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See PulseAudio for the main article. Getting debug output from pulseaudio It can be useful to stop the daemon and start it in a terminal during debugging: $ systemctl --user stop pulseaudio.socket $ systemctl --user stop pulseaudio.service $ pulseaudio -v Add v's to increase verbosity. Volume Here you will find some hints on volume issues and why you may not hear anything. Auto-Mute Mode Auto-Mute Mode is a configurable setting from amixer. For more information, see ALSA#Disabling auto mute on startup. Muted audio device If one experiences no audio output via any means while using ALSA, attempt to unmute the sound card. To do this, launch alsamixer and make sure each column has a green 00 under it (this can be toggled by pressing m): $ alsamixer -c 0 Note: alsamixer will not tell you which output device is set as the default. One possible cause of no sound after install is that PulseAudio detects the wrong output device as a default. Install pavucontrol and check if there is any output on the pavucontrol panel when playing a .wav file. In setups with multiple outputs (e.g. 'Headphone' and 'Speaker') using plain amixer to toggle Master can trigger PulseAudio to mute the active output too, but it does not necessarily unmute it when Master is toggled back to be unmuted. [1] To resolve this, amixer must have the device flag set to 'pulse': $ amixer -D pulse sset Master toggle This will cause amixer to ask PulseAudio to do the toggling rather than toggling it directly. Because of this, PulseAudio will correctly unmute Master as well as any applicable output. Muted application If a specific application is muted or low while all else seems to be in order, it may be due to individual sink-input settings. With the offending application playing audio, run: $ pacmd list-sink-inputs Find and make note of the index of the corresponding sink input. The properties: application.name and application.process.binary, among others, should help here. Ensure sane settings are present, specifically those of muted and volume. If the sink is muted, it can be unmuted by: $ pacmd set-sink-input-mute false If the volume needs adjusting, it can be set to 100% by: $ pacmd set-sink-input-volume 0x10000 Note: If pacmd reports 0 sink input(s), double-check that the application is playing audio. If it is still absent, verify that other applications show up as sink inputs. Check /usr/share/alsa-card-profile/mixer/paths/analog-output.conf.common. If the volume does not appear to increment/decrement properly using alsamixer or amixer, it may be due to PulseAudio having a larger number of increments (65537 to be exact). Try using larger values when changing volume (e.g. amixer set Master 655+). Per-application volumes change when the Master volume is adjusted This is because PulseAudio uses flat volumes by default, instead of relative volumes, relative to an absolute master volume. If this is found to be inconvenient, asinine, or otherwise undesireable, relative volumes can be enabled by disabling flat volumes in the PulseAudio daemon's configuration file: /etc/pulse/daemon.conf or ~/.config/pulse/daemon.conf flat-volumes = no and then restarting PulseAudio by executing $ pulseaudio -k $ pulseaudio --start Volume gets louder every time a new application is started Per default, it seems as if changing the volume in an application sets the global system volume to that level instead of only affecting the respective application. Applications setting their volume on startup will therefore cause the system volume to "jump". Fix this by disabling flat volumes, as demonstrated in the previous section. When Pulse comes back after a few seconds, applications will not alter the global system volume anymore but have their own volume level again. Note: A previously installed and removed pulseaudio-equalizer may leave behind remnants of the setup in ~/.config/pulse/default.pa or ~/.pulse/default.pa which can also cause maximized volume trouble. Comment that out as needed. Add the following: /etc/pulseaudio/default.pa load-module module-alsa-sink sink_name=delta_out device=hw:M2496 format=s24le channels=10 channel_map=left,right,aux0,aux1,aux2,aux3,aux4,aux5,aux6,aux7 load-module module-alsa-source source_name=delta_in device=hw:M2496 format=s24le channels=12 channel_map=left,right,aux0,aux1,aux2,aux3,aux4,aux5,aux6,aux7,aux8,aux9 set-default-sink delta_out set-default-source delta_in No sound below a volume cutoff or Clipping on a particular output device Known issue (will not fix): source/pulseaudio/+bug/223133 If sound does not play when PulseAudio's volume is set below a certain level, or if you hear clipping on output even at low volume (including bluetooth devices), try setting ignore_dB=1 in /etc/pulse/default.pa: /etc/pulse/default.pa load-module module-udev-detect ignore_dB=1 However, be aware that it may cause another bug preventing PulseAudio to unmute speakers when headphones or other audio devices are unplugged. Low volume for internal microphone If you experience low volume on internal notebook microphone, try setting: /etc/pulse/default.pa set-source-volume 1 300000 Clients alter master output volume (a.k.a. volume jumps to 100% after running application) If changing the volume in specific applications or simply running an application changes the master output volume this is likely due to flat volumes mode of pulseaudio. Before disabling it, KDE users should try lowering their system notifications volume in System Settings -> Application and System Notifications -> Manage Notifications under the Player Settings tab to something reasonable. Changing the Event Sounds volume in KMix or another volume mixer application will not help here. This should make the flat-volumes mode work out as intended, if it does not work, some other application is likely requesting 100% volume when its playing something. If all else fails, you can try to disable flat-volumes: /etc/pulse/daemon.conf flat-volumes = no Then restart PulseAudio daemon: # pulseaudio -k # pulseaudio --start No sound after resume from suspend If audio generally works, but stops after resume from suspend, try "reloading" PulseAudio by executing: $ /usr/bin/pasuspender /bin/true This is better than completely killing and restarting it (pulseaudio -k followed by pulseaudio --start), because it does not break already running applications. If the above fixes your problem, you may wish to automate it, by creating a systemd service file. 1. Create the template service file in /etc/systemd/system/resume-fix-pulseaudio@.service: [Unit] Description=Fix PulseAudio after resume from suspend After=suspend.target [Service] User=%I Type=oneshot Environment="XDG_RUNTIME_DIR=/run/user/%U" ExecStart=/usr/bin/pasuspender /bin/true [Install] WantedBy=suspend.target 2. Enable it for your user account # systemctl enable resume-fix-pulseaudio@YOUR_USERNAME_HERE.service 3. Reload systemd # systemctl --system daemon-reload ALSA channels mute when headphones are plugged/unplugged improperly If when you unplug your headphones or plug them in the audio remains muted in alsamixer on the wrong channel due to it being set to 0%, you may be able to fix it by opening /etc/pulse/default.pa and commenting out the line: load-module module-switch-on-port-available Then restart the pulseaudio backend: $ systemctl --user restart pulseaudio Volume resets to 50% every few seconds Install alsa-tools and use: $ hdajackretask Set "Not Connected" to everything but the ports you are using. It seems the other unused audio ports on the motherboard interfere with the used ones. Then if you want use the Boot Override to save this change between reboots. There is a possibility it is the Front Green Headphone that is causing the bug, if you need it override the Front Microphone to Headphone and the Front Green Headphone to "Not Connected" and use the Front Microphone port as your headphone port. More info about this problem: [2]. Volume low/too quiet on analog headphones/speakers If you added the ignore_dB=1 option earlier to the load-module module-udev-detect line in your /etc/pulse/default.pa, try removing it. Delay when changing volume using media keys Edit /etc/pulse/daemon.conf and set enable-deferred-volume = no. This might cause some sound crackles when changing volume, in that case you might want to leave that option enabled and tweak the deferred-volume-safety-margin-usec and deferred-volume-extra-delay-usec options instead. Microphone Microphone not detected by PulseAudio Note: For bluetooth headsets you may be seeking this thread instead. Determine the card and device number of your mic: $ arecord -l **** List of CAPTURE Hardware Devices **** card 0: PCH [HDA Intel PCH], device 0: ALC269VC Analog [ALC269VC Analog] Subdevices: 1/1 Subdevice #0: subdevice #0 In hw:CARD,DEVICE notation, you would specify the above device as hw:0,0. Then, edit /etc/pulse/default.pa and insert a load-module line specifying your device as follows: load-module module-alsa-source device=hw:0,0 # the line above should be somewhere before the line below .ifexists module-udev-detect.so Finally, restart pulseaudio to apply the new settings: $ pulseaudio -k ; pulseaudio -D If everything worked correctly, you should now see your mic show up when running pavucontrol (under the Input Devices tab). PulseAudio uses wrong microphone If PulseAudio uses the wrong microphone, and changing the Input Device with Pavucontrol did not help, take a look at alsamixer. It seems that Pavucontrol does not always set the input source correctly. $ alsamixer Press F6 and choose your sound card, e.g. HDA Intel. Now press F5 to display all items. Try to find the item: Input Source. With the up/down arrow keys you are able to change the input source. Now try if the correct microphone is used for recording. No microphone on ThinkPad T400/T500/T420 Run: $ alsamixer -c 0 Unmute and maximize the volume of the "Internal Mic". Once you see the device with: $ arecord -l you might still need to adjust the settings. The microphone and the audio jack are duplexed. Set the configuration of the internal audio in pavucontrol to Analog Stereo Duplex. No microphone input on Acer Aspire One and Lenovo Ideapad 310-15ISK/330-15ARR Install pavucontrol, unlink the microphone channels and turn down the left one to 0. Some applications (e.g. chromium) can change microphone levels causing the same issue, a workaround is to remap stereo input to mono and use the remapped device as default. Static noise in microphone recording If we are getting static noise in Skype, gnome-sound-recorder, arecord, etc.'s recordings, then the sound card sample rate is incorrect. That is why there is static noise in Linux microphone recordings. To fix this, we need to set the sampling rate in /etc/pulse/daemon.conf for the sound hardware. In addition to the guide below, since PulseAudio 11 it is possible to set avoid-resampling = yes in daemon.conf. Determine sound cards in the system (1/5) This requires alsa-utils and related packages to be installed: $ arecord --list-devices **** List of CAPTURE Hardware Devices **** card 0: Intel [HDA Intel], device 0: ALC888 Analog [ALC888 Analog] Subdevices: 1/1 Subdevice #0: subdevice #0 card 0: Intel [HDA Intel], device 2: ALC888 Analog [ALC888 Analog] Subdevices: 1/1 Subdevice #0: subdevice #0 The sound card is hw:x,y where x is the card number and y is the device number. In the above example, it is hw:0,0. Determine sampling rate of the sound card (2/5) We aim to find the highest sample rate supported by the hw:0,0 sound card using a trial-and-error procedure starting from a low value. When the top value is reached, we got a warning message: arecord -f dat -r 60000 -D hw:0,0 -d 5 test.wav "Recording WAVE 'test.wav' : Signed 16 bit Little Endian, Rate 60000 Hz, Stereo Warning: rate is not accurate (requested = 60000Hz, got = 44100Hz) please, try the plug plugin observe, the got = 44100Hz. This is the maximum sampling rate of our card. Setting the sound card's sampling rate into PulseAudio configuration (3/5) The default sampling rate in PulseAudio: $ grep "default-sample-rate" /etc/pulse/daemon.conf ; default-sample-rate = 48000 48000 is disabled and needs to be changed to 44100: # sed 's/; default-sample-rate = 48000/default-sample-rate = 44100/g' -i /etc/pulse/daemon.conf Restart PulseAudio to apply the new settings (4/5) $ pulseaudio -k $ pulseaudio --start Finally check by recording and playing it back (5/5) Let us record some voice using a microphone for, say, 10 seconds. Make sure the microphone is not muted and all $ arecord -f cd -d 10 test-mic.wav After 10 seconds, let us play the recording... $ aplay test-mic.wav Now hopefully, there is no static noise in microphone recording anymore. Another Possible Cause Another possible cause is that your mic has two channels but only one channel can provide a valid sound signal. Some information can be found here. The solution is to remap the stereo input to a mono input: 1. Find your source name from the following command; mine is alsa_input.pci-0000_00_1f.3.analog-stereo pacmd list-sources | grep 'name:.*input' 2. Edit /etc/pulse/default.pa and add the following lines, where INPUT_NAME is name of the input source from above step: load-module module-remap-source source_name=record_mono master=INPUT_NAME master_channel_map=front-left channel_map=mono set-default-source record_mono 3. Restart PulseAudio: $ pulseaudio -k $ pulseaudio --start Now arecord hopefully works. You may still need to change the RecordStream from setting to Remapped Built-in Audio Analog Stereo of a specific application in the Recording tab of pavucontrol. If using a USB microphone Try plugging it into a different port (eg: ports at the back rather than front). No microphone on Steam or Skype with enable-remixing = no When you set enable-remixing = no on /etc/pulse/daemon.conf you may find that your microphone has stopped working on certain applications like Skype or Steam. This happens because these applications capture the microphone as mono only and because remixing is disabled, Pulseaudio will no longer remix your stereo microphone to mono. To fix this you need to tell Pulseaudio to do this for you: 1. Find the name of the source # pacmd list-sources Example output edited for brevity, the name you need is in bold: index: 2 name: driver: flags: HARDWARE HW_MUTE_CTRL HW_VOLUME_CTRL DECIBEL_VOLUME LATENCY DYNAMIC_LATENCY 2. Add a remap rule to /etc/pulse/default.pa, use the name you found with the previous command, here we will use alsa_input.pci-0000_00_14.2.analog-stereo as an example: /etc/pulse/default.pa ### Remap microphone to mono load-module module-remap-source master=alsa_input.pci-0000_00_14.2.analog-stereo master_channel_map=front-left,front-right channels=2 channel_map=mono,mono 3. Restart Pulseaudio # pulseaudio -k Note: Pulseaudio may fail to start if you do not exit a program that was using the microphone (e.g. if you tested on Steam before modifying the file), in which case you should exit the application and manually start Pulseaudio # pulseaudio --start If your microphone volume creeps up automatically and causes the sound to be distorted, you can fix it by disabling mic boost: In all files /usr/share/alsa-card-profile/mixer/paths/analog-input*.conf, Under [Element Capture] set volume to zero Under [Element Internal Mic Boost] set volume to zero. Under [Element Int Mic Boost] set volume to zero. Under [Element Mic Boost] set volume to zero. including all variations such as [Element Headphone Mic Boost] and [Element Mic Boost (+20dB)] Then restart PulseAudio: # pulseaudio -k Microphone crackling with Realtek ALC892 Sometimes ALC892 chips have crackling sound while recording using a microphone. Some Pulseaudio config changes may help: /etc/pulse/daemon.conf resample-method = src-sinc-best-quality default-sample-format = s16le default-sample-rate = 48000 and add the use_ucm option to /etc/pulse/default.pa load-module module-udev-detect use_ucm=0 tsched=0 then restart pulseaudio. Microphone crackling with Azalia chipsets Some Azalia based chips have popping/crackling noise and distortion while recording using a microphone with PulseAudio. This can be fixed by loading the snd-hda-intel module with position_fix set to an appropriate value. This tells the module to use various DMA pointer fixes. Use trial and error to determine which value works for you. (source[dead link 2021-05-17 ⓘ]) Create a new modprobe.d config: /etc/modprobe.d/azalia-microphone.conf options snd-hda-intel position_fix=1 Valid values for position_fix are: 0 = Auto 1 = None 2 = POSBUF 3 = FIFO size then reload your modules. Echo test If you are unsure about your microphone setup, you can hear the input from the microphone in real-time by enabling the loopback module (source): $ pactl load-module module-loopback The module will show up in the Recording tab of the pavucontrol program, where the source and volume can be configured. While latency should be low, it should be sufficient to get a feeling of the sound quality as you will hear yourself speak in the microphone. To make the change permanent, add the following pulseaudio configuration: /etc/pulse/default.pa load-module module-loopback Watch out for feedback! Be ready to lower all volumes in case the microphone picks up the output from the loudspeakers. Naturally, it is better to run such a test with headphones. Audio quality Glitches, skips or crackling The newer implementation of the PulseAudio sound server uses timer-based audio scheduling instead of the traditional, interrupt-driven approach. Timer-based scheduling may expose issues in some ALSA drivers. On the other hand, other drivers might be glitchy without it on, so check to see what works on your system. To turn timer-based scheduling off add tsched=0 in /etc/pulse/default.pa: /etc/pulse/default.pa load-module module-udev-detect tsched=0 Then restart the PulseAudio server: $ pulseaudio -k $ pulseaudio --start Do the reverse to enable timer-based scheduling, if not already enabled by default. If you are using Intel's IOMMU and experience glitches and/or skips, add intel_iommu=igfx_off to your kernel command line. Some Intel audio cards using the snd-hda-intel module need the otions vid=8086 pid=8ca0 snoop=0. In order to set them permanently, create/modify the following file including the line below. /etc/modprobe.d/sound.conf options snd-hda-intel vid=8086 pid=8ca0 snoop=0 Please report any such cards to PulseAudio Broken Sound Driver page Static noise when using headphones Time-based scheduling may be causing this, disable it as explained in #Glitches, skips or crackling. Another reason you are encountering static noise in your headphone jack could be ALSA's loopback mixing. Make sure you have alsa-utils installed, launch alsamixer, then select your audio device (pressing F6), navigate all the way left using the left arrow, and stop on Loopback, if Enabled disable it using the down arrow. This should not impact audio playback or microphone recording negatively, unless you require loopback mixing. Some notebook models, like Dell XPS 13 9360, suffer from continuous hissing sound when a headphone is plugged in. Yet another reason for this symptom could be the power-saving mode of your audio device.[3] If you followed Power management#Audio, revert the changes and check if it solves the problem. Setting the default fragment number and buffer size in PulseAudio Disabling timer-based scheduling (0/4) By default, PulseAudio uses timer-based scheduling. In this mode, fixed-size fragments are not used at all, and so the default-fragments and default-fragment-size-msec parameters are ignored. To turn timer-based scheduling off add tsched=0 in /etc/pulse/default.pa: /etc/pulse/default.pa load-module module-udev-detect tsched=0 Decide on audio device parameters (1/4) Instructions below will cause PulseAudio to use a fixed size and number for audio fragments. These settings directly affect latency and power consumption. The latency is determined as default-fragments * default-fragment-size-msec, and the interrupt rate (i.e. how often the application is notified that some sound has indeed been played) is 1000 / default-fragment-size-msec. The considerations are: The total number of fragments must be within the limits accepted by the hardware. Most sound cards are OK with two or more fragments, but some require three or more. Giving the sound card more fragments than strictly necessary increases the latency, does not change power consumption, and does not remove the load from the scheduler. Therefore, it is advised only in the cases when the interrupts are not reliably delivered to the CPU, and one extra fragment beyond the minimum required should always be enough. Giving the sound card bigger fragments increases latency and decreases power consumption. Some applications (games and VoIP) request low latency and produce glitches when PulseAudio cannot fulfill their request. Setting the latency too low will put a lot of stress on the scheduler, also resulting in glitches. If one does not care about excessive power consumption, then 2 or 3 fragments, 5 ms each, are a reasonable choice. Modify PulseAudio's configuration file (2/4) /etc/pulse/daemon.conf default-fragments = 3 default-fragment-size-msec = 5 Restart the PulseAudio daemon (3/4) $ systemctl --user restart pulseaudio.service See if it helps (4/4) Run your applications, listen to the sounds they produce, inspect the journal. If the skips/pops/glitches are occasional and mostly correlated to the system being highly loaded: this is a scheduler problem, the latency needs to be increased. If there is metallic sound with the wrong speed from all applications: the most common reason is that you are trying to configure the fragment size which is way too small, like 1 ms. Do not do this. If some, but not all, applications produce a constant stream of glitches: this is an application that assumes a low-latency setup. So the fragment size must be decreased so that the application request becomes valid. Choppy sound with analog surround sound setup The low-frequency effects (LFE) or Subwoofer channel is not remixed per default. To enable it the following needs to be set in /etc/pulse/daemon.conf : /etc/pulse/daemon.conf remixing-produce-lfe = yes remixing-consume-lfe = yes You should also consider to set a proper crossover frequency for the LFE- channel. The crossover frequency is the frequency up to which the audio signal is rerouted to the LFE sink. The optimal crossover frequency in Hz depends on the size of all your speakers. /etc/pulse/daemon.conf lfe-crossover-freq = Laggy sound This issue is due to incorrect buffer sizes. First verify that the variables default-fragments and default-fragment-size-msec are not being set to non default values in the file /etc/pulse/daemon.conf. If the issue is still present, try setting them to the following values: /etc/pulse/daemon.conf default-fragments = 5 default-fragment-size-msec = 2 Choppy/distorted sound This can result from an incorrectly set sample rate. Try the following setting: /etc/pulse/daemon.conf avoid-resampling = yes #(Needs PA11 or higher) default-sample-rate = 48000 and restart the PulseAudio server. If one experiences choppy sound in applications using OpenAL, change the sample rate in /etc/openal/alsoft.conf: /etc/openal/alsoft.conf frequency = 48000 Setting the PCM volume above 0 dB can cause clipping. Running alsamixer will allow you to see if this is the problem and if so fix it. Note that ALSA may not correctly export the dB information to PulseAudio. Try the following: /etc/pulse/default.pa load-module module-udev-detect ignore_dB=1 and restart the PulseAudio server. See also #No sound below a volume cutoff or Clipping on a particular output device. Sound stuttering when streaming over network When streaming over Wi-Fi using module-native-protocol-tcp you can experience periodic sound stuttering with buffer underruns. As a workaround you can try to use rtp protocol. On sender side create rtp sink: /etc/pulse/default.pa load-module module-null-sink sink_name=rtp load-module module-rtp-send source=rtp.monitor and switch to it: /etc/pulse/default.pa set-default-sink rtp On receiver side load rtp module: /etc/pulse/default.pa load-module module-rtp-recv Pops when starting and stopping playback PulseAudio can suspend sinks after a period of inactivity. This can make an audible noise (like a crack/pop/scratch). Sometimes even when move the slider volume, or open and close windows (KDE4). This behavior is enabled in default configuration files: /etc/pulse/default.pa load-module module-suspend-on-idle /etc/pulse/system.pa load-module module-suspend-on-idle Commenting that line in relevant file fixes that issue. A better solution is to add the following file: ~/.config/pulse/default.pa .include /etc/pulse/default.pa .nofail unload-module module-suspend-on-idle .fail For some devices (eg. Bose Quietcomfort 35 II) setting high enough volume of the device (usually via physical buttons or a slider) eliminates the audible noise after stopping playback. Hardware and Cards No HDMI sound output after some time with the monitor turned off The monitor is connected via HDMI/DisplayPort, and the audio jack is plugged in the headphone jack of the monitor, but PulseAudio insists that it is unplugged: pactl list sinks ... hdmi-output-0: HDMI / DisplayPort (priority: 5900, not available) ... This leads to no sound coming from HDMI output. A workaround for this is to switch to another VT and back again. If that does not work, try: turn off your monitor, switch to another VT, turn on your monitor, and switch back. This problem has been reported by ATI/Nvidia/Intel users. Another workaround could be to disable the switch-on-port-available module by commenting it in /etc/pulse/default.pa [4]: /etc/pulse/default.pa ... ### Should be after module-*-restore but before module-*-detect #load-module module-switch-on-port-available ... No HDMI sound using a headless server You might want to use HDMI audio with your a/v receiver but no display. HDMI requires a video signal, which we have from the virtual terminal. By default, this signal is turned off after 600 seconds, thus the audio sink gets lost as well. To prevent screen blanking, add consoleblank=0 to the kernel command line. No cards If PulseAudio starts, run pacmd list. If no cards are reported, make sure that the ALSA devices are not in use: $ fuser -v /dev/snd/* $ fuser -v /dev/dsp Make sure any applications using the pcm or dsp files are shut down before restarting PulseAudio. Starting an application interrupts other app's sound If you have trouble with some applications (such as TeamSpeak or Mumble) interrupting sound output of already running applications (such as Deadbeaf), you can solve this by commenting out the line load-module module-role-cork in /etc/pulse/default.pa like shown below: /etc/pulse/default.pa ### Cork music/video streams when a phone stream is active # load-module module-role-cork Then restart pulseaudio by using your normal user account with $ pulseaudio -k $ pulseaudio --start The only device shown is "dummy output" or newly connected cards are not detected If the only playback device is the Dummy Output, PulseAudio cannot access your sound devices. It is possible there is an issue with logind giving permissions, see General troubleshooting#Session permissions for more information. An application might also not have been configured to work with PulseAudio. This happens with FluidSynth for example. To see which application is responsible for a direct access to the sound card via alsa, run the following command: # fuser -v /dev/snd/* Try to close these applications. pulseaudio, if running, should take again precedence over these applications and all the applications relying on pulseaudio should work again like expected. No HDMI 5/7.1 Selection for Device If you are unable to select 5/7.1 channel output for a working HDMI device, then turning off "stream device reading" in /etc/pulse/default.pa might help. See #Fallback device is not respected. Failed to create sink input: sink is suspended If you do not have any output sound and receive dozens of error messages related to a suspended sink from running journalctl -b as root, then backup first and then delete your user-specific pulse folders: $ rm -r ~/.pulse ~/.pulse-cookie ~/.config/pulse Simultaneous output to multiple sound cards / devices Simultaneous output to two different devices can be very useful. For example, being able to send audio to your A/V receiver via your graphics card's HDMI output, while also sending the same audio through the analogue output of your motherboard's built-in audio. This is much less hassle than it used to be (in this example, we are using GNOME desktop). Using paprefs, simply select "Add virtual output device for simultaneous output on all local sound cards" from under the "Simultaneous Output" tab. Then, under GNOME's "sound settings", select the simultaneous output you have just created. If this does not work, try adding the following to ~/.asoundrc: pcm.dsp { type plug slave.pcm "dmix" } Tip: Simultaneous output can also be achieved manually using alsamixer. Disable "auto mute" item, then unmute other output sources you want to hear and increase their volume. This can be useful for users who have multiple sound sources and want to play them on different sinks/outputs. An example use-case for this would be if you play music and also voice chat and want to output music to speakers (in this case Digital S/PDIF) and voice to headphones. (Analog) This is sometimes auto detected by PulseAudio but not always. If you know that your sound card can output to both Analog and S/PDIF at the same time and PulseAudio does not have this option in its profiles in pavucontrol, or veromix then you probably need to create a configuration file for your sound card. More in detail you need to create a profile-set for your specific sound card. This is done in two steps mostly. Create udev rule to make PulseAudio choose your PulseAudio configuration file specific to the sound card. Create the actual configuration. Create a pulseaudio udev rule. Note: This is only an example for Asus Xonar Essence STX. Read udev to find out the correct values. Note: Your configuration file should have lower number than the original PulseAudio rule to take effect. /usr/lib/udev/rules.d/90-pulseaudio-Xonar-STX.rules ACTION=="change", SUBSYSTEM=="sound", KERNEL=="card*", \ ATTRS{subsystem_vendor}=="0x1043", ATTRS{subsystem_device}=="0x835c", ENV{PULSE_PROFILE_SET}="asus-xonar-essence-stx.conf" Now, create a configuration file. If you bother, you can start from scratch and make it saucy. However you can also use the default configuration file, rename it, and then add your profile there that you know works. Less pretty but also faster. To enable multiple sinks for Asus Xonar Essence STX you need only to add this in. Note: asus-xonar-essence-stx.conf also includes all code/mappings from default.conf. /usr/share/alsa-card-profile/mixer/profile-sets/asus-xonar-essence-stx.conf [Profile analog-stereo+iec958-stereo] description = Analog Stereo Duplex + Digital Stereo Output input-mappings = analog-stereo output-mappings = analog-stereo iec958-stereo skip-probe = yes This will auto-profile your Asus Xonar Essence STX with default profiles and add your own profile so you can have multiple sinks. You need to create another profile in the configuration file if you want to have the same functionality with AC3 Digital 5.1 output. See PulseAudio article about profiles Some profiles like SPDIF are not enabled by default on the card Some profiles like IEC-958 (i.e. S/PDIF) may not be enabled by default on the selected sink. Each time the system starts up, the card profile is disabled and the pulseaudio daemon cannot select it. You have to add the profile selection to you default.pa file. Verify the card and profile name with : $ pacmd list-cards Then edit the config to add the profile ~/.config/pulse/default.pa ## Replace with your card name and the profile you want to activate set-card-profile alsa_card.pci-0000_00_1b.0 output:iec958-stereo+input:analog-stereo Pulse audio will add this profile the pool of available profiles Only S/PDIF output available This might happen if PulseAudio use the wrong output device. Firstly, set up proper card profile: $ pacmd set-card-profile alsa_card.pci-0000_00_1f.3 output:analog-stereo or $ pacmd set-card-profile alsa_card.pci-0000_00_1f.3 output:analog-stereo+input:analog-stereo Replace alsa_card.pci-0000_00_1f.3 with your card, and output:analog-stereo or output:analog-stereo+input:analog-stereo with your profile, remember to choose the profile with analog. Using shell auto completion could help you a lot. One could also use check available cards and profiles with: $ pacmd list-cards One might also need to set sink port by: $ pacmd set-sink-port alsa_output.pci-0000_00_1f.3.analog-stereo analog-output-headphones Check available sink ports with: $ pacmd list-sinks To keep these setting, add them to PulseAudio's configuration file default.pa. ~/.config/pulse/default.pa .include /etc/pulse/default.pa set-card-profile alsa_card.pci-0000_00_1f.3 output:analog-stereo+input:analog-stereo set-sink-port alsa_output.pci-0000_00_1f.3.analog-stereo analog-output-headphones Crackling or popping USB sound issues Although not related to PulseAudio, it is also possible that the card is not powered properly by it's USB port, or that it does not provide enough bandwidth. Try connecting the USB DAC directly to your computer's USB ports, avoiding any hubs or docks. Bluetooth See Bluetooth headset#Troubleshooting. Applications Audacity To work with PulseAudio, Audacity requires the pulseaudio-alsa is installed. This provides the pulse:... playback and recording devices. When starting Audacity you may find that your headphones no longer work. This can be because Audacity is trying to use them as a recording device. To fix this, open Audacity, then set its recording device to pulse:Internal Mic:0. Under some circumstances, playback may be distorted, very fast, or freeze, as discussed in the Audacity Wiki's Linux Issues page. The solution proposed in this page may work: start Audacity with: $ env PULSE_LATENCY_MSEC=30 audacity If the solution above does not fix this issue, one may wish to temporarily disable pulseaudio while running Audacity by using the pasuspender command: $ pasuspender -- audacity Then, be sure to select the appropriate ALSA input and output devices in Audacity. See also #Setting the default fragment number and buffer size in PulseAudio. OpenAL Some games may prevent you from switching the output device. Trying to move the sink with pactl gives the following error: pactl move-sink-input 11 alsa_output.pci-0000_01_00.1.hdmi-stereo-extra1 Failure: Invalid argument OpenAL needs to be configured to allow moving the sink: ~/.alsoftrc [pulse] allow-moves = true Other Issues Cannot update configuration of sound device in pavucontrol pavucontrol is a handy GUI utility for configuring PulseAudio. Under its 'Configuration' tab, you can select different profiles for each of your sound devices e.g. analogue stereo, digital output (IEC958), HDMI 5.1 Surround etc. However, you may run into an instance where selecting a different profile for a card results in the pulse daemon crashing and auto restarting without the new selection "sticking". If this occurs, use the other useful GUI tool, paprefs, to check under the "Simultaneous Output" tab for a virtual simultaneous device. If this setting is active (checked), it will prevent you changing any card's profile in pavucontrol. Uncheck this setting, then adjust your profile in pavucontrol prior to re-enabling simultaneous output in paprefs. Failed to create sink input: sink is suspended If you do not have any output sound and receive dozens of error messages related to a suspended sink from running journalctl -b as root, then backup first and then delete your user-specific pulse folders: $ rm -r ~/.pulse ~/.pulse-cookie ~/.config/pulse Pulse overwrites ALSA settings PulseAudio usually overwrites the ALSA settings—for example set with alsamixer—at start-up, even when the ALSA daemon is loaded. Since there seems to be no other way to restrict this behaviour, a workaround is to restore the ALSA settings again after PulseAudio has started. Add the following command to .xinitrc or .bash_profile or any other autostart file: restore_alsa() { while [-z "$(pidof pulseaudio)"]; do sleep 0.5 done alsactl -f /var/lib/alsa/asound.state restore } restore_alsa & Daemon startup failed Try resetting PulseAudio: $ rm -rf /tmp/pulse* ~/.pulse* ~/.config/pulse $ pulseaudio -k $ pulseaudio --start Check that options for sinks are set up correctly. If you configured in default.pa to load and use the OSS modules then check with lsof that /dev/dsp device is not used by another application. Set a preferred working resample method. Use pulseaudio --dump-resample-methods to see a list with all available resample methods you can use. To get details about currently appeared unfixed errors or just get status of daemon use commands like pax11publish -d and pulseaudio -v where v option can be used multiple time to set verbosity of log output equal to the --log-level[=LEVEL] option where LEVEL is from 0 to 4. See the #Outputs by PulseAudio error status check utilities section. See also man pages for pax11publish(1) and pulseaudio(1) for more details. Outputs by PulseAudio error status check utilities If the pax11publish -d shows error like: N: [pulseaudio] main.c: User-configured server at "user", refusing to start/autospawn. then run pax11publish -r command then could be also good to logout and login again. If the pulseaudio -vvv command shows error like: E: [pulseaudio] module-udev-detect.c: You apparently ran out of inotify watches, probably because Tracker/Beagle took them all away. I wished people would do their homework first and fix inotify before using it for watching whole directory trees which is something the current inotify is certainly not useful for. Please make sure to drop the Tracker/Beagle guys a line complaining about their broken use of inotify. This can be resolved temporary by: $ echo 100000 > /proc/sys/fs/inotify/max_user_watches For permanent use save settings in the 99-sysctl.conf file: /etc/sysctl.d/99-sysctl.conf # Increase inotify max watchs per user fs.inotify.max_user_watches = 100000 Warning: It may cause much bigger consumption of memory by kernel. See also Daemon already running On some systems, PulseAudio may be started multiple times. journalctl will report: [pulseaudio] pid.c: Daemon already running. Make sure to use only one method of autostarting applications. pulseaudio includes these files: /etc/X11/xinit/xinitrc.d/pulseaudio /etc/xdg/autostart/pulseaudio.desktop /etc/xdg/autostart/pulseaudio-kde.desktop Also check user autostart files and directories, such as xinitrc, ~/.config/autostart/ etc. Subwoofer stops working after end of every song Known issue: source/pulseaudio/+bug/494099 To fix this, enable-lfe-remixing = yes must be set as described in #Choppy sound with analog surround sound setup. Unable to select surround configuration other than "Surround 4.0" If you are unable to set 5.1 surround output in pavucontrol because it only shows "Analog Surround 4.0 Output", open the ALSA mixer and change the output configuration there to 6 channels. Then restart pulseaudio, and pavucontrol will list many more options. Realtime scheduling If rtkit does not work, you can manually set up your system to run PulseAudio with real-time scheduling, which can help performance. To do this, add the following lines to /etc/security/limits.conf: @pulse-rt - rtprio 9 @pulse-rt - nice -11 Afterwards, you need to add your user to the pulse-rt group: # gpasswd -a pulse-rt pactl "invalid option" error with negative percentage arguments pactl commands that take negative percentage arguments will fail with an 'invalid option' error. Use the standard shell '--' pseudo argument to disable argument parsing before the negative argument. e.g. pactl set-sink-volume 1 -- -5%. Fallback device is not respected PulseAudio does not have a true default device. Instead it uses a "fallback", which only applies to new sound streams. This means previously run applications are not affected by the newly set fallback device. gnome-control-center, mate-media and paswitchAUR handle this gracefully. Alternatively: 1. Move the old streams in pavucontrol manually to the new sound card. 2. Stop Pulse, erase the "stream-volumes" in ~/.config/pulse and/or ~/.pulse and restart Pulse. This also resets application volumes. 3. Disable stream device reading. This may be not wanted when using different soundcards with different applications. /etc/pulse/default.pa load-module module-stream-restore restore_device=false RTP/UDP packet flood In some cases the default configuration might flood the network with UDP packets.[5] To fix this problem, launch paprefs and disable "Multicast/RTP Sender".[6] New audio source streams auto-select to "blank" stream instead of BT headphones Example scenario: Restarting, stopping, fast forwarding a Youtube video on Firefox 68.0.1 on KDE + Arch might set the "sink" associated with it in PulseAudio to a blank state, and then output sound over laptop speakers. Inconsistently reproducible on Dell 9360. Fix seems to be to kill and restart pulseaudio. pulseaudio -k pulseaudio --start

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