CATHODE FOLLOWERS

I read with interest Christopher Paul's article "Rehabilitating Cathode Followers" (May '06, p. 36), and I fear that it neglects the whole point of using the cathode follower.

The cathode follower gives you large amounts of current gain when you need them. I don't think it's fair to compare a common-cathode voltage amplifier with a cathode follower—both going into 100k loads—because that's not how the cathode follower is used. Try them both with a 50Ω load and then see which one does a better job.

The cathode follower should clearly be *better* at dealing with capacitive loads than a voltage amplifier, specifically because the output impedance is lower and this will swamp a larger load capacitance.

A fairer comparison would be between the cathode follower and a voltage amplifier combined with an output transformer to get the same output impedance. In this case, the follower *should* perform more poorly into a capacitive load. Though, of course, you now have all the other difficulties that come along with having a transformer in the signal path.

Scott Dorsey
Kludge Audio
Williamsburg, Va.

Christopher Paul responds:

Mr. Dorsey seems to have been on the side of the angels from the start when it comes to acknowledging the ability of cathode followers (CFs) to drive capacitive loads. I wonder whether he is aware of how many people hold the view that a CF is simply unsuitable for such a task. I wrote the article to address this commonly held view, not to compare the abilities of CFs and common-cathode amplifiers (CCAs) to drive 100kΩ resistive loads. I don't believe I made that comparison.

Mr. Dorsey intrigues with his suggestion of comparing the abilities of CFs and CCAs to drive a 50Ω load. Which does he say would do better, if either? And if we were to use a transformer with a CCA, would it not be only fair to consider one with a CF? Which would better drive a capacitive load, and why?

If there are simple answers to these questions, perhaps Mr. Dorsey could provide them in a letter. If not, I for one would be interested in an article from him on those topics.

It's about time someone (Christopher Paul) "acquitted" the cathode follower ("Rehabilitating Cathode Followers," May '06). I used them (6SN7) driving KT88s in my 125W per channel amp ("Mad Katy," June '06, p. 6) with no problem (slew distortion or any other) driving the fairly high input capacitance of 4 × KT88 in parallel push-pull. The bandwidth of the CF is ≈500kHz, clean.

However, I'd like to point out one significant difference between the CF and the cathode-coupled amplifier (CCA), besides its near-unity voltage gain: Set up a CF and a CCA, as you've done in your article, and have each driving a purely resistive load, say 100k as you've shown on p. 36. Apply a 20kHz sine wave input to each, with each circuit driven to just below visible distortion. Now connect a 47μF cap across each 100k load. The CF will have the distortion shown in your Fig. 9.

But the CCA output won't be distorted; it will be attenuated but clean. That's because (1) the CCA has much more gain, so its input drive level is much lower than that of the CF (for the same just-below-clipping output condition); and (2) if the CCA tube didn't cut off with the 100k load, it still wouldn't with the load cap added.

The CCA has a modest amount of negative feedback (NFB) from the 2k cathode resistor's degeneration, but this NFB hardly "cares" what the plate output voltage is doing. Therefore, the equivalent output structure is a current source (gmV_g) in parallel with the plate resistance, and the 20k plate bias R.

This will drive any linear impedance, even a short circuit, without load-caused distortion.

The CF, however, is quite different—it has nearly 100% NFB from (and sensitive to) the cathode signal. This will make the tube respond to low (capacitive or other) load impedance by trying to "crank up the current drive" in an effort to keep the cathode output voltage from falling.

So the CCA advantage is that once the maximum input signal is determined for undistorted output with a moderate resistive load, then no amount of capacitance will produce slew (or other) distortion. The "normal" low operating distortion will change somewhat, but no severe distortion will be induced.

This is good for driving unspecified cable lengths. The relatively high output impedance and poor HF bandwidth can be improved by selecting a high-current tube and/or using an L-pad at the output.

I used a solid-state equivalent, that is, an open-collector transistor output, to drive long microphone cables ("A Purist Recording Mixer," *aX*, Feb. '06, Fig. 6, p. 55).

But as Mr. Paul's excellent article shows, the cathode follower is perfectly able to drive capacitive (or other difficult) loads, if properly designed.

Dennis Colin
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Christopher Paul responds:

I'd like to thank Mr. Colin for his kind remarks about my article as well as for his thought-provoking (at least to me) comments on CF and CCA "Drivers on the Edge" (DOTE), which deliver the maximum currents they can to resistive loads. I agree with his analysis of how each will perform after adding capacitance. But I disagree with his conclusion that the advantage in such a case goes to the CCA.

Before I explain why, I'd like to point out that both of us are focusing on standalone drivers—that is, on those outside of feedback loops. As an aside, note that the frequency response deficit of a CCA DOTE would be converted into CF DOTE-like distortion by the extra gain of a feedback loop, which always attempts to extend frequency response.

Returning to stand-alone DOTEs, I think it's important to start any design with a statement of a required drive level for a specified range of loads, all concurrent with a flat frequency response. If loads vary too much and these requirements aren't met, then the design fails. The CF fails with occasional peak distortion, and the CCA with a signal level-independent frequency response deficit.

I don't think it's very useful to ask which design fails in the superior manner. The better question to ask is how to prevent failure in the first place. Mr. Colin's answer of a higher current triode is a necessary part of the solution for either DOTE.

It is instructive to return to the wide range of useful sub-DOTE level circuits and to apply to them some of the insights gained from Mr. Colin's letter. Remove the load capacitance from the DOTEs and reduce the drive levels so both circuits have a comfortable margin with their resistive loads. Now begin to add back load capacitance. The first problem you'll encounter is a CCA frequency response deficit.

Mr. Colin's letter shows how the CCA can overcome this, essentially by paralleling an additional resistance with the load and driving the CCA harder to achieve the original levels with the new load. That means, however, that the CCA must supply more current to its load in the mid and low bands, and therefore should be expected to produce more distortion there than the CF. Not a plus for the CCA.

Depending on signal levels, further capacitance increases would next cause one of two things. You could reach DOTE limits—a case already discussed—or you could reach a frequency response deficit for the CF. In the latter case, it is interesting to estimate the ratio of the capacitances which would cause frequency response problems for the drivers. This ratio is affected somewhat by the original load resistance, the one that the CF is still driving. That resistance is usually reasonably large, so make an approximation by ignoring it.

The CF output impedance is about $1/g_m$. That of the CCA is approximately $(1/g_m + R_k)G$, where G is the unloaded gain of the CCA and R_k is its cathode resistor. For small and medium signal triodes, R_k is typically around $2/g_m$, so the CCA output impedance would be about $3G/g_m$, or 3G times that of the CF. (You can make the 3 disappear by capacitively bypassing R_k.) In many cases, that's a pretty big difference in the values of load capacitance that each design can tolerate without exhibiting frequency response deficits.

Unlike the CCA, the CF automatically compensates for a wide range of load capacitance, delivering only the currents needed by the load. Accordingly, it is an excellent driver for Mr. Colin's cables of uncertain capacitance. On the other hand, compensating the CCA for bandwidth requires of it higher dynamic currents to produce the original signal levels. This probably results in higher distortions than with the CF, a point I had missed in my article. And finally, consider that CCA DOTEs within performance-enhancing feedback loops take on CF-like distortion and frequency response characteristics.

Even more so than before, I think the advantage in driving complex loads goes to the CF.

AMP ABUSE

I, too, enjoyed reading Jim Carlyle's article in February 2006 *aX* about using clamped bias in a Class AB2 vacuum tube amplifier (p. 6). In a May 2006 Xpress Mail letter (p. 70), Bob Cordell asks the right (or *a* right) question: "Has Jim solved a problem that really only exists on the test bench unless the amplifier is being abused?" but neglects to define "abuse." To the musical purist, abuse might constitute anything producing audible distortion (whatever *that* may be), while to musical instrument and pro-sound designers "abuse" might rather be anything exceeding the power, voltage, or thermal envelopes of the device under test.

Similarly, an acoustic and jazz guitarist might have different ideas of "abuse" than a thrash-metal devotee. The latter might at some times actually desire crossover distortion, while at other times want "max headroom, loud and clean" (from the same amp, at the kick of a foot switch, of course!). But whatever the possibly questionable merits of crossover distortion in the realm of musical instrument tone generation, I think it safe to say that thus far the only artist to have successfully incorporated the almost (but not quite) totally unrelated phenomenon of hangup (or blocking) distortion is Neil Young.

My understanding is that hangup distortion occurs when a capacitively coupled tube output stage *with global feedback around the driver* (such as Mr. Carlyle describes) is pushed so hard that the output tube saturates well before the driver. At this point the feedback disconnects and the driver tube output suddenly peaks as well, as any further increment in its own input signal is no longer attenuated by feedback. And the driver input *is* incremented. . . The power tube grid is then suddenly pulled *far*, far above its cathode with a concomitant rush of negative charge to the grid side of the coupling cap.

Now when the grid swings negative on the next half cycle you get not just crossover distortion, but full cutoff of the output tube (class B)—which is where the output tube stays (sometimes for as long as a second) until the grid bias resistor can eventually bleed off the excess charge. It is only with the greatest artistic skill that this effect may be coaxed into anything resembling music.

Blocking distortion has traditionally been controlled in MI amplifier design though judicious choice of dynamic range of the driver, coupled with small-value output tube bias resistors (in other words, sound hi-fi design). The former avoids or limits AB2 operation in the first place, while the second allows rapid bias re-equalization and avoids blocking. Fender's "master-volume" series silver face Pro and Twin Reverb amplifiers typify this conservative driver approach, as well as the general apathy it enjoys among most blues and rock guitarists.

Note that by making bleed resistor R variable, Mr. Carlyle's "Clamped Bias" becomes quite versatile, at least from the viewpoint of a guitar amp designer. My own suspicion is Mr. Carlyle's invention will indeed prove to be some sort of breakthrough, although not necessarily in traditional hi-fi design. Give it some time, kick it around. There are many inventive tinkerers Out There, and I look forward to experimenting with "Clamped Bias" in my own modifications and designs.

Thanks, Jim!

Reference

"The Ultimate Tone, Vols. 1 – 5" Kevin O'Connor, Power Press Publishing, London, Canada. (It's *all* in there somewhere...)

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TUBE AMP DESIGN

Richard Sears' article ("6AS7 Current-Balanced Push-Pull Amplifier," May '06, p. 42) is interesting in the way that it systematically investigates the use of constant-current source in the design of tube amplifiers. It is clear that this author is fascinated by these devices and the numerous advantages they may offer.

Cathode-coupled phase splitters, or "Schmitt phase splitters," have equal output impedances but unbalanced output gains. The strategy to balance outputs is to strongly increase the common-cathode resistor. But doing so with a resistor requires some mounting tricks to bias tubes properly.

An elegant solution is to use a constant-current source, which exhibits low DC and high AC resistances. Constant-current sources are easy to set up using solid-state devices rather than tubes, and I am glad to notice that mixing tubes and solid-state devices in the design was not a philosophical blocking for Richard Sears.

Unfortunately, "golden-eared" audiophiles do not like the resulting sound. I have often heard this objection as soon as active devices are involved, meaning that their transient behavior is not as clean as supposed and generate probably unwanted harmonic and/or intermodulation distortion. Is author Sears aware of this judgment, and what is his opinion about it?

Concerning the output push-pull power section, it is clear that a strict balance of anode DC currents is essential to avoid OPT core saturation. This requirement exists for all push-pull topologies and all OPT types. Many solutions have been proposed to solve it correctly, the most popular being the grid bias adjustment to make anode current equal. Richard Sears' solution, which consists of implementing in constant-current source in the cathodes, is a clever one and has been suggested many times by designers. It has the advantage of making the output stage design simpler, the DC balance more robust, and the power supply modulation protection more efficient.

Nevertheless, I am still afraid of the "golden ears" judgment, and from that point of view, listening sessions with comments made by audiophiles would have been greatly appreciated. Why not consider the output push-pull power stage as a Schmitt phase splitter itself delivering output power? This seems possible and would lead to a great simplification for the driving section.

To conclude, Richard Sears' work deserves compliments and will encourage amateurs to follow and set up their own concepts and ideas, for the benefit of the whole audio community. I am waiting, with great interest, for the next iteration of this work with DC coupling between the driving and power sections, as promised by the author.

Pierre Touzelet
Vélizy, France

Richard Sears responds:

I would like to thank Mr. Touzelet for taking the time to write and for the complimentary tone of his letter.

The issue Pierre raises regarding audiophile acceptance of the use of solid-state devices in tube amplifiers is a tricky one. However, the use of solid-state constant-current sources is expanding and in some circles is fairly well accepted. Pierre states that the "golden ears" reaction to the use of hybrid audio circuits is negative: he asks whether I am aware of this issue and what my opinion is. I am aware of this issue and my view is that audiophile opinion should never stand in the way of experimentation.

Pierre further expressed a desire to know more about the results of listening sessions with audiophiles. In my article, I referred to some development done with the help of John Dahlman, who is an audiophile and professional bassist and also a member of the Washington, D.C., DIY Audio Group. I also exposed this design to another member of the D.C. group, who listened using his Phy KM30 drivers on Phy specification sand-filled open-baffles.

In comparison with a 45 SE amp (with 6SL7 drive and James iron), my push-pull design was clearly superior. It was much more open, exhibiting extraordinary microdynamic resolution, a very stable image, and authentic tonal balance. John has told me that this amp is *the* one he would choose to use when recording for mastering reference playback. In comparison to my own 50, 300B, and 845 SE amplifiers, it just does a better job of getting out of the way of the music and that is surely the point.

I designed this amplifier as a "thought experiment" to see what could be done using inexpensive power toroids as output transformers. Pierre states that "this author is clearly fascinated by these devices" (solid-state constant-current sources). Actually, *my primary interest is in reducing the dependence of the sonic performance of an audio circuit on the power supply,* and minimizing the number of capacitors in the signal path, especially those in the B+ signal bypass loop, which are more expensive to source in the large value required in comparison to coupling capacitors *at the same level of quality.* Constant-current sources are a very powerful tool in this regard. This is exemplified by the fact that I was able to eliminate the B+ bypass capacitor across the balanced circuits.

Furthermore, the parts cost for this design was, at the time of building, around $250. This is extremely low and was a part of my objective. I can state that the performance of the resulting design is a delightful confirmation of the efficacy of my thought experiment. I do intend to build the next iteration, a fully DC-coupled version; however, I am not sure whether the DC stability of such a circuit will be acceptable.

Pierre posits the concept of making the output stage perform both phase-splitting and output transfer duties. This notion did occur to me. The reason I did not pursue is that if the output stage is driven by a single-ended signal, then the drive voltage to the grounded grid side will be via the cathode coupling. This means that the voltage across the MOSFETs will be heavily modulated, resulting in any nonlinearity of the MOSFETs under heavy modulation affecting the sound.

With balanced drive, the cathode signal will be due to any balance correction and thus vanishingly small. This is also true of the drive stage, which is driven by a single-ended signal; however, the voltage modulation of the drive stage current source is reduced by the gain of the drive stage. It would be interesting to see what sonic changes are audible if the drive stage were driven using a single-ended to balanced transformer. As I have stated, my design is the result of a thought experiment and I encourage further experimentation by readers.

HORN SUB

I've been thinking about the horn subwoofer in the "Table Tuba" (June '06, p. 20), and I think the basic folded horn idea has a lot of promise. The driver mounting baffle could perhaps be isolated easily enough from the rest of the assembly with something like wrestling mat gaskets to suppress the mechanical "thud" impulse when a bass drum is kicked, and so on. Then the signal to the stereo or four, five, or seven "surround sound" speakers could be delayed by some kind of electronic circuitry with a fine-tuning delay adjustment for a simultaneous arrival of this low-frequency sound wave. A delay of about one millisecond per foot of horn length should be close.

Mounting the speaker baffle reminds me of the technical challenge that submarine designers probably face (shock mounting parts and acoustic isolation for stealth to avoid detection and location).

Nelson Revis
Springfield, Mo.

PS. As you may have suspected, I'm a believer in the sealed enclosure type of speakers (more realistic percussion sound reproduction).

Bill Fitzmaurice responds:

Regarding the baffle, the way the Table Tuba is braced, the baffle is inert, as is the entire box, so isolation is not required. This bracing scheme is the same as is employed in my Tuba 36 pro-sound subs, which operate with as much as 1kW input with no discernible vibration.

As for delay, you can use it if you wish, but it's really not required. Most of the pros who use Tubas have systems with delay capabilities, and none of them have found using delay to be audible when the subs are crossed over at 100Hz or lower. The same psychoacoustic and physiologic properties that render low frequencies non-directional also render time adjustment moot, though that situation is radically different above 200Hz or so.

As soon as my latest issue of *audioXpress* arrived, I eagerly read Bill Fitzmaurice's latest contribution: The Table Tuba. Being a horn enthusiast, I always am intrigued by his projects, but I am a bit disappointed by the lack of theoretical background. I think most subscribers would appreciate a few paragraphs about how Bill arrived at this design (and therefore how they might modify it for their own purposes—no guarantees, of course!).

Here is my little bit of "reverse engineering," but I certainly would appreciate the real story from Bill.

The design is obviously a series of conical horns, one feeding into the other. The early part of the horn is a very close approximation of a pure exponential horn with a flare rate of 30Hz (up to about 46″ or a width of 2.56). At that point, the Table Tuba's rate of expansion slows dramatically compared to a true exponential design (which would have a width of 28.66 at 131″).

If you look at exponential/hyperbolic flares (still at 30Hz), you can approximate the Table Tuba by decreasing M as you progress out the horn: by 70″ the M ~ .75, by 90″ M ~ .5, and by 107″ M ~ .25, but at the end M > .25 (**Table 1**).

This certainly makes a more compact horn, but what is the theoretical basis for changing M as you progress down the horn? Of course, something totally different may be happening here, and again, Bill's comments would be most welcome.

TABLE 1: TABLE TUBA DISTANCE WIDTH

M = 1 M = .75 M = .5 M = .25
0 0.75
0.75
15.75 1.14
1.13
20.47 1.3
1.25
35.43 2.01
2.06
46.06 2.68
2.56
62.99 4.29 3.46
3.62
70.87 5.35 4.21 3.31
4
89.76 9.06 6.97 5.39 3.94 4.88
107.48 14.84 11.54 8.62 6.14 5.88
131.1 28.66 22.13 16.42 11.54 14

Mark Parker
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Bill Fitzmaurice responds:
There is no M factor in the Table Tuba, because it is neither an exponential nor a hyperbolic horn. The taper is a constant conic through the bulk of the horn, save the flare of the last section.

SUB DRIVER

Having just read Jerry Stump's article ("A Snail Tail Subwoofer," May '06, p. 18), I noticed that he uses a Peerless DVC8 part # 831858. Because I cannot find a match for this driver anywhere, maybe he can suggest where to obtain this driver or another driver that gives about the same performance in his enclosure. He notes that the amplifier is available from Madisound, and his published F_s for this driver = 22, which must be pretty low for an 8″ driver.

Alan Smith
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Jerry Stump responds:
I am pretty sure the 831858 Peerless driver that I used was a sale item at Madisound probably several years ago. It is alleged to have been in what is now the CC line and has a very good reputation as a high performance but inexpensive sub. It does have a very low F_s and the low Q_{ts} is with the voice coils in parallel. I think it is about .39 for a single 8Ω voice coil.

Here are the parameters for the Peerless driver: F_s 23.4Hz, V_{as} 2.5 ft^3, Q_{ts} .23, 89.9 dB/W, P_e 200W, 1.4mH with the voice coils hooked in parallel. The design in the article has an F_3 of about 45Hz. I used Harris Technologies' BassBox design software.

I wrote the article to demonstrate a concept for putting long port ducts in small enclosures, and the rest of my project was kind of incidental. But here are some suggestions.

Go to the Parts Express website. You will find DIY Resources on the lower left-hand corner of the home page. Click on Speaker Wizard. Follow the steps, and include your intended enclosure internal dimensions. Several products from their catalog will pop up, including port sizes and F_3.

Here are the four drivers that came up on my initial search. I did a quick and dirty job with BassBox to get the approximate rectangular port sizes. The original enclosure dimensions I used for these calculations was about 1.1 ft^3 unless noted.

	Cost	F_s	Q_{ts}	V_{as} (ft^3)	F_3	Port (Rect)
Dayton ST210	49.50	48	.4	1.34	48	1.5 × 8.5 × 16
Dayton DC200	18.65	34	.38	2.04	34	.75 × 8.5 × 14
Pioneer B20GR30-51F	34.50	47	.26	2.44	47	1 × 8.5 × 15.5
HiVi D8G	91.45	40	.53	1.10	40	.5 × 8.5 × 19

You will find *lots* of 8s that will work in enclosures in this size range. The Dayton Quatro is a car sub, the Silver Flute is very low Q, and the Tang Band is an interesting elliptical unit. The Silver Flute seems closest to the Peerless driver.

	Cost	F_s	Q_{ts}	V_{as} (ft^3)	F_3	Port
(2) Dayton 7″ Aluminum	18.15 ea.	36	.42	.75	32	1 × 8.5 × 14 *1.5 ft^3.
Silver Flute W20RC38-04 (Madisound)	31.20	31	.23	2.5	40	1 × 8.5 × 27
Dayton Quatro 8″	49.75	36	.4	.82	31	.75 × 8.5 × 16
Tang Band W69-1042	41.57	35	.35	1.34	38	1 × 8.5 × 17.5

I have used the Dayton 7″ aluminum cone units, and a pair would work well mounted in the Snail Tail enclosure, one on each side, if you bump the internal volume up to about 1.5 ft^3 net. F_3 for that combo is about 32Hz.

Within its acoustic output limits, I have been very happy with my sub and the little PSB Alpha Intro LRs. *aX*

What's New on the aX website?

Web-exclusive content:

- Excel program and Gerber file from George Danavaras' "Equalize The Bass Response Of Your Room" (*aX* 8/06).
- A clearer version of the PCB Stuffing Guide (Fig. 5) from Jan Didden's "Greening of the Behringer Ultra-Drive Pro DCX2496" (*aX* 8/06)
- Classics–History of Audio
  "Feedback Amplifiers" by H.S. Black–from *Bell Laboratories Record*, June 1934
  "Loudness and Pitch" by Harvey Fletcher–from *Bell Laboratories Record*, January 1935

Articles from Past issues:

- "Renovating the Scott 222C, Part 2: Modifications," By Charles Hansen
- "Renovating the Scott 222C, Part 3," By Charles Hansen
- "Renovating the Scott 222C, Part 4: Listening & Measurements," By Charles Hansen

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