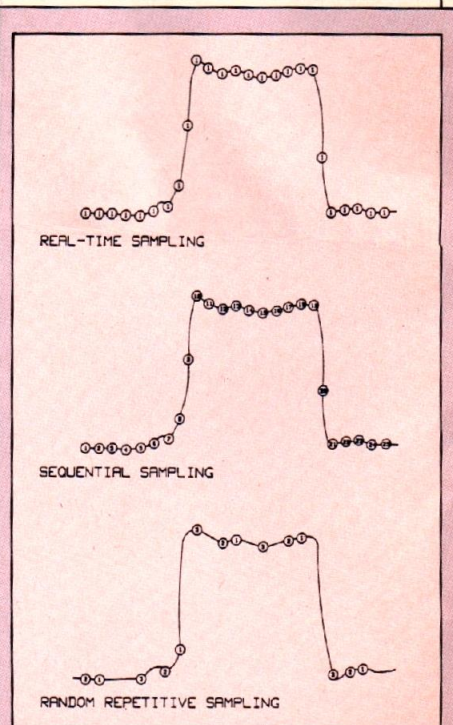


Figure 4

The Nyquist limit is an absolute upper limit; there can be no information above the Nyquist limit. Landau proved that, "data cannot be transmitted as samples at a rate higher than the Nyquist rate, regardless of the location of the sampling instants, the nature of the set of frequencies which the signals occupy, or the methods of construction." This situation is quite different from the typical Gaussian amplifier response.

Nyquist's theorem comes out of information theory. The error rate or signal-to-noise ratio of a communication channel as a function of information transfer rate is affected in similar ways by bit rate (for digital transmission) and by bandwidth (for analog transmission).

In oscilloscope measurements, the implication of Nyquist's theorem is that we can determine times and voltages on a signal using a real-time digitizing oscilloscope with approximately the same error as would result from using an analog oscilloscope with a bandwidth equal to half the sampling rate in the optimum case.

Thus, in selecting a real-time digitizing oscilloscope, the digitizing rate should be selected to be at least six divided by the smallest rise time or pulse width to be measured.

To illustrate why this is not quite the same as simply stating that the bandwidth is half the sample rate, consider a sine wave with a frequency F , where $2F$ is the sample rate (figure 8). In this situation, it is obvious from the figure that we can't even be sure there is any signal present, much less what its amplitude is.

Now consider the case where the sampling rate is four times the sine wave frequency, as in figure 9. In either of the situations illustrated in figure 9, although the signal data is somewhat skimpy, we can reconstruct the signal from the data. The algorithm used for reconstruction must include some assumptions about the shape of the signal, since the sampled data doesn't indicate whether it was a sine wave, a square wave, or a triangle wave, for instance (figure 10). Let's assume that we decide to reconstruct the signal as a sine wave. The effect on our uncer-

tainty about the shape of the original signal is exactly the same as band-limiting the signal. Had we input a square wave, a sine wave, or a triangle wave into an analog oscilloscope with a frequency equal to or greater than the oscilloscope's -3 dB bandwidth, we would see a sine wave on the screen. This is an intuitive illustration of the similarity between the effects of sampling rate and bandwidth.

If the sampling rate is ten times the signal frequency, as in figure 11, an excellent reconstruction is possible by merely connecting the samples with straight vectors. With this sample density, the eye can interpret the original signal visually with negligible error.

As the signal frequency approaches the Nyquist limit, the assumptions in the reconstruction algorithm regarding the signal shape must become more dominant. Figure 8 shows that at the Nyquist limit, the assumptions inherent in the reconstruction algorithm dominate the reconstructed waveshape entirely, since the sampled signal data leaves the waveshape entirely up to the imagination. The effect is the same as a filter with a very high equivalent Q .

The effect on the time domain and frequency domain response of the oscilloscope is the same as using extreme underdamping to extend the bandwidth of an analog oscilloscope. For this reason, few digitizing oscilloscopes or waveform analyzers offer reconstruction algorithms that work close to the Nyquist limit. For example, the HP 54200A/D has a 200 megasample/second digitizing rate. Its digital reconstruction filter is limited to a 50 MHz bandwidth, or one-fourth the sampling rate. This ensures minimum distortion in the reconstructed signal due to undersampling effects.

Some digitizing oscilloscopes have a bandwidth many times higher than their sampling rate. For example, the HP 54100A/D has a sampling rate of 40 MHz and a bandwidth of 1 GHz. This is achieved by using a wideband preamp and a narrow-aperture RF sampler ahead of the A/D converter. To view a high-frequency signal, many samples acquired at different times are overlaid in correct time relationship to one another (figure 4). The time rela-

tionship is preserved by the trigger circuit. The bandwidth for repetitive signals is limited only by the bandwidth of the preamplifier and the aperture width of the sampler.

Single shot limitation

The example of Figure 12 shows why digitizing oscilloscopes that are optimized for single-shot measurements, or that operate only in the real-time sampling mode like the HP 54200A/D, often incorporate analog bandwidth limiting and/or digital filter-

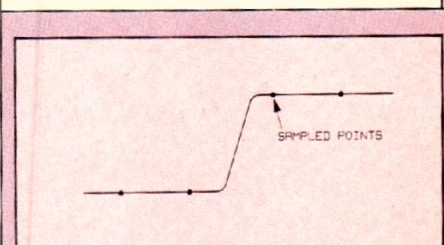


Figure 5

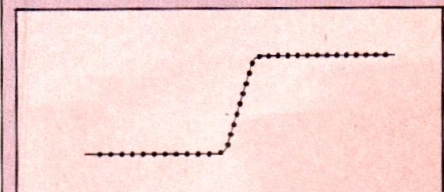


Figure 6

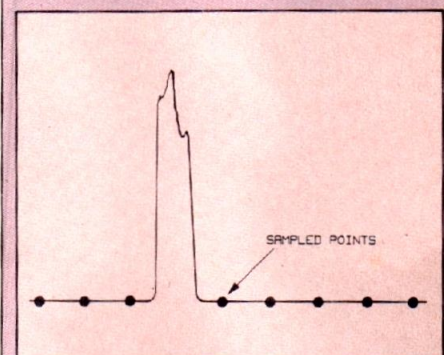


Figure 7

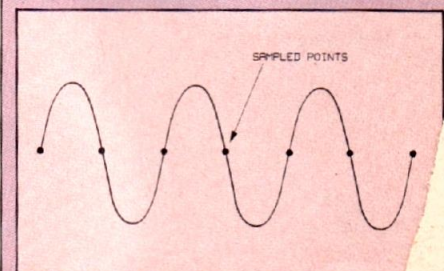


Figure 8

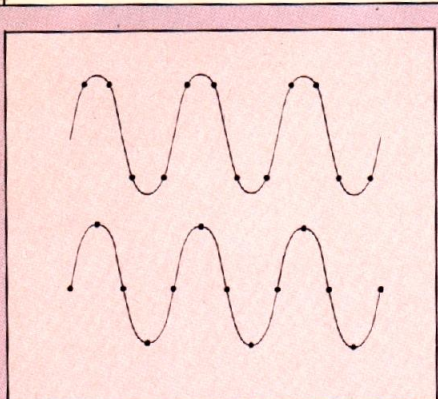


Figure 9

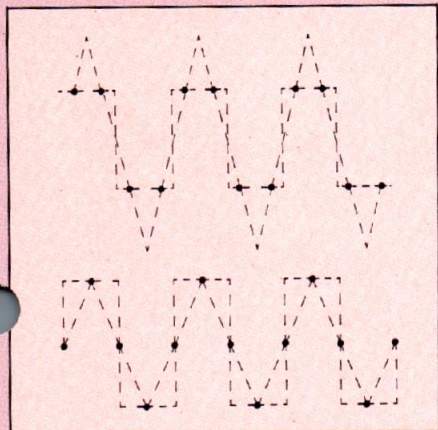


Figure 10

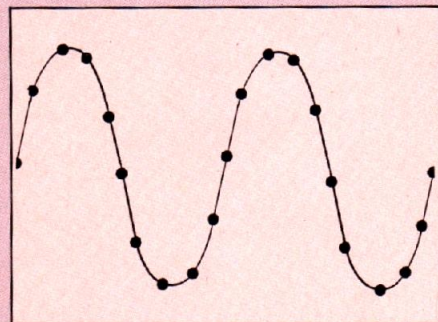


Figure 11

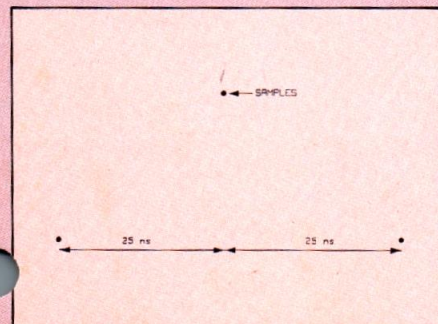


Figure 12

ing. If the incoming signal had been band-limited to remove frequencies above 20 MHz, or the data were digitally filtered, then the ambiguity in figure 12 would not be present; the value of the sample acquired on the pulse peak would be quite small.

Where digital filtering or analog band-limiting is not used, the oscilloscope user must use some judgment in interpreting the data in a single-shot measurement. The effective "bandwidth" of the measurement in terms of useful data about the signal has an absolute limit at the Nyquist rate. In the absence of digital filtering and reconstruction, a more conservative approach is to assume that any frequency components above one-tenth the sampling rate may be aliased, since you must depend on the eye for reconstructing the signal (refer to figures 8, 9, and 11). This would imply that you should be cautious about the results of single-shot measurements above 4 MHz using the HP 54100A/D, although there is some information present about the input signal up to 20 MHz.

Selecting the right scope

The first question to ask when selecting an oscilloscope, analog or digitizing, is "what is the measurement requirement?" To select the right digitizing oscilloscope for a given application, it is necessary to look at three items: the repetitive bandwidth, the single-shot bandwidth (if different), and the sampling rate. Each of these must be evaluated in light of its effect on time-interval or amplitude measurement error or uncertainty.

The bottom line

The bottom line is that we really don't care what the relationship is between sampling rate and bandwidth; what we care about is the effect each has on the measurement. Since there is not a simple, unambiguous formula to relate sampling rate to bandwidth, it is far better and easier to analyze the effect of sampling rate on the measurement directly rather than indirectly through relating sampling rate to bandwidth.

In addition to the familiar time and frequency domains, we can use the probability domain to advantage in analyzing digitizing oscilloscopes relative to an application requirement and comparing them to one another. **CEE**

Fig. 1. Propagation delay measurement

Fig. 2. Measuring the height of a glitch to determine whether it crosses the logic threshold

Fig. 3. A narrow pulse applied to a 'scope

Fig. 4. The number on each sampled point indicates the acquisition cycle on which that point was sampled.

Sequential sampling and random repetitive sampling require repetitive signals to reconstruct the digitized signal from samples acquired on successive acquisition cycles. Real-time sampling acquires all the samples on one cycle.

Fig. 5. The edge can't be located precisely because of signal undersampling

Fig. 6. With a higher sampling rate, the edge can be located with good resolution, and its characteristics may be determined

Fig. 7. A glitch may be missed altogether if the sample interval is wider than the glitch

Fig. 8. A sine wave with the frequency F , sampled at a rate of $2F$

Fig. 9. A sine wave with frequency F , sampled at a rate $4F$, shown in two sampling-phase possibilities. The samples contain sufficient data to recover the signal's correct amplitude and phase in either situation illustrated.

Fig. 10. Either set of samples in figure 9 might represent a square wave, a triangle wave, or any other arbitrary waveshape. The important point to note is that if we reconstruct a sine wave from the samples, we have as much information about the original signal as if we had passed the signal through a band-limiting filter that removed all frequencies but the fundamental. This is equivalent to viewing the signal with an oscilloscope with a bandwidth equal to F .

Fig. 11. A sine wave with frequency F sampled at a rate of $10F$; reconstruction can be accomplished by joining the dots

Fig. 12. Given only the samples acquired in a single acquisition, the signal information can be very limited

555 timer ~~is~~ waveforms in multiple scope display

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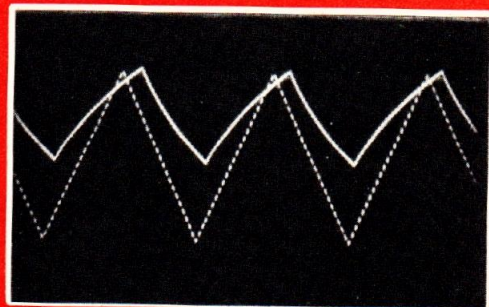
When two or more analog signals referenced to a common base line are simultaneously displayed on an oscilloscope, it is sometimes difficult to distinguish between the waveforms. By using a 555 integrated-circuit timer, dotted or dashed line markers can be added easily to one or more analog signals (Fig. 1).

By means of the circuit shown in Fig. 2, the 555 is connected as an astable multivibrator. The scope's trace position at any instant is determined by the sum of the voltages across R_4 and R_5 . When the square wave across R_4 is zero, the trace position is determined solely by the analog input signal. When the square wave across R_4 is not zero, it drives the trace off the screen. The fast rise and fall of the pulse cannot be seen at normal scope intensity.

The supply voltage V_{CC} , and, consequently, the output voltage of the 555, must be greater than the peak-to-peak voltage of the analog signal, which can range from 4.5 v to 16 v. Also, for proper operation, the square-wave frequency should be at least 5 to 10 times the analog frequency. The timer's frequency is easily

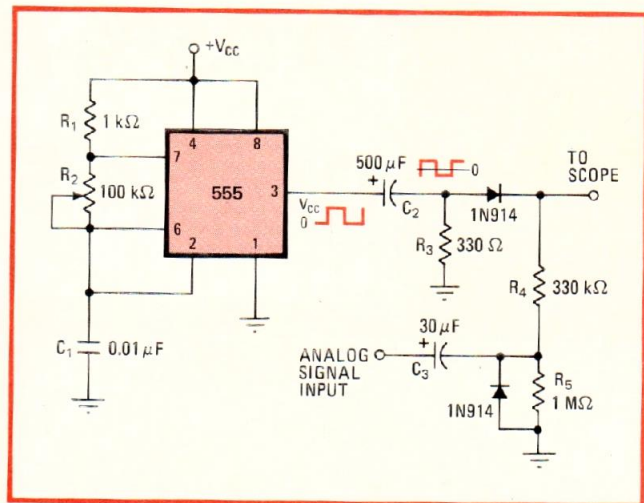


(a)



(b)

Tagging. When two or more waveforms are displayed on a single scope (a), identifying markers can be added to the analog input signals to display them as dotted or dashed traces (b).



2. Dots all, folks. A 555 integrated-circuit timer generates pulses that drive the scope trace off scale with fast rise times and fall times, to produce a dotted or dashed display of a waveform. The timer frequency, set by R_2 , determines the appearance of the trace.

adjusted by varying R_2 . The value of the frequency in hertz is $1.44/(R_1 + 2R_2)C_1$, where resistance and capacitance values are in ohms and farads, respectively. With the components shown, the circuit can handle without distortion input signals having a peak-to-peak voltage of 1 V. □

Circuit Ideas

Deflection amplifier

The amplifier is designed for use with an electrostatically deflected tube, and combines the frequency-response of a cascode amplifier with the linearity of a constant current-fed long-tailed pair. Adjust the

value of R to give 3mA through each load resistor. The output transistors need small heat sinks.

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