

new/usr/src/cmd/audio/audioplay/audioplay.c

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*****
28755 Thu Jan 17 14:27:12 2019
new/usr/src/cmd/audio/audioplay/audioplay.c
10101 audio tools need smatch fixes
*****
1 /*
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21 /*
22 * Copyright 2009 Sun Microsystems, Inc. All rights reserved.
23 * Use is subject to license terms.
24 */
26 /*
27  * Copyright (c) 2018, Joyent, Inc.
28  */
30 /* Command-line audio play utility */
32 #include <stdio.h>
33 #include <errno.h>
34 #include <ctype.h>
35 #include <string.h>
36 #include <stdlib.h>
37 #include <fcntl.h>
38 #include <signal.h>
39 #include <locale.h>
40 #include <limits.h> /* All occurrences of INT_MAX used to be ~0 (by MCA) */
41 #include <unistd.h>
42 #include <stropts.h>
43 #include <sys/types.h>
44 #include <sys/file.h>
45 #include <sys/stat.h>
46 #include <sys/param.h>
47 #include <sys/ioctl.h>
48 #include <sys/mman.h>
49 #include <netinet/in.h>
51 #include <libaudio.h>
52 #include <audio_device.h>
53 #include <audio_encode.h>
55 /* localization stuff */
56 #define MGET(s) (char *)gettext(s)
58 #if !defined(TEXT_DOMAIN) /* Should be defined by cc -D */
59 #define TEXT_DOMAIN "SYS_TEST" /* Use this only if it weren't */
60 #endif
```

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62 #define Error (void) fprintf
64
65 /* Local variables */
66 static char *prog;
68 static char prog_opts[] = "VEiv:d:?" ; /* getopt() flags */
70 static char *Stdin;
72 #define MAX_GAIN (100) /* maximum gain */
74 /*
75  * This defines the tolerable sample rate error as a ratio between the
76  * sample rates of the audio data and the audio device.
77  */
78 #define SAMPLE_RATE_THRESHOLD (.01)
80 #define BUFFER_LEN 10 /* seconds - for file i/o */
81 #define ADPCM_SIZE (1000*8) /* adpcm conversion output buf size */
82 #define SWAP_SIZE (8192)
83 /* swap bytes conversion output buf size */
85 static unsigned Volume = INT_MAX; /* output volume */
86 static double Savevol; /* saved volume level */
88 static int Verbose = FALSE; /* verbose messages */
89 static int Immediate = FALSE;
90 /* don't hang waiting for device */
91 static int Errdetect = FALSE; /* don't worry about underrun */
92 static char *Audio_dev = "/dev/audio";
94 static int NetEndian = TRUE; /* endian nature of the machine */
96 static int Audio_fd = -1; /* file descriptor for audio device */
97
98 static int Audio_ctlfd = -1; /* file descriptor for control device */
99
100 static Audio_hdr Save_hdr; /* saved audio header for device */
101
102 static Audio_hdr Dev_hdr; /* audio header for device */
103 static char *Ifile; /* current filename */
104 static Audio_hdr File_hdr; /* audio header for file */
105 static unsigned Decode = AUDIO_ENCODING_NONE; /* decode type, if any */
106
108 static unsigned char *buf = NULL; /* dynamically alloc'd */
109 static unsigned bufsiz = 0; /* size of output buffer */
110 static unsigned char adpcm_buf[ADPCM_SIZE + 32]; /* for adpcm conversion */
111
112 static unsigned char swap_buf[SWAP_SIZE + 32]; /* for byte swap conversion */
113
114 static unsigned char *inbuf; /* current input buffer pointer */
115
116 static unsigned insiz; /* current input buffer size */
118 /*
119  * The decode_g72x() function is capable of decoding only one channel
120  * at a time and so multichannel data must be decomposed (using demux())
121  * function below ) into its constituent channels and each passed
122  * separately to the decode_g72x() function. Encoded input channels are
123  * stored in **in_ch_data and decoded output channels in **out_ch_data.
124  * Once each channel has been decoded they are recombined (see mux())
125  * function below) before being written to the audio device. For each
126  * channel and adpcm state structure is created.
127  */
```

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129 /* adpcm state structures */
130 static struct audio_g72x_state *adpcm_state = NULL;
131 static unsigned char **in_ch_data = NULL; /* input channels */
132 static unsigned char **out_ch_data = NULL; /* output channels */
133 static int out_ch_size; /* output channel size */

135 static char *Audio_path = NULL;
136 /* path to search for audio files */

138 /* Global variables */
139 extern int getopt(int, char *const *, const char *);
140 extern int optind;
141 extern char *optarg;

143 /* Local functions */
144 static void usage(void);
145 static void sigint(int sig);
146 static void open_audio(void);
147 static int path_open(char *fname, int flags, mode_t mode, char *path);
148 static int parse_unsigned(char *str, unsigned *dst, char *flag);
149 static int reconfig(void);
150 static void initmux(int unitsz, int unitsp);
151 static void demux(int unitsz, int cnt);
152 static void mux(char *);
153 static void freemux(void);

156 static void
157 usage(void)
158 {
159     Error(stderr, MGET("Play an audio file -- usage:\n")
160           "\t%s [-iV] [-v vol] [-d dev] [file ...]\n"
161           "where:\n"
162           "\t-i\tDon't hang if audio device is busy\n"
163           "\t-v\tPrint verbose warning messages\n"
164           "\t-V\tSet output volume (0 - %d)\n"
165           "\t-d\tSpecify audio device (default: /dev/audio)\n"
166           "\tfile\tList of files to play\n"
167           "\t\tIf no files specified, read stdin\n"),
168          prog, MAX_GAIN);
169     exit(1);
170 }

```

unchanged portion omitted

```

250 /* Play a list of audio files. */
251 int
252 main(int argc, char **argv) {
253     int errorStatus = 0;
254     int i;
255     int c;
256     int cnt;
257     int file_type;
258     int rem;
259     int outsiz;
260     int tsize;
261     int len;
262     int err;
263     int ifd;
264     int stdinseen;
265     int regular;
266     int swapBytes;
267     int frame;
268     char *outbuf;
269     caddr_t mapaddr;
270     struct stat st;

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271     char *cp;
272     char ctldev[MAXPATHLEN];

274     (void) setlocale(LC_ALL, "");
275     (void) textdomain(TEXT_DOMAIN);

277     /* Get the program name */
278     prog = strrchr(argv[0], '/');
279     if (prog == NULL)
280         prog = argv[0];
281     else
282         prog++;
283     Stdin = MGET("(stdin)");

285     /* Check AUDIODEV environment for audio device name */
286     if (cp = getenv("AUDIODEV")) {
287         Audio_dev = cp;
288     }

290     /* Parse the command line arguments */
291     err = 0;
292     while ((i = getopt(argc, argv, prog_opts)) != EOF) {
293         switch (i) {
294             case 'v':
295                 if (parse_unsigned(optarg, &Volume, "-v")) {
296                     err++;
297                 } else if (Volume > MAX_GAIN) {
298                     Error(stderr, MGET("%s: invalid value "\n"
299                                     "for -v\n"), prog);
300                     err++;
301                 }
302                 break;
303             case 'd':
304                 Audio_dev = optarg;
305                 break;
306             case 'V':
307                 Verbose = TRUE;
308                 break;
309             case 'E':
310                 Errdetect = TRUE;
311                 break;
312             case 'i':
313                 Immediate = TRUE;
314                 break;
315             case '?':
316                 usage();
317                 /*NOTREACHED*/
318             }
319     }
320     if (err > 0)
321         exit(1);

323     argc -= optind; /* update arg pointers */
324     argv += optind;

326     /* Validate and open the audio device */
327     err = stat(Audio_dev, &st);
328     if (err < 0) {
329         Error(stderr, MGET("%s: cannot stat "), prog);
330         perror(Audio_dev);
331         exit(1);
332     }
333     if (!S_ISCHR(st.st_mode)) {
334         Error(stderr, MGET("%s: %s is not an audio device\n"), prog,
335               Audio_dev);
336         exit(1);

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337     }
339     /* This should probably use audio_ctl instead of open_audio */
340     if ((argc <= 0) && isatty(fileno(stdin))) {
341         Error(stderr, MGET("%s: No files and stdin is a tty.\n"), prog);
342         exit(1);
343     }
345     /* Check on the -i status now. */
346     Audio_fd = open(Audio_dev, O_WRONLY | O_NONBLOCK);
347     if ((Audio_fd < 0) && (errno == EBUSY)) {
348         if (Immediate) {
349             Error(stderr, MGET("%s: %s is busy\n"), prog,
350                 Audio_dev);
351             exit(1);
352         }
353     }
354     (void) close(Audio_fd);
355     Audio_fd = -1;
357     /* Try to open the control device and save the current format */
358     (void) sprintf(ctlddev, sizeof(ctlddev), "%sctl", Audio_dev);
359     Audio_ctlfd = open(ctlddev, O_RDWR);
360     if (Audio_ctlfd >= 0) {
361         /*
362          * wait for the device to become available then get the
363          * controls. We want to save the format that is left when the
364          * device is in a quiescent state. So wait until then.
365          */
366         Audio_fd = open(Audio_dev, O_WRONLY);
367         (void) close(Audio_fd);
368         Audio_fd = -1;
369         if (audio_get_play_config(Audio_ctlfd, &Save_hdr)
370             != AUDIO_SUCCESS) {
371             (void) close(Audio_ctlfd);
372             Audio_ctlfd = -1;
373         }
374     }
376     /* store AUDIOPATH so we don't keep doing getenv() */
377     Audio_path = getenv("AUDIOPATH");
379     /* Set up SIGINT handler to flush output */
380     (void) signal(SIGINT, sigint);
382     /* Set the endian nature of the machine. */
383     if ((ulong_t)1 != htonl((ulong_t)1)) {
384         NetEndian = FALSE;
385     }
387     /* If no filenames, read stdin */
388     stdinseen = FALSE;
389     if (argc <= 0) {
390         Ifile = Stdin;
391     } else {
392         Ifile = *argv++;
393         argc--;
394     }
396     /* Loop through all filenames */
397     do {
398         /* Interpret "-" filename to mean stdin */
399         if (strcmp(Ifile, "-") == 0)
400             Ifile = Stdin;
402         if (Ifile == Stdin) {

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403             if (stdinseen) {
404                 Error(stderr,
405                     MGET("%s: stdin already processed\n"),
406                     prog);
407                 goto nextfile;
408             }
409             stdinseen = TRUE;
410             ifd = fileno(stdin);
411         } else {
412             if ((ifd = path_open(Ifile, O_RDONLY, 0, Audio_path))
413                 < 0) {
414                 Error(stderr, MGET("%s: cannot open "), prog);
415                 perror(Ifile);
416                 errorStatus++;
417                 goto nextfile;
418             }
419         }
421         /* Check to make sure this is an audio file */
422         err = audio_read_filehdr(ifd, &File_hdr, &file_type,
423             (char *)NULL, 0);
424         if (err != AUDIO_SUCCESS) {
425             Error(stderr,
426                 MGET("%s: %s is not a valid audio file\n"),
427                 prog, Ifile);
428             errorStatus++;
429             goto closeinput;
430         }
432         /* If G.72X adpcm, set flags for conversion */
433         if ((File_hdr.encoding == AUDIO_ENCODING_G721) &&
434             (File_hdr.samples_per_unit == 2) &&
435             (File_hdr.bytes_per_unit == 1)) {
436             Decode = AUDIO_ENCODING_G721;
437             File_hdr.encoding = AUDIO_ENCODING_ULAW;
438             File_hdr.samples_per_unit = 1;
439             File_hdr.bytes_per_unit = 1;
440             adpcm_state = (struct audio_g72x_state *)malloc
441                 (sizeof(*adpcm_state) * File_hdr.channels);
442             for (i = 0; i < File_hdr.channels; i++) {
443                 g721_init_state(&adpcm_state[i]);
444             }
445         } else if ((File_hdr.encoding == AUDIO_ENCODING_G723) &&
446             (File_hdr.samples_per_unit == 8) &&
447             (File_hdr.bytes_per_unit == 3)) {
448             Decode = AUDIO_ENCODING_G723;
449             File_hdr.encoding = AUDIO_ENCODING_ULAW;
450             File_hdr.samples_per_unit = 1;
451             File_hdr.bytes_per_unit = 1;
452             adpcm_state = (struct audio_g72x_state *)malloc
453                 (sizeof(*adpcm_state) * File_hdr.channels);
454             for (i = 0; i < File_hdr.channels; i++) {
455                 g723_init_state(&adpcm_state[i]);
456             }
457         } else {
458             Decode = AUDIO_ENCODING_NONE;
459         }
461         /* Check the device configuration */
462         open_audio();
463         if (audio_cmp_hdr(&Dev_hdr, &File_hdr) != 0) {
464             /*
465              * The device does not match the input file.
466              * Wait for any old output to drain, then attempt
467              * to reconfigure the audio device to match the
468              * input data.

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469     */
470     if (audio_drain(Audio_fd, FALSE) != AUDIO_SUCCESS) {
471         /* Flush any remaining audio */
472         (void) ioctl(Audio_fd, I_FLUSH, FLUSHW);

474         Error(stderr, MGET("%s: "), prog);
475         perror(MGET("AUDIO_DRAIN error"));
476         exit(1);
477     }

479     /* Flush any remaining audio */
480     (void) ioctl(Audio_fd, I_FLUSH, FLUSHW);

482     if (!reconfig()) {
483         errorStatus++;
484         goto closeinput;
485     }
486 }

489 /* try to do the mmaping - for regular files only ... */
490 err = fstat(ifd, &st);
491 if (err < 0) {
492     Error(stderr, MGET("%s: cannot stat "), prog);
493     perror(ifile);
494     exit(1);
495 }
496 regular = (S_ISREG(st.st_mode));

499 /* If regular file, map it.  Else, allocate a buffer */
500 mapaddr = 0;

502 /*
503  * This should compare to MAP_FAILED not -1, can't
504  * find MAP_FAILED
505  */
506 if (regular && ((mapaddr = mmap(0, st.st_size, PROT_READ,
507     MAP_SHARED, ifd, 0)) != MAP_FAILED)) {

509     (void) madvise(mapaddr, st.st_size, MADV_SEQUENTIAL);

511     /* Skip the file header and set the proper size */
512     cnt = lseek(ifd, 0, SEEK_CUR);
513     if (cnt < 0) {
514         perror("lseek");
515         exit(1);
516     }
517     inbuf = (unsigned char *) mapaddr + cnt;
518     len = cnt = st.st_size - cnt;
519 } else { /* Not a regular file, or map failed */

521     /* mark is so. */
522     mapaddr = 0;

524     /* Allocate buffer to hold 10 seconds of data */
525     cnt = BUFFER_LEN * File_hdr.sample_rate *
526         File_hdr.bytes_per_unit * File_hdr.channels;
527     if (bufsiz != cnt) {
528         if (buf != NULL) {
529             (void) free(buf);
530         }
531         buf = (unsigned char *) malloc(cnt);
532         if (buf == NULL) {
533             Error(stderr,
534                 MGET("%s: couldn't allocate %dK "

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535         "buf\n"), prog, bufsiz / 1000);
536         exit(1);
537     }
538     inbuf = buf;
539     bufsiz = cnt;
540 }
541 }

543 /* Set buffer sizes and pointers for conversion, if any */
544 switch (Decode) {
545     default:
546     case AUDIO_ENCODING_NONE:
547         insiz = bufsiz;
548         outbuf = (char *)buf;
549         break;
550     case AUDIO_ENCODING_G721:
551         insiz = ADPCM_SIZE / 2;
552         outbuf = (char *)adpcm_buf;
553         initmux(1, 2);
554         break;
555     case AUDIO_ENCODING_G723:
556         insiz = (ADPCM_SIZE * 3) / 8;
557         outbuf = (char *)adpcm_buf;
558         initmux(3, 8);
559         break;
560 }

562 /*
563  * 8-bit audio isn't a problem, however 16-bit audio is.
564  * If the file is an endian that is different from the machine
565  * then the bytes will need to be swapped.
566  *
567  * Note: Because the G.72X conversions produce 8bit output,
568  * they don't require a byte swap before display and so
569  * this scheme works just fine.  If a conversion is added
570  * that produces a 16 bit result and therefore requires
571  * byte swapping before output, then a mechanism
572  * for chaining the two conversions will have to be built.
573  *
574  * Note: The following if() could be simplified, but then
575  * it gets to be very hard to read.  So it's left as is.
576  */

578     if (File_hdr.bytes_per_unit == 2 &&
579         ((!NetEndian && file_type == FILE_AIFF) ||
580          (!NetEndian && file_type == FILE_AU) ||
581          (NetEndian && file_type == FILE_WAV))) {
582         swapBytes = TRUE;
583     } else {
584         swapBytes = FALSE;
585     }

587     if (swapBytes) {
588         /* Read in interal number of sample frames. */
589         frame = File_hdr.bytes_per_unit * File_hdr.channels;
590         insiz = (SWAP_SIZE / frame) * frame;
591         /* make the output buffer the swap buffer. */
592         outbuf = (char *)swap_buf;
593     }

595     /*
596     * At this point, we're all ready to copy the data.
597     */
598     if (mapaddr == 0) { /* Not mmapped, do it a buffer at a time. */
599         inbuf = buf;
600         frame = File_hdr.bytes_per_unit * File_hdr.channels;

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601     rem = 0;
602     while ((cnt = read(ifd, inbuf+rem, insiz-rem)) >= 0) {
603         /*
604          * We need to ensure only an integral number of
605          * samples is ever written to the audio device.
606          */
607         cnt = cnt + rem;
608         rem = cnt % frame;
609         cnt = cnt - rem;

611         /*
612          * If decoding adpcm, or swapping bytes do it
613          * now.
614          *
615          * We treat the swapping like a separate
616          * encoding here because the G.72X encodings
617          * decode to single byte output samples. If
618          * another encoding is added and it produces
619          * multi-byte output samples this will have to
620          * be changed.
621          */
622         if (Decode == AUDIO_ENCODING_G721) {
623             outsiz = 0;
624             demux(1, cnt / File_hdr.channels);
625             for (c = 0; c < File_hdr.channels; c++) {
626                 err = g721_decode(in_ch_data[c],
627                                 cnt / File_hdr.channels,
628                                 &File_hdr,
629                                 (void*)out_ch_data[c],
630                                 &tsiz,
631                                 &adpcm_state[c]);
632                 outsiz = outsiz + tsiz;
633                 if (err != AUDIO_SUCCESS) {
634                     Error(stderr, MGET(
635                         "%s: error decoding g721\n"),
636                         prog);
637                     errorStatus++;
638                     break;
639                 }
640             }
641             mux(outbuf);
642             cnt = outsiz;
643         } else if (Decode == AUDIO_ENCODING_G723) {
644             outsiz = 0;
645             demux(3, cnt / File_hdr.channels);
646             for (c = 0; c < File_hdr.channels; c++) {
647                 err = g723_decode(in_ch_data[c],
648                                 cnt / File_hdr.channels,
649                                 &File_hdr,
650                                 (void*)out_ch_data[c],
651                                 &tsiz,
652                                 &adpcm_state[c]);
653                 outsiz = outsiz + tsiz;
654                 if (err != AUDIO_SUCCESS) {
655                     Error(stderr, MGET(
656                         "%s: error decoding g723\n"),
657                         prog);
658                     errorStatus++;
659                     break;
660                 }
661             }
662             mux(outbuf);
663             cnt = outsiz;
664         } else if (swapBytes) {
665             swab((char *)inbuf, outbuf, cnt);
666         }

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668         /* If input EOF, write an eof marker */
669         err = write(Audio_fd, outbuf, cnt);

671         if (err < 0) {
672             perror("write");
673             errorStatus++;
674             break;
675         } else if (err != cnt) {
676             Error(stderr,
677                 MGET("%s: output error: "), prog);
678             perror("");
679             errorStatus++;
680             break;
681         }
682         if (cnt == 0) {
683             break;
684         }
685         /* Move remainder to the front of the buffer */
686         if (rem != 0) {
687             (void *)memcpy(inbuf, inbuf + cnt, rem);
688         }

690     }
691     if (cnt < 0) {
692         Error(stderr, MGET("%s: error reading "), prog);
693         perror("file");
694         errorStatus++;
695     }
696     } else {
697         /* We're mmaped */
698         if ((Decode != AUDIO_ENCODING_NONE) || swapBytes) {
699             /* Transform data if we have to. */
700             for (i = 0; i <= len; i += cnt) {
701                 cnt = insiz;
702                 if ((i + cnt) > len) {
703                     cnt = len - i;
704                 }
705                 if (Decode == AUDIO_ENCODING_G721) {
706                     outsiz = 0;
707                     demux(1, cnt / File_hdr.channels);
708                     for (c = 0; c < File_hdr.channels;
709                         c++) {
710                         err = g721_decode(
711                             in_ch_data[c],
712                             cnt / File_hdr.channels,
713                             &File_hdr,
714                             (void*)out_ch_data[c],
715                             &tsiz,
716                             &adpcm_state[c]);
717                         outsiz = outsiz + tsiz;
718                         if (err != AUDIO_SUCCESS) {
719                             Error(stderr, MGET(
720                                 "%s: error decoding "
721                                 "g721\n"), prog);
722                             errorStatus++;
723                             break;
724                         }
725                     }
726                 }
727                 mux(outbuf);
728             } else if
729             (Decode == AUDIO_ENCODING_G723) {
730                 outsiz = 0;
731                 demux(3,
732                     cnt / File_hdr.channels);
733                 for (c = 0;

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733         c < File_hdr.channels;
734         c++) {
735             err = g723_decode(
736                 in_ch_data[c],
737                 cnt /
738                 File_hdr.channels,
739                 &File_hdr,
740                 (void*)out_ch_data[c],
741                 &tsize,
742                 &adpcm_state[c]);
743             outsiz = outsiz + tsize;
744             if (err != AUDIO_SUCCESS) {
745                 Error(stderr, MGET(
746                     "%s: error "
747                     "decoding g723\n"),
748                     prog);
749                 errorStatus++;
750                 break;
751             }
752         }
753         mux(outbuf);
754     } else if (swapBytes) {
755         swab((char *)inbuf, outbuf,
756             cnt);
757         outsiz = cnt;
758     }
759     inbuf += cnt;

761     /* If input EOF, write an eof marker */
762     err = write(Audio_fd, (char *)outbuf,
763         outsiz);
764     if (err < 0) {
765         perror("write");
766         errorStatus++;
767     } else if (outsiz == 0) {
768         break;
769     }

771 } else {
772     /* write the whole thing at once! */
773     err = write(Audio_fd, inbuf, len);
774     if (err < 0) {
775         perror("write");
776         errorStatus++;
777     }
778     if (err != len) {
779         Error(stderr,
780             MGET("%s: output error: ", prog);
781             perror("");
782             errorStatus++;
783         }
784     err = write(Audio_fd, inbuf, 0);
785     if (err < 0) {
786         perror("write");
787         errorStatus++;
788     }
789 }

793 /* Free memory if decoding ADPCM */
794 switch (Decode) {
795 case AUDIO_ENCODING_G721:
796 case AUDIO_ENCODING_G723:
797     freemux();
798     break;

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799         default:
800             break;
801     }

803 closeinput:;
804     if (mapaddr != 0)
805         (void) munmap(mapaddr, st.st_size);
806     (void) close(ifd); /* close input file */
807     if (Errrdetect) {
808         cnt = 0;
809         (void) audio_set_play_error(Audio_fd,
810             (unsigned int *)&cnt);
811         audio_set_play_error(Audio_fd, (unsigned int *)&cnt);
812         if (cnt) {
813             Error(stderr,
814                 MGET("%s: output underflow in %s\n"),
815                 Ifile, prog);
816             errorStatus++;
817         }
818     }
819     nextfile:;
820     } while ((argc > 0) && (argc--, (Ifile = *argv++) != NULL));

821     /*
822     * Though drain is implicit on close(), it's performed here
823     * to ensure that the volume is reset after all output is complete.
824     */
825     (void) audio_drain(Audio_fd, FALSE);

827     /* Flush any remaining audio */
828     (void) ioctl(Audio_fd, I_FLUSH, FLUSHW);

830     if (Volume != INT_MAX)
831         (void) audio_set_play_gain(Audio_fd, &Savevol);
832     if ((Audio_ctlfd >= 0) && (audio_cmp_hdr(&Save_hdr, &Dev_hdr) != 0)) {
833         (void) audio_set_play_config(Audio_fd, &Save_hdr);
834     }
835     (void) close(Audio_fd); /* close output */
836     return (errorStatus);
837 }

unchanged_portion_omitted_

```

```

*****
20769 Thu Jan 17 14:27:12 2019
new/usr/src/cmd/audio/audiorecord/audiorecord.c
10101 audio tools need smatch fixes
*****
1 /*
2  * CDDL HEADER START
3  *
4  * The contents of this file are subject to the terms of the
5  * Common Development and Distribution License (the "License").
6  * You may not use this file except in compliance with the License.
7  *
8  * You can obtain a copy of the license at usr/src/OPENSOLARIS.LICENSE
9  * or http://www.opensolaris.org/os/licensing.
10 * See the License for the specific language governing permissions
11 * and limitations under the License.
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13 * When distributing Covered Code, include this CDDL HEADER in each
14 * file and include the License file at usr/src/OPENSOLARIS.LICENSE.
15 * If applicable, add the following below this CDDL HEADER, with the
16 * fields enclosed by brackets "[]" replaced with your own identifying
17 * information: Portions Copyright [yyyy] [name of copyright owner]
18 *
19 * CDDL HEADER END
20 */
21 /*
22 * Copyright 2009 Sun Microsystems, Inc. All rights reserved.
23 * Use is subject to license terms.
24 */
26 /*
27 * Copyright (c) 2018, Joyent, Inc.
28 */
30 /* Command-line audio record utility */
32 #include <stdio.h>
33 #include <libgen.h>
34 #include <errno.h>
35 #include <ctype.h>
36 #include <math.h>
37 #include <stdlib.h>
38 #include <unistd.h>
39 #include <string.h>
40 #include <strings.h>
41 #include <locale.h>
42 #include <fcntl.h>
43 #include <signal.h>
44 #include <limits.h> /* All occurrences of INT_MAX used to be ~0 (by MCA) */
45 #include <sys/types.h>
46 #include <sys/file.h>
47 #include <sys/stat.h>
48 #include <sys/param.h>
49 #include <stropts.h>
50 #include <poll.h>
51 #include <sys/ioctl.h>
52 #include <netinet/in.h>
54 #include <libaudio.h>
55 #include <audio_device.h>
57 #define irint(d) ((int)d)
59 /* localization stuff */
60 #define MGET(s) (char *)gettext(s)

```

```

62 #if !defined(TEXT_DOMAIN) /* Should be defined by cc -D */
63 #define TEXT_DOMAIN "SYS_TEST" /* Use this only if it weren't */
64 #endif
66 #define Error (void) fprintf
68 /* Local variables */
69 static char *prog;
70 static char prog_opts[] = "aft:v:d:i:e:s:c:T?"; /* getopt() flags */
71 static char *Stdout;
73 /* XXX - the input buffer size should depend on sample_rate */
74 #define AUDIO_BUFSIZ (1024 * 64)
75 static unsigned char buf[AUDIO_BUFSIZ];
76 static char swapBuf[AUDIO_BUFSIZ]; /* for byte swapping */
78 #define MAX_GAIN (100) /* maximum gain */
81 static char *Info = NULL; /* pointer to info data */
82 static unsigned Ilen = 0; /* length of info data */
83 static unsigned Volume = INT_MAX; /* record volume */
84 static double Savevol; /* saved volume */
85 static unsigned Sample_rate = 0;
86 static unsigned Channels = 0;
87 static unsigned Precision = 0; /* based on encoding */
88 static unsigned Encoding = 0;
90 static int NetEndian = TRUE; /* endian nature of the machines */
92 static int Append = FALSE; /* append to output file */
93 static int Force = FALSE; /* ignore rate differences on append */
94 static double Time = -1.; /* recording time */
95 static unsigned Limit = AUDIO_UNKNOWN_SIZE; /* recording limit */
96 static char *Audio_dev = "/dev/audio";
98 static int Audio_fd = -1;
99 /* file descriptor for audio device */
100 static Audio_hdr Dev_hdr; /* audio header for device */
101 static Audio_hdr Save_hdr; /* saved audio device header */
102 static char *Ofile; /* current filename */
103 static int File_type = FILE_AU; /* audio file type */
104 static int File_type_set = FALSE; /* file type specified as arg */
105 static Audio_hdr File_hdr; /* audio header for file */
106 static int Cleanup = FALSE; /* SIGINT sets this flag */
107 static unsigned Size = 0; /* Size of output file */
108 static unsigned Oldsize = 0;
109 /* Size of input file, if append */
111 /* Global variables */
112 extern int getopt();
113 extern int optind;
114 extern char *optarg;
116 /* Local Functions */
117 static void usage(void);
118 static void sigint(int sig);
119 static int parse_unsigned(char *str, unsigned *dst, char *flag);
120 static int parse_sample_rate(char *s, unsigned *rate);
123 static void
124 usage(void)
125 {
126     Error(stderr, MGET("Record an audio file -- usage:\n")
127         "\t%s [-af] [-v vol]\n");

```

```

128     "\t%. *s [-c channels] [-s rate] [-e encoding]\n"
129     "\t%. *s [-t time] [-i info] [-d dev] [-T au|wav|aif[f]] [file]\n"
130     "where:\n"
131     "\t-a\tAppend to output file\n"
132     "\t-f\tIgnore sample rate differences on append\n"
133     "\t-v\tSet record volume (0 - %d)\n"
134     "\t-c\tSpecify number of channels to record\n"
135     "\t-s\tSpecify rate in samples per second\n"
136     "\t-e\tSpecify encoding (ulaw | alaw | [u]linear | linear8 )\n"
137     "\t-t\tSpecify record time (hh:mm:ss.dd)\n"
138     "\t-i\tSpecify a file header information string\n"
139     "\t-d\tSpecify audio device (default: /dev/audio)\n"
140     "\t-T\tSpecify the audio file type (default: au)\n"
141     "\tfile\tRecord to named file\n"
142     "\t\tIf no file specified, write to stdout\n"
143     "\t\tDefault audio encoding is ulaw, 8khz, mono\n"
144     "\t\tIf -t is not specified, record until ^C\n",
145     prog,
146     strlen(prog), " ",
147     strlen(prog), " ",
148     MAX_GAIN);
149     exit(1);
150 }

```

unchanged portion omitted

```

741 /*
742  * set the sample rate. assume anything is ok. check later on to make sure
743  * the sample rate is valid.
744  */
745 static int
746 parse_sample_rate(char *s, unsigned *rate)
747 {
748     char      *cp;
749     double    drate;
750
751     /*
752      * check if it's "cd" or "dat" or "voice". these also set
753      * the precision and encoding, etc.
754      */
755     if (strcasecmp(s, "dat") == 0) {
756         drate = 48000.0;
757     } else if (strcasecmp(s, "cd") == 0) {
758         drate = 44100.0;
759     } else if (strcasecmp(s, "voice") == 0) {
760         drate = 8000.0;
761     } else {
762         /* just do an atof */
763         drate = atof(s);
764
765         /*
766          * if the first non-digit is a "k" multiply by 1000,
767          * if it's an "h", leave it alone. anything else,
768          * return an error.
769          */
770
771         /*
772          * XXX bug alert: could have multiple "." in string
773          * and mess things up.
774          */
775         for (cp = s; *cp && (isdigit(*cp) || (*cp == '.')); cp++)
776             /* NOP */;
777         if (*cp != NULL) {
778             if ((*cp == 'k') || (*cp == 'K')) {
779                 drate *= 1000.0;
780             } else if ((*cp != 'h') && (*cp != 'H')) {
781             } else if ((*cp != 'h') || (*cp != 'H')) {

```

```

781     /* bogus! */
782     Error(stderr,
783           MGET("invalid sample rate: %s\n"), s);
784     return (1);
785     }
786     }
787
788     }
789
790     *rate = irint(drate);
791     return (0);
792 }

```

unchanged portion omitted