adds character and texture to that percussive part. Let's apply it.

7 In the Selection-based Processing window, click the Apply button.

After a moment, the effect is "printed" to the selection, and the waveform is redrawn to represent the processed audio signal.

8 Close the Selection-based Processing window.

9 Click the background of the workspace to clear the selected section.

10 Press Z to zoom out.

11 Unsolo the Acoustic Gt track.

12 Turn off Cycle mode.

13 Listen to the CH Break Down section going into the CH Final section.

In the CH Break Down section, you hear the processed percussive guitar on the right side of the stereo field. At the beginning of the section, it sounds like an answer to the vocalists singing the song title, "Big." In the CH Final section, the fuzz effect stops and the acoustic guitar sounds clean again.

You can now apply plug-ins to selected portions of a track, which is a new tool you can use to add interest to parts of your songs. Feel free to be creative with that technique! Try using selection-based processing, for example, to add delay to only the last word of a sung phrase, or reverb to the last snare of the song.

Using an EQ Plug-in

The sound of an instrument consists of several frequencies mixed together in varying amounts. By applying an EQ plug-in to attenuate or boost certain ranges of frequency, you can alter the timbre of the sound, much as you would change the sound of your music player by tweaking the bass or treble EQ settings.

EQ plug-ins can shape the sound of your instruments, focusing them in a specific frequency range and helping each instrument cut through the mix without boosting the overall level of its channel strip. Equalizing (EQ'ing) an instrument can also decrease unwanted frequencies in its recording and keep it from masking another instrument in the same frequency range.

To shape the frequency spectrum of a vocal track, you will use the Channel EQ plug-in to attenuate some of its low rumbling while boosting some of the high frequencies to give it some air.

1 In the Marker track, drag the V2 marker up to the ruler, and start playback.
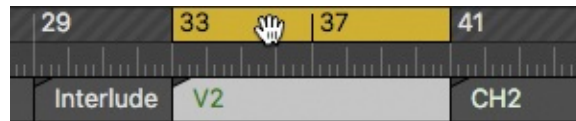
Feel free to keep repeating the V2 section throughout the whole exercise, or to stop and restart playback as you choose.

Several layered vocal tracks are located at the bottom of the workspace. You will solo one vocal track that hasn't yet been processed and EQ it.

2 Select the Ami Vox track (track 29) so that you can see its channel strip in the inspector, and then solo it (or press S).

3 At the top of the Ami Vox channel strip, click the EQ display.

In the Audio FX area, a Channel EQ plug-in is inserted in the first available slot, and the Channel EQ window opens.

The Channel EQ plug-in allows you to adjust eight bands of EQ. You can toggle a band on and off by clicking the button at the top of that band. By default, the first and last bands are turned off, and all the other bands are turned on. Each band's settings are shown below the graphic display in the EQ band fields. All the bands that are turned on by default have their Gain

parameters set to 0.0 dB, and in the graphic display, the EQ curve is flat, which means that the Channel EQ is not currently affecting the audio signal on the channel strip.

The Analyzer button toggles the frequency analyzer, which displays the frequency spectrum curve of the sound on the graphic display when the track is playing.

Note

> By default, the frequency analyzer displays the frequency spectrum of the sound at the output of the Channel EQ plug-in. You can click the Post EQ button inside the Analyzer button to switch the frequency analyzer to Pre EQ. It will then display the frequency spectrum of the sound at the input of the Channel EQ, before being processed by the Channel EQ plug-in.

**4** Make sure the Analyzer button is on, and listen to the soloed track.

A curve appears in the graphic display, showing the sound's frequency spectrum in real time. Listen closely to the vocals as you watch the occasional movement in the very low range of frequencies (to the left). Feel free to raise your monitoring volume on your audio interface, or use the Volume fader on the Ami Vox channel strip. You can hear some low-frequency noises in the recording, especially at the beginning of bar 35. In vocal recordings, those low-frequency noises are often generated by the

singer accidentally touching the mic stand, or by the mic cable rubbing against the mic stand.
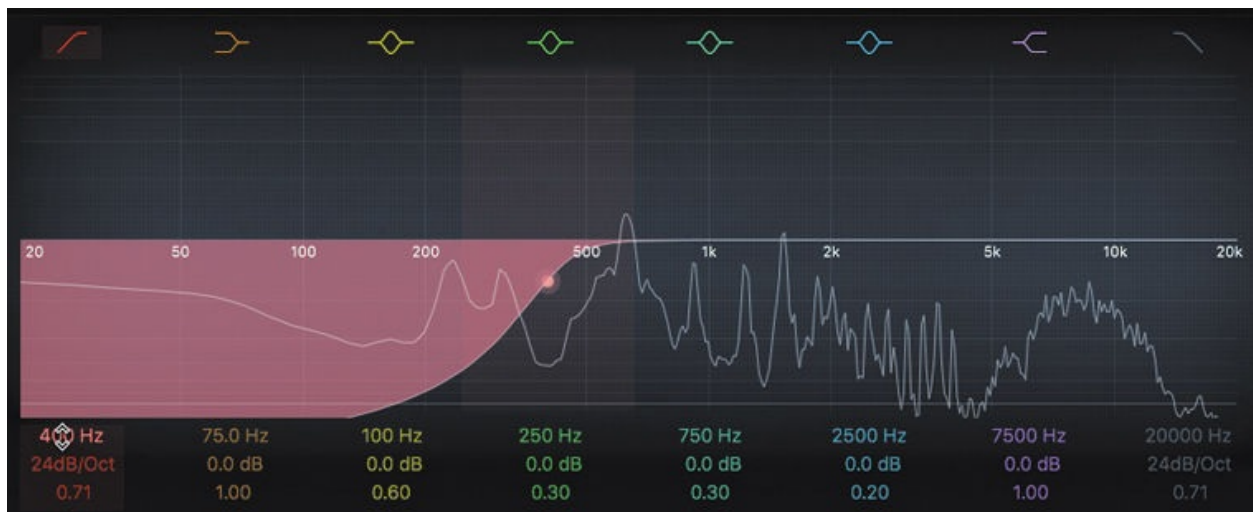
You will filter out those very low frequencies to attenuate the unwanted noises.

5 At the upper left, click the first Band On/Off button to turn on that EQ band.
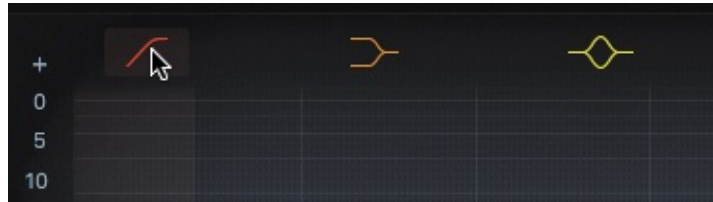
The first EQ band's shape appears on the graphic display. You can see that the low frequencies are slightly attenuated around 20 Hz.

6 In the EQ band field, drag the Frequency parameter of the first band up to 400 Hz.

The EQ band shape updates in the graphic display. In the frequency curve displayed by the Analyzer, you can watch the low-frequency content disappear from the vocal signal. You can hear the low-frequency noises completely disappear. However, you've changed the timbre of the vocals, and they now sound a bit thinner, almost as if coming from a cheap sound system.

You have to adjust the Frequency parameter of that first band so that the low-frequency noises are attenuated as much as possible without affecting the vocals.

7 Drag the frequency of the first band down to 220 Hz.

You can hear the full vocal sound, and not the low-frequency noises.

Mission accomplished!

Now you will attenuate the low/mid frequencies to remove the "mud" from the vocals. Instead of adjusting the numerical settings in the parameter section, you'll drag the pointer in the graphic display to adjust the shape of individual bands.

8 Position the mouse pointer over the upper half of the graphic display, and without pressing the mouse button, move the pointer from left to right.

| 20 | 50 | 100 | 200 | 500 | 1k | 2k | 5k | 10k | 20k |

220 Hz	75.0 Hz	100 Hz	250 Hz	750 Hz	2500 Hz	7500 Hz	20000 Hz
24dB/Oct	0.0 dB	0.0 dB	0.0 dB	0.0 dB	0.0 dB	0.0 dB	24dB/Oct
0.71	1.00	0.60	0.30	0.30	0.20	1.00	0.71

As you move the pointer horizontally, the EQ band fields are shaded in different colors at the bottom of the Channel EQ to show you which EQ band is selected. You can shape the curve of the selected band by dragging in the graphic display: ► To adjust the gain, drag vertically.

► To adjust the frequency, drag horizontally.

► To adjust the Q (or width, or resonance), vertically Option-Command-drag the pivot point (which appears at that band's frequency).

You first need to adjust the band's gain to see its shape on the graphic display.

9 Position the pointer to select the fifth band, which is currently set to a frequency of 750 Hz.

10 Drag down so that the Gain parameter below reads –7.0 dB.

220 Hz	75.0 Hz	100 Hz	250 Hz	750 Hz	2500 Hz	7500 Hz	20000 Hz
24dB/Oct	0.0 dB	0.0 dB	0.0 dB	-7.0 dB	0.0 dB	0.0 dB	24dB/Oct
0.71	1.00	0.60	0.30	0.30	0.20	1.00	0.71

The shape of the selected EQ band appears on the graphic display, and the settings below are adjusted accordingly.

Now, while listening to the vocal, you will adjust both the Q and the frequency of the EQ band you are attenuating.

11 Drag the band to the left to set its frequency to 470 Hz.

12 Option-Command-drag the pivot point down to widen the band of EQ until the Q field for that band reads 0.10.

| 220 Hz	75.0 Hz	100 Hz	250 Hz	470 Hz	2500 Hz	7500 Hz	20000 Hz
24dB/Oct	0.0 dB	0.0 dB	0.0 dB	-7.0 dB	0.0 dB	0.0 dB	24dB/Oct
0.71	1.00	0.60	0.30	0.10	0.20	1.00	0.71

The vocal sounds less muddy already. Remember to click that EQ band's On/Off button to compare the vocal sound with and without that EQ band applied.

Now you can add some air to the vocal by boosting its high frequencies.

13 Drag the values in the EQ band fields, and set the next-to-last EQ band to the following parameters: ► Set the frequency to 6800 Hz.

► Set the gain to +3.0 dB.

► Leave the Q at 1.00.
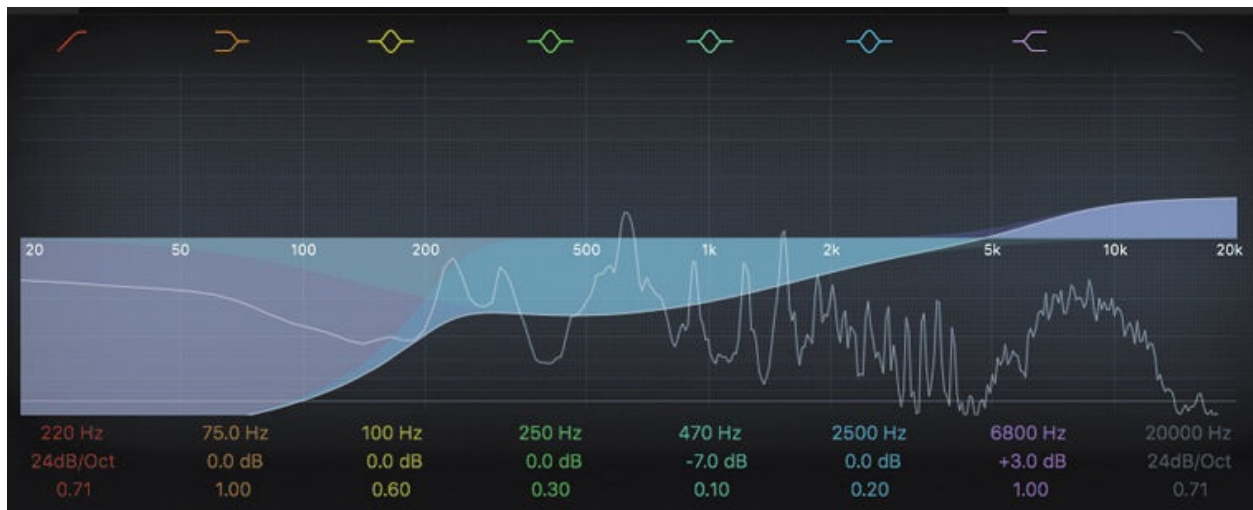
The vocal now sounds a bit less dull, and more present. Let's compare the sound of the vocal with and without the Channel EQ.

14 In the plug-in header, click the On/Off button.

The button is dimmed to indicate that the Channel EQ is off. On the Channel EQ graphic display, the curve disappears because the audio signal is no longer routed through the plug-in.

You can now hear the unprocessed vocal recording. The low-frequency noises are back, and the vocal is boomy and slightly dull.

15 Turn the Channel EQ on. The noises disappear, and the vocals sound clear and distinct.

16 Press S to unsolo the Ami Vox track.

Let's now adjust the level of the Ami Vox track.

17 On the Ami Vox channel strip in the inspector, drag the Volume fader all the way down.
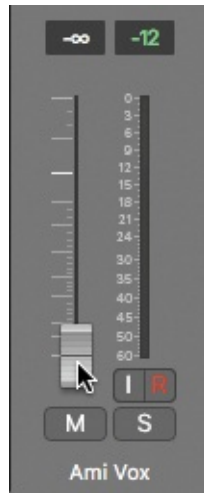
Ami Vox

Listen to the verse a few times to become familiar with the sound of the mix without the Ami Vox track.

18 Slowly raise the Ami Vox Volume fader to around –9.0 dB.

As you try to find the right level for the vocal you just EQ'ed, you may notice that some words are louder than others. You will later use a compressor to give the vocal a more consistent level.

19 Stop playback.

Instead of closing the Channel EQ window, you can recall the current screenset to return it to its locked state.

20 Press 1 to recall screenset 1.

The screenset is recalled, and the Channel EQ window is no longer open.

By applying an EQ plug-in to the vocal, you shaped its frequency spectrum to eliminate unwanted low-frequency noises and clarify the vocal, establishing its appropriate place in the frequency spectrum of the mix.

Using Delay and Reverberation

In a natural environment, an instrument's sound is reflected by the floor, walls, and ceiling of the room. Those reflections, combined with the direct sound of the instrument, allow listeners to determine the distance from the instrument to their ears and understand the nature of the acoustic space.

When recording instruments, you can choose to record in an absorbent studio to primarily record the direct sound of each instrument. Then you apply delay and reverberation plug-ins to create artificial reflections, giving you total control over the apparent placement and depth of the instruments.

Adding Delay to the Vocals

Adding Delay to the Vocals

Delay can be applied so you can clearly hear the distinct repeats of the sound echoing, or it can be applied with shorter delay times to create an ambience similar to a reverb.

You will now add a Tape Delay plug-in to a vocal track to create a classic slap-back delay, a vintage effect popularized by the Beatles, who often used it liberally on John Lennon's voice.

1 In the Tracks area, select the Duvid Vox track (track 30) to show its channel strip in the inspector, and solo it.

29		M S	Ami Vox	●
30		M S	Duvid Vox	●
31		M S	Gang Vox 1	●

2 Start playback.

3 On the Duvid Vox channel strip, in the Audio FX area, click below the Compressor, and choose Delay > Tape Delay.

| ○	Input
Channel EQ	
Compressor	
Tape Delay	

The Tape Delay plug-in is inserted, and its window opens.

You hear the reflections applied to the vocals as an echo. By default, the Tape Delay is automatically synchronized to the project tempo so that reflections are produced every quarter note. To get a slap-back effect, you needn't synchronize the delay to the grid, and you want the repeats to occur faster.

4 Click the Tempo Sync button to disable it.
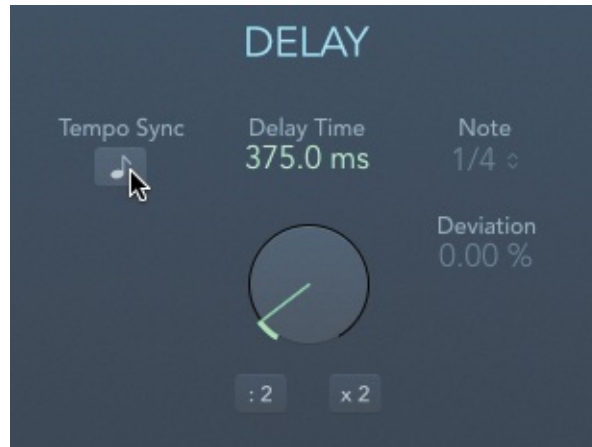
To the right of the Tempo Sync button, the Delay Time is now highlighted, indicating that you can change its value directly in the Delay Time field.

**5** Double-click the Delay Time field, and enter *90* ms.
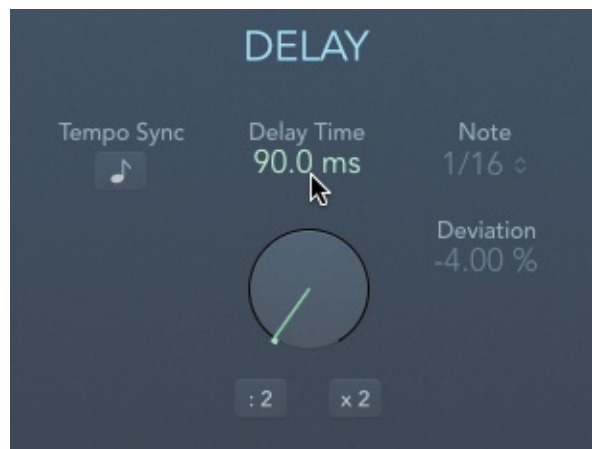
The delay is now very short, and you can no longer hear distinct repeats. Instead the delay creates an ambiance similar to recording the vocal in a very small reflective room.

You can drag the two Output sliders on the bottom right to adjust the level balance between Dry (the sound not affected by the plug-in) and Wet (the reflections generated by the plug-in). You will bring the Dry signal all the way up to its maximum level so that you maintain the same level of direct sound as you had before inserting the plug-in.

6 Make sure the Dry slider is up to 100%.

You can now adjust the level of the delay effect with the Wet slider. Let's make the effect a little more subtle.

7 Drag the Wet slider down to 14%.
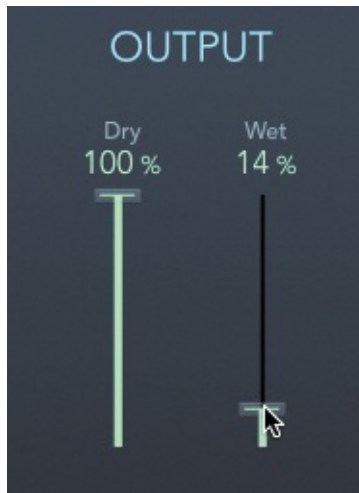
Here again, because you were focused on the sound of the delay, you may feel as if you just turned the effect down so much that you don't hear it any more. To fully hear the effect of the Tape Delay, let's toggle that plug-in on and off.

8 In the plug-in header, click the On/Off button to turn off the plug-in. The vocal sounds much drier.

9 Turn the plug-in back on. The Tape Delay adds a little bit of slap-back echo, creating an ambience while keeping the vocal sound fairly natural.

10 Keep the Duvid Vox track soloed, and stop playback.

11 Close the Tape Delay plug-in window.

Adding Reverb Using Bus Sends

You could insert a reverb plug-in directly into the channel strip of the instrument you want to process, just as you inserted a delay plug-in in the previous exercise. However, when you use reverb to simulate the sound of instruments in a room, applying an individual reverb plug-in for each instrument would require a lot of processing power. You would also have to readjust the parameters of each reverb plug-in every time you wanted to change the acoustics of your artificial room.

Instead, you can use a bus send to route some of the signal from a channel strip to a new auxiliary channel strip. When you have set up an aux channel strip with the desired reverb plug-in, you can add reverberation to any instrument by sending some of its signal to that aux for processing.

In this exercise, you will use a bus send to route a vocal track to an aux, and insert the reverberation plug-in on the aux channel strip. Later you'll send another vocal track to the same aux to give it the same reverb sound.

A track is still soloed from the previous exercise. To unsolo it and solo a new track, you can Option-click the new track's Solo button.

1 Select the Gang Vox 1 track (track 31) and Option-click its Solo button. The Duvid Vox track is unsoloed, and the Gang Vox 1 track is soloed.

2 On the Gang Vox 1 channel strip, click the Send slot, and choose Bus > Bus 7.

When you click a Send slot and choose an available bus, a new aux channel strip is added to the Mixer. In the inspector, the channel strip to the right shows the new Aux 6 channel strip, and you can see the signal path from the lead vocal channel strip to the aux channel strip. At the top of the Aux 6 channel strip, the input is set to Bus 7.

Tip

On the left channel strip, click the desired Send or Output slot to display the corresponding output or auxiliary channel strip on the right in the inspector. When you click a send, make sure that you click the right area of the Send slot to avoid toggling its On/Off

Let's insert the reverb plug-in on the aux.

3 On the Aux 6 channel strip, click the top of the Audio FX section, and choose Reverb > Space Designer to insert that plug-in.

4 In the Marker track, drag the CH2 marker up into the ruler, and start playback.

The vocals sound dry, and no movement appears on the Aux 6 channel strip's meter. On the Gang Vox 1 channel strip, you need to raise the Send level knob to determine how much of the vocal signal you're sending to Aux 6 for processing by the reverb.

5 On the Gang Vox 1 channel strip, Option-click the Send Level knob next to Bus 7.

The Bus 7 send briefly displays the send level, 0.0 (dB), and you can hear the reverb. Let's choose a bigger reverb sound.

6 In the Space Designer plug-in header, from the Settings pop-up menu, choose Large Spaces > Rooms > 02.3s Clean Room.
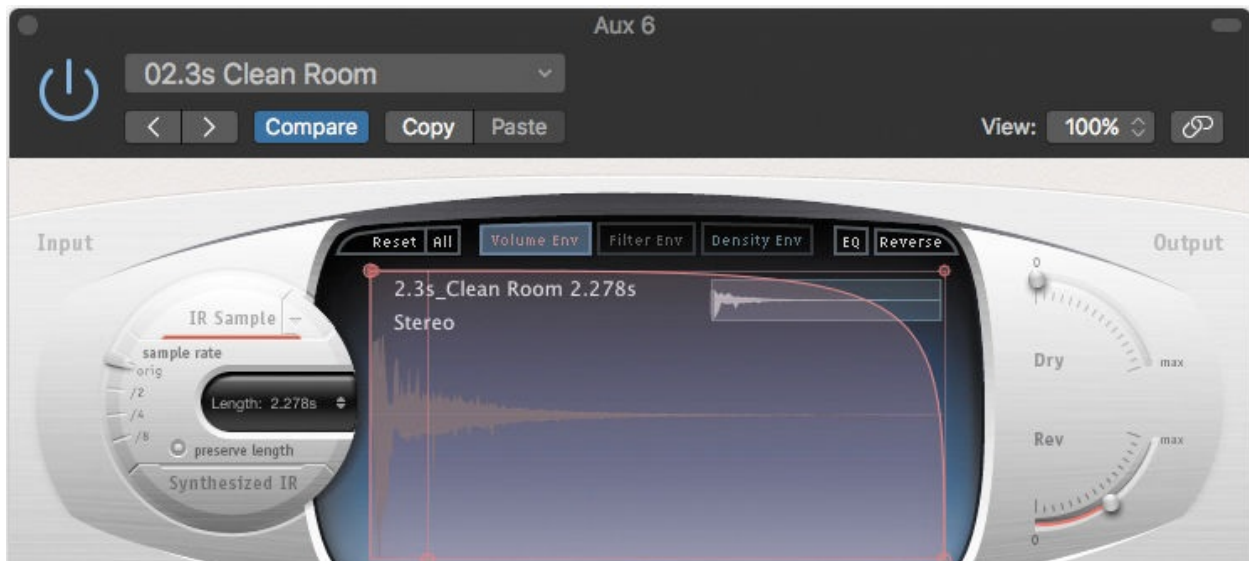
The reverb sound is bigger. In fact, you now have too much reverb. You'll need to adjust the send level on the Gang Vox 1 channel strip to determine how much of the dry vocal sound is routed to the reverb plug-in.

7 On the Gang Vox 1 channel strip, drag the Bus 7 Send Level knob down to around –12.0 (dB).

You can now send the audio signal from any channel strip to Bus 7 to route it to Aux 6 and add the same reverb effect to its sound. The Gang Vox 1 track was doubled on the Gang Vox 2 track, so let's open the Mixer and send the Gang Vox 2 track to the same reverb.

8 Press 2 to recall screenset 2.

Because you previously selected the Gang Vox 1 track in the Tracks area, the Gang Vox 1 channel strip (31) is selected in the Mixer, making it easy to locate.

9 Option-click the Gang Vox 2 channel strip's Solo button. Gang Vox 1 is unsoloed, and Gang Vox 2 is soloed. You can hear the dry recording.

10 On the Gang Vox 2 channel strip (32), click the top Send slot, and choose Bus > Bus 7 → Aux 6.

11 Drag the Send Level knob up to around –12.0 (dB).
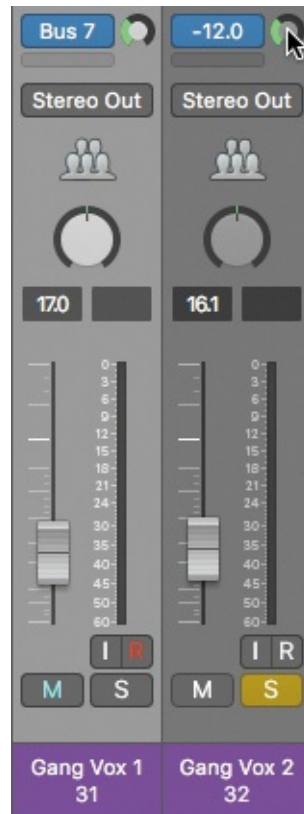
Let's hear both tracks together.

12 Click the Solo button at the bottom of the Gang Vox 1 channel strip.

Both tracks and their reverbs sound as if they are located in the center of the stereo field. Let's spread them out to widen the ensemble.

13 Pan the Gang Vox 1 to the left and the Gang Vox 2 to the right, choosing Pan values around –40 and +40.

The vocals are spread out in the stereo field and sound closer to you. However, the reverb stayed in the center. Let's look at the Aux 6 channel strip to see what happened.

14 Place the mouse pointer over a Bus 7 send on one of the Gang Vox channel strips, and double-click the 7.

The Aux 6 channel strip is selected (and if necessary, the Mixer scrolls to show it). At the top of the channel strip, the Format button next to the input (Bus 7) shows a single circle, indicating a mono input. Let's change this channel's input format to stereo.

15 On the Aux 6 channel strip, click the Format button.

The reverb of each track is positioned according to the track's panning in the stereo field, and the reverb now sounds wider.

You can click the Solo Kill/Recall button in the Tracks area to toggle the soloed state of currently soloed tracks.

16 Press 1 to recall screenset 1.

17 At the top of the track headers, click the Solo Kill/Recall button (or press Control-Command-Option-S).
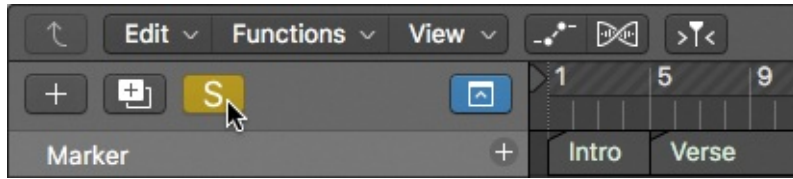
Both Gang Vox tracks are unsoloed, and you can hear the entire mix. If you feel that the Gang Vox tracks require further attention, you can easily solo them again.

18 Click the Solo Kill/Recall button.

Both Gang Vox tracks are soloed again.

19 Click the Solo Kill/Recall button to unsolo all tracks.

Using Dynamic Processing Plug-ins

The dynamic range of a recording is the difference between the quietest and loudest parts of the recording. Sometimes too much dynamic range in a recording can be a problem because the loudest parts become too loud and the quietest parts drop to barely audible levels. You can use dynamic processing plug-ins such as a compressor or a limiter to adjust the dynamic range of an audio signal, usually to make the audio signal level more consistent over time.

Dynamic processing can be applied to a single channel strip, to a submix of a group of instruments, or to an entire mix.

Using the Compressor

When recording an instrument, the musician rarely plays all the notes at the same volume. Singers need more energy to reach higher pitches, and they relax to sing low pitches, resulting in uneven loudness throughout a melody line. This variation can become a challenge when mixing, because some of the notes stick out and others are buried in the mix.

A compressor attenuates a signal when its level reaches a specific threshold. You can use it to lower the volume of loud notes and then raise the overall level of the instrument to increase the volume of softer notes.

In this exercise, you will apply a compressor plug-in to even out the dynamic range of a vocal track, making sure that you can hear all the words at the same level.

1 In the Tracks area, drag the V2 marker up to the ruler.

2 Select and solo the Ami Vox track (track 29).

3 Start playback.

The level of the vocal recording fluctuates as the singer hits different pitches. On the level meter in the Ami Vox channel strip, you can see higher notes reach higher levels than lower notes.

4 On the Ami Vox channel strip, click below the Channel EQ plug-in, and choose Dynamics > Compressor.

The Gain Reduction meter shows by how many decibels the Compressor is attenuating the audio signal. The little bit of activity on the meter indicates that the Compressor is barely attenuating the vocals.

Note

On channel strips, you can also use the Gain Reduction meter (below the Setting button) to see how much gain reduction is applied by the Compressor plug-in, or any other dynamic plug-in inserted in the Audio FX section.

Although the Compressor plug-in has many parameters, you will adjust only the most important parameters, which are located below the Gain Reduction meter: the Threshold, Ratio, and Make Up knobs, and the Auto Gain buttons. The Make Up and Auto Gain parameters help compensate for the gain reduction by applying a constant gain at the output of the Compressor. To focus only on the gain reduction applied by the Compressor, let's eliminate all gain applied at the output.

5 Ensure that the Make Up knob is set to 0.0 dB, and click the Auto Gain Off button.

The Compressor can now apply only gain reduction, so it can only turn down the volume when the vocals reach levels higher than the Threshold parameter. Remember to turn the Compressor on and off as you adjust it to compare the sound of the vocal with and without the compression effect.

Above the Gain Reduction meter, you can choose from different models based on vintage hardware compressors. Except for Platinum, which is a transparent compressor, each circuit type adds its own color to the signal.

6 Click the Vintage Opto button.

The graphic interface of the Compressor changes to adopt the look of the classic analog compressor that the Vintage Opto mode is modeled after: the Universal Audio LA-2A.

The Gain Reduction meter is inactive. Right now, the Compressor isn't having any effect on the vocals. You will now adjust the threshold so that all notes trigger the Compressor.

7 Drag the Threshold knob down to –34.5 dB.

On the Gain Reduction meter, you can see the Compressor being triggered

by all the notes, working harder on higher-pitched notes, while barely attenuating the lower-pitched notes.

You can adjust the amount of compression with the Ratio slider, which changes the ratio by which the signal is reduced when it exceeds the threshold.

8 Ensure that the Ratio knob is set to 2.0:1.
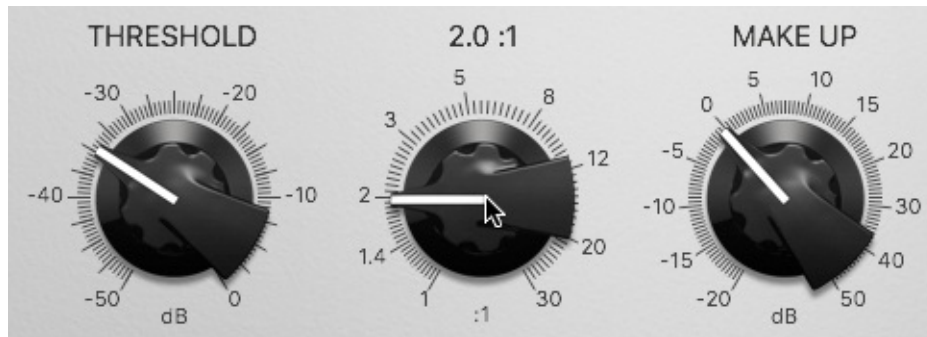
The vocal level sounds more consistent. You can clearly hear the singer even when he sings lower pitches, and higher-pitched notes no longer jump out.

Don't hesitate to experiment using higher ratio values to increase the amounts of compression. Be aware, however, that high ratio values may attenuate the high-pitched notes too much and produce an unnatural-sounding result.

Now that the Compressor is reducing the gain, the vocal sounds lower in volume, so you can apply some make-up gain.

9 Drag the Make Up knob up to 2.5 dB.

The vocals have roughly the same overall loudness as when you first started compressing them, making it easy to hear the effect of the Compressor by toggling it on and off.

10 Stop playback.

11 Unsolo the Ami Vox track, turn off Cycle mode, and close the

Compressor plug-in window.

Compressing and Limiting the Mix

On a professional project, you would usually send your final mix to a mastering engineer, who would put a final polish on the audio file using subtle amounts of EQ, compression, reverb, or other processing.

When you don't have the budget to hire a mastering engineer, you can master your own mix by inserting plug-ins on the Stereo Out channel strip, as in this exercise. You will start by using a compressor to make the mix level more consistent throughout the song, and then apply a limiter to raise the perceived loudness without clipping the Stereo Out channel strip audio.

1 Start playback at the beginning of the song.

2 On the Output channel strip, click the Audio FX area, and choose Dynamics > Compressor.

This time you will use a setting designed to emulate the soft compression of analog tape recorders.

3 In the Compressor, from the Setting menu, choose Compressor by Type > Platinum Analog Tape.

Note

> If you've adjusted the volume of the song using the Volume slider that appears in the control bar or the Volume fader on the Master channel strip in the Mixer, make sure you Option-click it now to return it to 0 dB.

The Output channel strip is clipping, as indicated by the red peak level display. In the Compressor window, the gain reduction shows a few decibels of gain reduction, but the Auto Gain parameter applies enough make-up gain to make the mix clip.

**4** In the Compressor, click the Auto Gain Off button.

The peak level display continues to display the most recent maximum peak value. You need to reset it to determine if the new compressor setting still clips the mix.

5 On the Output channel strip, click the peak level display to reset it. The mix is no longer clipping.

You will now insert a limiter after the Compressor to raise the gain without clipping the output. A limiter works similarly to a compressor, but attenuates the signal so that the output signal never exceeds a specific volume level.

6 On the Output channel strip, click below the Compressor, and choose Dynamics > Adaptive Limiter.

Adaptive Limiter
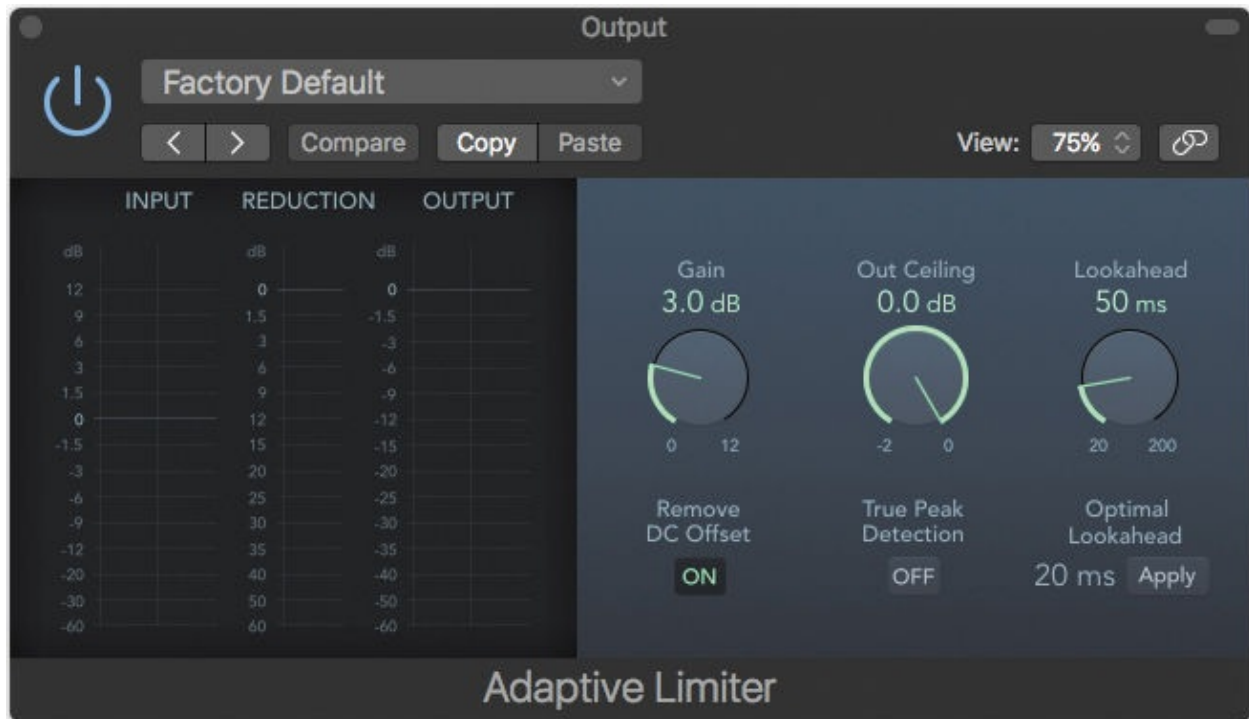
The Input meters display the signal level at the input of the plug-in; the Reduction meters display the level of gain attenuation applied by the limiter; the OUTPUT meters display the signal level at the output of the plug-in. The Out Ceiling knob is set to 0.0 dB, ensuring that the audio signal will never go over 0 dBFS on the Stereo Out channel strip.

To adjust the Adaptive Limiter, dial the Gain knob to your taste: the more gain you apply, the louder it will sound, but also the more distortion will be generated by the limiter. Deciding on the right balance between loudness and acceptable distortion can be influenced by many factors like the music genre (jazz or pop?), or how the music will be distributed (movie theater or Internet streaming?).

7 Drag the gain up to 8.7 dB.

Keep playing the song, and make sure you don't hear any unwanted distortion.

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8 In the inspector, on the Output channel strips, click the Compressor's and the Adaptive Limiter's On/Off buttons to toggle them on and off. You can also drag down on the two buttons to quickly toggle both plug-ins.

The Compressor makes your mix sound more consistent. The Adaptive Limiter gives you a nice gain boost on the whole mix and ensures that no clipping occurs at the output.

You have finished your mix using effect plug-ins and adjusting the four main parameters of the instrument sounds (volume levels, pan position, frequency, and distance) to give each sound its own place in the stereo sound field.

Using a Few Tips and Tricks

As with any other art, mixing requires a combination of skill, experience, and talent. It takes practice to learn how to apply mixing techniques efficiently, and even more practice to learn to listen. Here are a few tips and tricks that will help you perfect your craft and become better at mixing your projects.

Take a Break

After you mix for a while, listening to the same song for the hundredth time, you can lose your objectivity and experience ear fatigue. Take frequent short breaks

while mixing, and return to the mix with rested ears. You will be able to better judge your results.

Listen to Your Mix Outside the Studio

When you feel that your mix is pretty advanced and you are happy with the way it sounds in your studio, copy it to a portable music player and listen to it in another room or, even better, in your car while driving. You will probably hear things you didn't notice in your studio, and miss things you could hear clearly in your studio. You can take notes and return to your studio to rework the mix. Obviously, the mix will never sound the same in the studio and in the car, but it's the mixing engineer's job to make sure that all the instruments can be heard in most situations.

Compare Your Mix with Commercial Mixes

Compare your mix with commercial mixes you like. Build a small library of good-sounding mixes in the same genre of music as the songs you are mixing. You can open a new Logic project, and place your mix on one track and a professional mix on another track so that you can solo and compare them.

Lesson Review

1. Identify the four main instrumental sound components you can adjust to give each instrument its place in a mix.

2. For what purpose do you use a bus send?

3. How can you make a group of tracks appear as one track in the Tracks area without changing its audio routing?

4. How can you submix a group of tracks to, for example, process the submix with audio effect plug-ins?

5. What does a compressor do?

6. What does a limiter do?

7. In the inspector, how can you choose which channel strip is displayed in the right channel strip?

8. How do you change the mono or stereo format of the input of a channel strip?

9. How can you toggle the soloed state of currently soloed tracks?

Answers

1. Volume level, pan position, frequency spectrum, and distance

2. You use a bus send to route some of the signal from a channel strip to an aux channel strip, usually to be processed by plug-ins.

3. Select the tracks, choose Track > Create Track Stack, and choose a Folder Stack.

4. Select the tracks, choose Track > Create Track Stack, and choose a Summing Stack.

5. It attenuates the level of a signal after that signal goes over a certain threshold, giving the signal a more consistent level.

6. It works in a similar fashion to a compressor, but it makes sure that the signal is attenuated so it never goes over a specific output ceiling.

7. On the left channel strip, click any Send slot or the Output slot to display the corresponding channel strip on the right.

8. Click the Format button in the Input slot.

9. At the top of the track headers, click the Solo Kill/Recall button.

Keyboard Shortcuts

Tracks

Command-Shift-D	Creates a Track Stack
Control-Option-Command-S	Toggles all soloed tracks' solo buttons
S	Solos the selected track

Windows

Command-2	Opens a Mixer window
Number keys	Recalls the corresponding screenset (on alphanumerical keypad)
V	Hides or shows all plug-in windows