APPLICATION NOTE

Guide-line for CASP software design, tone /volume part

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Abstract

The audio part of the Car radio Analog Signal Processor (CASP) provides new features like Analog Step Interpolation (ASI), Audio Blend Control (ABC), and fully programmable loudness. Introducing and use of additional system parameters into the software makes the system very flexible in application for different radio concepts. Background and requirements for software control are described.

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Author(s):

U. Schillhof
Systems Laboratory Hamburg, Germany

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Summary

The internal functions of the audio part of the Car radio Analog Signal Processor (CASP) as well as the background and requirements for software control are described. The volume, balance, and fader control as the most complex parts are explained in detail with a proposal for software implementation.

CONTENTS

1. Introduction

The Car radio Analog Signal Processor TEA688x (CASP) is a monolithic bipolar Integrated Circuit (IC) providing the stereo decoder function and ignition noise blanking facility combined with the source selector and tone/volume control for AM/FM car radio applications. Control of all settings and alignments as well as read of the detector outputs are performed via the I^2C-bus.

This report was written to support radio setmakers as well as internal colleagues in software specification and design for use together with the CASP. The focus of this report is set on the functions of the tone/volume part of the IC.

2. Tone/volume concept of CASP

The tone/volume or audio control part of the CASP has a number of new features and more flexibility in software control compared with today's available circuits for audio control. The consequence is the need of a more sophisticated software. For this, the know how about the used audio control concept is necessary.

Fig. 1 shows a very simplified blockdiagram of the audio control part. The source selector selects the internal source (AM/FM) or one out of three external stereo and two mono sources respectively. The control information has to be send with the byte SSEL. The SSEL byte controls also the speed of the Analog Step Interpolation (ASI) and Audio Blend Control (ABC) with two bits. ASI and ABC is implemented into the functional blocks for volume, loudness, balance, fader and bass control.

The first blocks of the main control part are the loudness control blocks for left and right. In "on" mode these blocks allow a frequency response which depends on the volume setting according to the contour of ear. The effect of bass (and treble) boost depends on the attenuation setting of these blocks and is basically determined by the external circuitry. In "off" mode the loudness blocks are linear attenuators. The register LOUD defines the attenuation setting and mode of the loudness blocks.

The loudness blocks are followed by the volume_1 blocks. The placing of these blocks behind the loudness blocks gives optimal signal-to-noise behaviour when the gain is < 0 dB (normal listening condition, Dolby and CD specifications). The volume_1 blocks provide frequency linear gain settings for sources with low levels as well as attenuation for high input levels to avoid internal signal clipping effects when the tone controls are in boost positions. The volume_1 controls are always used in the upper portion of the volume control range in combination with the loudness attenuators. The settings are defined by the register VOLU1.

The next functions in the signal path are the tone control blocks with active treble and bass control stages. The treble control in front of the bass control gives a better signal-to-noise behaviour. Treble and bass control have the separate control registers TRBL and BASS.

The last signal processing is done in the volume_2 blocks. Two of them are used for the left channel and two for the right. These four outputs which can drive via power amplifiers two groups of loudspeakers (front and rear). The volume_2 blocks provide an attenuation range from 0 dB to 68 dB and a mute setting. Each volume_2 block can be controlled independently from the other. Therefore the four control registers VOL2_LF, VOL2_LR, VOL2_RF, and VOL2_RR are provided.

The volume_2 stages for the front outputs additionally have a signal adder function, so called chime adder. With this the chime input signal can be summed with the left front and/or with the right front audio, or be turned off.

3. I²C-bus control of CASP

The standard I^2C-bus is used for communication with the external controller [1], [2]. The CASP is always a slave on the bus and has write as well as read functions. Fig. 2 shows the used structure. The 6 higher bits of the CASP chip address (module address) are fixed to 001100. The last significant bit can be set via the external input pin ADR (ground = 0, V_{CC} = 1).

3.1 Write mode

In the write mode a subaddress byte has to be transmitted. The subaddress byte includes beside the data subaddress (four least significant bits SAD0 .... SAD3) also the two control bits AIOF and BOUT.

The data subaddress defines the location/function of the data byte(s) succeeding to the subaddress. The number of further data bytes is optional. Fig. 3 shows the subaddresses and data description of the CASP tone/volume part.

With the bit AIOF the feature of subaddress autoincrement can be switched on or off (AIOF = 0 --> autoincrement on, AIOF = 1 --> autoincrement off). If autoincrement is on, the internal subaddress pointer is automatically incremented after every reception of a data byte, so that a succeeding data byte is written into the next address location. This mode is advantageous when several data bytes have to be updated.

If the autoincrement is off, the subaddress remains on the value defined by the bits SAD0 .... SAD3 independently from the number of following data bytes. A succeeding data byte overwrites the previous value in the same location. This mode is advantageous when a function should be varied over a range e.g. for an alignment or test procedure.

The CASP has additional to the main I^2C-bus terminals SCL and SDA two further terminals QSCL and QSDA. These are assigned for the frontend/i.f. part of the receiver (NICE). With the bit BOUT = 1 the terminals QSCL and QSDA are enabled. In this case the CASP is transparent between SCL/QSCL and SDA/QSDA respectively and the frontend/i.f. part is connected with the I^2C-bus.

With the bit BOUT = 0 the terminals QSCL and QSDA are disabled. In this case the frontend/i.f. part is disconnected from the I^2C-bus.

This has advantages with view to possible cross talk effects between the bus and the very sensitive antenna or I.F. inputs of the receiver.

3.2 Read mode

The read mode works without subaddress control. After addressing the CASP with a first byte and the read/write-not bit = 1 the two available data bytes with information about the stereo decoder status, level and signal quality (ultrasonic noise and wide band AM noise) can be read out directly. This saves time and reduces bus loading for frequently polling these information.

4. Software implementation

As previously mentioned, the report concentrates on the functions of the tone/volume part. The stereo decoder and other weak signal processing parts are not described in this report.

Fig. 4 shows the structure of the proposed software control for the tone volume part. The event which requires a partly or complete update of the CASP may be a user control action like change of volume or an internal radio control demand e.g. a traffic message requires the change from the source cassette to radio and also a change in volume, gain adjust, and loudness start.

The user controlled parameters like volume, bass, treble, and others are largely independent from the used audio control IC. Internal radio control parameters like gain adjustment and loudness start are generally also needed but the available possibilities and parameters are influenced by the features of the used IC. The CASP provides a very high flexibility as it is shown in fig. 4.

User control and internal radio control are not independent from each other. The user control may have an influence on the internal radio control parameters and opposite. E.g. inserting a cassette causes a source selector switching and may also control a change in gain adjustment if the cassette player has a different output level than other sources. Or the difference in signal dynamic of the selectable sources may require a change of the parameter start of loudness.

If an update of the CASP is required, then easily a subroutine or function can be called for calculation of tone/volume data for CASP. This subroutine reads the data registers provided by the user control as well as internal radio control as input. Output are the data for the CASP specific registers. It can be written directly into those output buffers used by the I^2C-bus. Afterwards the I^2C-bus subroutine can be called for data transmission to the CASP (fig. 4).

The following chapters explain more in details about the different control parameters shown in fig. 4 and their processing by the calculation subroutine / function. Particular processing of volume, balance, and fader input is very complex and therefore explained in detail with a proposal for implementation in the appendix.

4.1 Analog Step Interpolation and Audio Blend Control

Analog Step Interpolation and Audio Blend Control are implemented for the blocks volume_1, loudness, all four volume_2 blocks , and for the bass control.

4.1.1 Analog Step Interpolation

The Analog Step Interpolation (ASI) smoothes out the transitions from any step to the next when audio control is performed. This minimizes step noise caused by modulation and offset effects. The reduction of step noise is as more effective as longer the transition time.

The transition is controlled by an internal start/stop saw-tooth generator build by internal current sources in combination with the external capacitor ASICAP [3]. The internal current sources are software programmable over a range of 1:24 by the bits ASI0 and ASI1 of the register SSEL (subaddress = 4, see fig. 3). The following period values can be programmed with ASICAP = 22nF:

ASI1	ASI0	transition time per step
0	0	0.83 ms
0	1	3.33 ms
1	0	8.33 ms
1	1	20 ms

A demonstration software for PCs should have option buttons, also referred as radio buttons, for user controllable ASI/ABC speed. The initial setting should be 11.

4.1.2 Audio Blend Control

In combination with ASI the Audio Blend Control (ABC) feature offers a quasi analog volume/sound variation with simplified software control.

The internal basic elements of the ABC are an up/down counter, a latch, and a comparator for each control block with ABC. The up/down counter represents the current setting of the corresponding control stage and has the same value as the latch in the steady-state condition. When a change should be performed, the latch receives control data as the new target setting via the I^2C-bus. The comparator takes notice of every deviation between the up/down counter (current setting) and the latch (target). In case of a difference the comparator starts the saw-tooth generator of the ASI system if it is in stop condition and sends a count signal as well as a sign signal to the up/down counter. This automatically effects a stepwise approaching to the new target setting.

One cycle of the saw-tooth generator is used for initialization and synchronization of the counter control. From this it follows that after acknowledge of the concerned data byte the first step is performed during the second cycle of the saw-tooth generator.

The ABC speed (repetition time of steps) depends on the ASI transition time respectively the period of the ASI saw-tooth generator. With the values given in 4.1.1 it can be calculated, that the time for a 40 dB change in volume (41 saw-tooth cycles) can be varied between min. 34 ms (ASI1/0 = 00) and max. 820 ms (ASI1/0 = 11).

The internal logic for ABC and ASI is developed in that way, that there is no restriction in I^2C-bus control. All ABC controlled blocks can temporally work parallel until their individual target value is reached.

Further it is not necessary to wait with a next I^2C-bus control command (new target value) until ABC has reached the previous target value. The ABC immediately follows the last received target value. That is e.g. important for rotary knob volume control to avoid rubber band effects when a person turns up and down in a fast sequence.

4.2 Mute control

The CASP has basically three different mute possibilities. Mute control has to be done via I^2C-bus in all cases.

4.2.1 Multiplex mute

The use of multiplex mute is intended for fast RDS update (Radio Data System) or test of alternative frequencies. The mute stage is a part of the stereo decoder / noise blanker circuitry. The soft mute/demute transition period is defined by the time constant of the ultra sonic noise average detector.

Multiplex mute can be activated by setting and deactivated by resetting the bit MMUT (subaddress 0, see [3]). Multiplex mute can only be used for tuning operations, because it influences only the radio source (FM and AM mono).

4.2.2 Audio mute

The audio mute function mutes all output stages of the CASP (volume_2). It is controllable via the bit AMUT (bit 7, subaddress 8 (VOLU1), see fig. 3). Mute is activated when AMUT = 1 and deactivated when AMUT = 0. The other settings for tone and volume are not influenced.

The audio mute works with ASI but without ABC. That means mute becomes active within a time of one ASI step after setting the AMUT bit via I^2C-bus. The mute/demute transition period is defined by the ASI system and programmable via the bits ASI1 and AS0 from 0.83 ms to 20 ms (see 4.1.1). The mute transition period can be made different from the demute transition period by an appropriate control of the ASI1/0 bits.

Audio mute is automatically active after power on. The chime adders are not affected by the AMUT bit. They can only be switched on or off by the bits CHML and CHMR (see 4.7)

4.2.3 Soft mute

The ABC and ASI features together allow a very flexible soft mute option with the volume_2 control blocks. For mute the following data have to be send:

subaddress 9, VOLU_2LF	=	CHML	0 0 0 0 0 1 1
subaddress A, VOLU_2RF	=	CHMR	0 0 0 0 0 1 1
subaddress B, VOLU_2LR	=	0	0 0 0 0 0 1 1
subaddress C, VOLU_2RR	=	0	0 0 0 0 0 1 1

CHML and CHMR are the chime adder control bits. The chime adder is on, if the corresponding bit is 1 and off if the bit is 0.

After receiving these data, all four volume_2 stages will automatically increase their attenuation step by step until the mute position is reached. The total time for the mute slope depends upon the cycle time of the steps (defined by the previous setting of the ASI1/ASI0 bits) and on the initial position of volume_2. The max. time is needed when the initial position is 0 dB (max gain). This time can be varied by the ASI1/ASI0 bits:

ASI1	ASI0	max. soft mute slope time
0	0	51 ms
0	1	203 ms
1	0	508 ms
1	1	1220 ms

Optionally it is also possible to change the ASI/ABC speed during the mute slope. So the mute can be started with a very soft slope and accelerated when a certain attenuation is reached. As an example the soft mute can be started with a ASI period of 8.3 ms and after 200 ms accelerated by setting the ASI period to 0.83 ms. Then the first 30 dB are attenuated very softly and the rest of attenuation range is done in further 26 ms.

For demute the four volume_2 values for the current volume/balance/fader settings have to be transmitted via I^2C-bus. These settings may be different from those before mute. That is the case if soft mute is used for changing the source. The new source may have a different setting in the parameter gain adjust. This may also influence the volume_2 settings. For this reason it is recommended to use the later described calculation subroutine instead of saving the old volume_2 settings. That has the advantage for microcontroller software that no extra registers are needed.

The demuting slope can also be made different from that for mute. Also changing the slope during demute is possible. But for demute it is senseful to start with a fast slope and reduce the speed after e.g. 25 .... 30 ms.

4.3 Source Selector control

Generally a change of source selector setting in a car radio is only indirectly possible or happens automatically, e.g. if the current mode is FM reception and the user inserts a cassette. For PC demo software option buttons should be used for directly selection of the desired source. The default setting should be source A.

Data preparation for source selector control is a simple setting of the source selector bits within the control byte regarding the truth table given in the data sheet [3]. Additionally the bits ASI0 and ASI1 for the ASI/ABC speed have to be inserted (see fig. 3). For the IC update the source selector data have to be transmitted with the subaddress 4. (register SSEL, fig. 1 and 4).

But a change in source selector setting needs a sequence of different I^2C-bus commands. It should be combined with a preceded audio mute (see 4.2.2) or soft mute (see 4.2.3) and succeeded to a short delay and finally demute as described. The delay can be needed for source settling (settling time of the synthesizer when switched over to radio source) or DC voltage settling at the input of the main control part (offset voltages at the output of the source selector caused by DC coupled sources).

With request for a new source the old source setting information must not be overwritten. It has to be stored because this information is needed, when the ASI/ABC speed should be changed for the mute operation before the real source switching is done.

4.4 Bass and treble control

Bass and treble have different control ranges (bass +/-18 dB and treble +/-14 dB) and also different truth tables (see [3]). For calculating the control data for CASP (BASS and TRBL) with the user controllable data as input, it has to be noticed that the two different successive CASP data 10000 / 10001 for bass and 0111 / 1000 for treble result both into linear settings.

The bass control provides two extra features. One is the ASI/ABC control. The implementation of ASI/ABC allows quasi analog control also for bass. That is a very useful feature when automatic control is used for anticlipping systems or drive speed dependent volume and sound control.

The second feature is the possibility of changing the filter characteristic. It can be set to unsymmetrical (shelving filter) or symmetrical (bandpass filter characteristic). Two bits BSYC and BSYB (subaddress 5, see fig. 3 and [3]) are provided to set the characteristic for boost and cut independently from each other:

BSYB	BSYC	bass control filter characteristic
0	0	boost and cut shelving characteristic
0	1	boost shelving and cut symmetrical
1	0	boost symmetrical and cut shelving
1	1	boost and cut symmetrical characteristic

The internal circuit is optimized for BSYB = 1 and BSYC = 0 (+/-18 dB, 2 dB steps, external T-filter). That provides in boost situation no or only small boost of the subsonic frequencies to avoid speaker overdriving with inaudible frequencies. In cut mode not only the audible bass frequencies but also the subsonic frequencies are attenuated

Normally these bits are fixed in a radio. But if the bits should be changed it is recommended to do this in the linear position of bass control or in the not active control range i.e. change of the bit BSCB during cut and BSYC during boost. Otherwise a little click may occur.

For PC control software both bits BSYB and BSYC should be controllable with e.g. a check box for nonexclusive choices. The default setting should be BSYB = 1 and BSYC = 0.

4.5 Loudness

If loudness is implemented the frequency response of a car radio's audio part should change according to the contour of ear when the user varies the volume setting . For this requirement the CASP has a separate loudness control block which allows a very flexible programming of this effect. The loudness block comprises in principle two attenuators with a range of 20 dB in steps of 1 dB. One attenuator is frequency independent (linear) and the second has increasing bass and optional also treble boost with increasing attenuation. The frequency dependency of the second attenuator is defined by external components.

The attenuation for both attenuators is controlled by the five bits LSN0 .... LSN4 (loudness control register LOUD, subaddress 7, see fig. 3), but only one of these attenuators is active switched on at the moment. Which one is defined by the msb bit LOFF of the same control register.

The linear attenuator is activated when LOFF = 1 (loudness off) and the frequency dependent attenuator is activated when LOFF = 0 (loudness on). With other words the loudness switch LOFF does not influence the attenuation of the loudness block referred to the mid frequency range. It only determines if the boost of low and high frequencies, which depends on the attenuation, is active or not.

So the loudness attenuators can be used in combination with the other volume attenuators (volume_1 and volume_2) for volume control. The partition of the whole volume control range into sections for volume_1, loudness, and volume_2 is fully flexible and has to be defined by the control software. The next paragraph (4.6) describes this in detail.

4.6 Volume, balance, and fader control

The user controlled input variable volume has to be processed by the software to get the control data for the CASP volume_1, loudness, and volume_2 blocks as shown in fig. 4. Balance and fader settings, also user controlled, cause an additionally attenuation in the volume_2 blocks for left or right (balance) and front or rear (fader).

Beside the user controlled input variables also some internal control information is necessary to define ranges and/or match the characteristic to certain requirements. The CASP has regarding ranges and characteristic a lot of flexibility. The definition of internally used parameters (or variables) make these complex connections easy to use.

4.6.1 Internally used parameters

The diagrams fig. 5 to fig. 9 show the gain/attenuation contribution of the internal CASP blocks when volume control is performed. Further the influences of a number of additionally parameters are shown.

4.6.2 Gain adjustment

The change of sources should not cause a jump in volume. Therefore a gain adjustment is necessary when the source selector switches over to a source with a different average level. This can be done with the parameter "gain adjust" (GAIN_ADJ). The parameter gain adjust defines the total gain of the CASP audio section when the volume is in max. position.

In fig. 5 the influence of gain adjust is shown. The left graph represents a standard application as a reference. Three vectors are shown for the attenuation (negative gain values) of loudness (left), gain/attenuation of volume_1, and attenuation of volume_2. For max. volume setting all vectors have the length of 0. The dot represents this situation (max. volume) as "volume decrement start". The arrows show the contribution of the different audio control blocks when the volume is decreased. Coming from max. volume, first the gain of volume_1 is reduced. After a span of 10 dB the volume_1 setting is frozen and additional attenuation is now performed by the loudness block. The loudness block has a range from 0 dB to -20 dB. After the loudness has reached -20 dB further attenuation is realized by use of volume_1 again. After volume_1 has reached the value -18 dB additional attenuation is done by the volume_2 attenuators. When the volume_2 attenuators have reached an attenuation of 48 dB, the total attenuation is 86 dB (volume_1 = -18 dB, loudness = -20 dB, volume_2 = -48 dB). The last volume step (-87 dB) activates mute because the total volume control range is limited to -87 dB for this application proposal.

The middle and the right graph show the changes when the parameter gain adjust is varied. The middle graph corresponds to the value GAIN_ADJ = +10 dB and the right to GAIN_ADJ = +20 dB. The gain for the max. volume position is now +10 dB (middle, GAIN_ADJ = +10 dB) or +20 dB (right, GAIN_ADJ = +20 dB). The parameter gain adjust can be varied in steps of 1 dB. For PC demo software it is recommended to limit the gain adjust range between 0 dB and +20 dB.

It can be seen, that in all examples loudness start remains 10 dB below max. volume, the minimum volume_1 setting remains on -18 dB and also the total volume control range remains -86 dB plus the mute step. The change is in the use of volume_1 with a higher max. gain and bigger control range. The bigger control range of volume_1 is compensated by a smaller control range of volume_2, so that the total control range remains the same.

4.6.3 Start of loudness

As previously explained (2. and 4.5) the loudness block provides a gradual transition from a linear frequency response to a response with bass (and optional treble) boost when the volume settings are decreased. Fig. 10 shows the proposed standard application with bass and treble boost. The frequency response is plotted for every volume step over the range from 0 dB to -86 dB. It can be seen that the frequency response is linear from 0 to -10 dB. The frequency response changes from linear to a curve with bass and treble boost in the range from -10 dB to -30 dB. This is the range of volume control which is performed by the loudness block. Below -30 dB all curves are parallel. That means there is no further influence on the frequency response.

The position of the loudness control range within the volume control range can be defined by the parameter "loudness start" (LDN_STRT). It can be varied in steps of 1 dB. Fig. 6 shows the influence of this parameter. The left graph is the standard application again as described in previous paragraph (4.6.2). The loudness start is here LDN_STRT = -10 dB. The middle graph shows an earlier start namely -5 dB. In the right graph the value for loudness start (-20 dB) is below the minimum volume_1 value (-18 dB). Therefore the whole 18 dB attenuation of volume_1 and in addition 2 dB of volume_2 have to be used before the loudness attenuation becomes active.

For PC demo software it is recommended to limit the loudness start range between 0 dB and -20 dB with the default value of -10 dB.

4.6.4 Minimum volume_1 value

Optimal signal-to-noise behaviour is reached when the gain is located in the first stages of an amplifier and the attenuation in the last stages. That is the reason for using two volume control blocks in the signal path of CASP namely volume_1 and volume_2. The splitting of attenuation is variable and can be varied by the parameter "minimum volume_1 setting" (MIN_VOL1). With respect to signal-to-noise ratio the volume_1 part should have as less attenuation as possible. But the minimum required attenuation is given by the gain of stages behind volume_1. Stages behind volume_1 which comprise gain are the tone control stages. The bass control has a maximum boost of 18 dB. Therefore the recommended value for MIN_VOL1 is -18 dB. The contribution of attenuation in the loudness block is on one hand very low (approx. 4 dB at 60 Hz) and on the other hand it doesn't help when the start of loudness is below MIN_VOL1.

The max. available attenuation of the volume_1 block is 36 dB. The reserve is built-in for special applications e.g. a bass boost of much more than 18 dB or boosters with high gain behind CASP.

Fig. 7 shows the influence of the parameter minimum volume_1 setting. As reference the left graph is the standard application again. In the middle the MIN_VOL1 is set to -15 dB. The lower attenuation of volume_1 is compensated by a higher attenuation in the volume_2 blocks (-51 dB instead of -48 dB). Thus volume control range and the other parameters are not changed. If the attenuation in the volume_1 block is higher (right graph) then less attenuation have to be used in the volume_2 blocks before the mute step is performed.

For PC demo software it is recommended to provide for MIN_VOL1 a range between -15 dB and -36 dB with an initial value of -18 dB.

4.6.5 Volume control range

The requirement for volume control is to reach full power with an average input signal on one hand and on the other hand to be able to attenuate the volume down to the limit of audibility. This requirement can be fulfilled with a control range of approx. 80 dB if 25 W power amplifiers are used. For the case of useing of additional booster amplifiers, additional approx. 6 dB are required. That results in a recommended value of 86 dB (plus the additional mute step) for the volume control range.

The volume control range should be neither too big nor to small to avoid a dead zone or a too early mute step. By use of the CASP setmakers have the possibility to vary the volume control range and match it to different applications e.g. if it is sensed by the radio control that a booster is connected the volume control range may be enlarged.

Fig. 8 shows the influence of the parameter "volume control range" (VOLU_RNG). It can be seen, that only the max. attenuation of the volume_2 parts is varied before the mute step limits the volume control.

For PC demo software it is recommended to provide for VOLU_RNG a range between -61 dB and -91 dB (61 dB means 60 dB continuous control range plus the mute step). The initial value should be -87 dB.

4.6.6 Minimal output attenuation

The maximal output voltage of CASP is 2 V (RMS). This level is used for booster outputs. For the internal output power amplifiers a smaller level is necessary. Most of the used power amplifiers need an input voltage of 0.5 V for full power. This conflict can be solved by using voltage dividers in front of the power amplifiers, if both - 2 V booster outputs and internal power amplifiers - are required in the same radio.

These voltage dividers improve the signal-to-noise ratio of the internal amplifiers nearly by the attenuation factor of the divider. A similar improvement can be reached by using a residual attenuation in the volume_2 attenuators.

This can be done by the parameter "min. output attenuation" (OUTP_ATT). But there are two restrictions to be mentioned. The max. available gain and the max. output voltage are reduced by the attenuation factor.

The fig. 9 shows how the OUTP_ATT parameter works. For max. volume setting the volume_2 attenuators have not more the attenuation 0 dB when OUTP_ATT > 0. This residual attenuation is compensated by a higher gain of the volume_1 part. So the total gain as well as the volume control range are not affected by this parameter as long as the volume_1 gain is not needed for the parameter gain adjust.

For PC demo software it is recommended to provide for OUTP_ATT a range between 0 dB and 12 dB. The initial value should be 0 dB

4.6.7 Balance and fader control

Balance and fader control is performed only by the four volume_2 attenuators. It is done by additional attenuation of the opposite channels. That means that the left channels (VOL2_LR and VOL2_LF, see fig. 1 and 3) are attenuated when the balance is turned to right respective the front channels (VOL2_RF and VOL2_LF) are attenuated when the fader is turned to rear.

The subroutine for CASP data calculation must summarise the attenuation contributions of volume, balance, and fader for each of the four volume_2 attenuators. In case the total attenuation setting is < 000100 (see truth table for fader decoder [3]) the value has to be set to 000011 (mute).

The control range for balance and fader is usually 30 dB with an additional mute step for the corresponding pair of channels. The control range in a radio is always fixed. But for a PC control software it is recommended to allow a variation of the balance and fader range between 21 dB and 41 dB (20 dB resp. 40 dB with an additional mute step). The initial setting for the balance and fader range should be 31 dB. The balance and fader setting itself should be the center position.

4.6.8 Proposal for software implementation

The previous paragraphs show that the concept of volume control with CASP is very flexible to use, especially with the additionally defined parameters. The consequence is a little more complex software which processes all the input parameters and calculates the data needed for I^2C-bus control of CASP.

For the implementation of volume, balance, and fader processing a proposal is given in the appendix. It is a basic like pseudo notation. Fig. 11 shows the input and output data for this routine as a proposal for use in a PC demo software.

The upper field contains the user controlled input data for volume balance and fader. The field below the user control input provides the input of the normally internal fixed or internal controlled parameters.

The third field displays the calculated values for the corresponding six CASP T/V registers. The single switching bits LOFF, AMUT, CHML, and CHMR (see fig. 3) are not yet implemented into this proposal. The register data are represented as decimal values. Additionally also the corresponding dB settings are displayed.

For a better overview an additional display field is provided with the values of the total gain calculated from the input of the CASP audio part to the four outputs.

A demo software for PCs should have in addition to the other controls like ASI/ABC, source selector tone control etc., all this controls and displays.

4.7 Chime adder

Each of the CASP output stages for front left and front right comprise a signal adder function, called chime adder. A signal which is applied to the chime/diagnostic input can be added to the output signals. The added signal is not influenced by volume or tone control. Each chime adder can be switched on or off by the corresponding bits CHML (chime adder front left) and CHMR (chime adder front right), see fig. 3.

4.8 Initialization of the tone/volume part

The initial condition after power on is audio mute (AMUT = 1) and the bus terminals QSCL and QSDA are disabled (BOUT = 0). The current setting of volume_1 is -36 dB and mute for the four volume_2 attenuators.

For proper working of CASP a complete update of all control registers to the desired settings is necessary. This can be done with the autoincrement feature for the subaddress. The status after power on is preserved until the data with subaddress 8 (VOUL1, volume_1 data and AMUT bit) have been written. The ABC system starts immediately activity after receiving the data byte with this subaddress 8. But it is not necessary to interrupt the I^2C-bus update protocol. It can be finished by writing the last four registers (volume_2).

Now the ABC control moves all settings of the blocks with ASI/ABC to the target values defined by the I^2C-bus update protocol. If the AMUT bit was 1 (audio mute active) then this process is inaudible, because the CASP remains muted. If this bit is cleared later (audio mute inactive) then the circuit is demuted with one ASI step as described in 4.2.2.

If the AMUT bit was 0 (audio mute inactive) then this process is audible. The volume is softly increased like quasi analog as described in 4.2.3. It starts from the mute position and moves to the desired value given by the I^2C-bus update protocol as described above.

5. References

[1] The I^2C-bus specification, Philips Components 9398 358 10011

[2] The I^2C-bus specification, "The I^2C-bus and how to use it" Philips Semiconductors 9398 393 40011

[3] Data sheet TEA688x, Philips Semiconductors

APPENDIX 1

```
Sub CALC_REG ()

'Start "calculation LOUD register"
'---------------------------------

'Start "calculation Loudness attenuation"
'========================================
                                        '? loudness attenuation necessary ?
If VOLUME < LDN_STRT Then               'loudness attenuation has to be performed
                                        '? max. loudness attenuation ?
  If VOLUME < (LDN_STRT - 20) Then
    LOUD = 11                           'set max. loudness attenuation
  Else
    LOUD = VOLUME - LDN_STRT + 31       'volume depending loudness attenuation
  End If                                '(31 = correction for 0 dB attenuation)
Else
  LOUD = 31                             'set no loudness attenuation
End If                                  'End "calculation Loudness attenuation"


'Start "calculation VOLU1 register"
'==================================

'Start "VOLU1 contribution"
'--------------------------
VOL_PLUS = VOLUME + GAIN_ADJ + OUTP_ATT         'short-cut variable
                                        '? loudness ctrl during VOLU1 ctrl ?
If MIN_VOL1 > (LDN_STRT + GAIN_ADJ + OUTP_ATT) Then     'no Volume_1 control
                                        'below loudness control
                                        '? volume within VOLU1 range ?
  If VOL_PLUS >= MIN_VOL1 Then
    VOLU1 = VOL_PLUS + 40               'volume depending setting of VOLU1
  Else                                  '(40 = correction for 0 dB gain of VOLU1)
    VOLU1 = MIN_VOL1 + 40               'set VOLU1 to MIN_VOL1
  End If
Else                                    'Volume_1 control below loudness control range
                                        ' is possible
                                        '? is volume setting above loudness start ?
  If VOLUME >= LDN_STRT Then
    VOLU1 = VOL_PLUS + 40               'Volume_1 control above loudness control
  Else
                                        '? is volume setting within loudness control range ?
    If VOLUME >= (LDN_STRT - 20) Then           '(20 = loudness range)
      VOLU1 = LDN_STRT + GAIN_ADJ + OUTP_ATT + 40       'within loudness control range

    Else                                'Volume is below loudness start
                                        '? Volume still within VOLU1 ctrl range ?
      If VOL_PLUS >= MIN_VOL1 - 20 Then
        VOLU1 = (VOL_PLUS + 20) + 40            '(20 = loudn. contribution)
      Else
        VOLU1 = MIN_VOL1 + 40                   'min VOLU1 setting
      End If
    End If
  End If
End If                                  'End "VOLU1 contribution"


'Start "Check VOLU1 register and set +20 dB if > 20 dB"
'------------------------------------------------------
```

```
                                '? VOLU1 out of range ?
If VOLU1 > 60 Then
VOLU1 = 60                          'set +20 dB
Else
End If                            'End "calculation VOLU1 register"



'Start "calculation VOLU2_xy registers"
'=====================================

'Start "Load volume control contribution into VOL2_xy registers"
'--------------------------------------------------------------
                                '? mute ?
If VOLUME < (VOLU_RNG + 1) Then
  VOL2_LF = 3                       'last step is mute or ...
Else                              'not mute position
                                '? Vol_2 variation above loudn. ?
  If (LDN_STRT + GAIN_ADJ + OUTP_ATT) >= MIN_VOL1 Then  'no Volume_2 control
                                '     above loudness control
                                '? Volume_2 contribution needed ?
    If VOL_PLUS < (MIN_VOL1 - 20) Then
      VOL2_LF = VOLUME + GAIN_ADJ - (MIN_VOL1 - 20) + 63 'calculate VOLU2_xy, because also
                                '  needed for Volume control
    Else
      VOL2_LF = -OUTP_ATT + 63              'load only output attenuation offset
    End If
  Else                            'Volume_2 control above
                                '      loudness control possible
                                '? Volume_2 contribution needed ?
    If VOL_PLUS < MIN_VOL1 Then             'VOLU2_xy also needed for
                                '      Volume control
                                '? Volume_2 control
      If VOLUME >= LDN_STRT Then            '      above loudness control ?

        VOL2_LF = VOLUME + GAIN_ADJ - MIN_VOL1 + 63     'Yes, Volume_2 control
                                '      above loudness control
      Else

        If VOLUME < (LDN_STRT - 20) Then
          VOL2_LF = VOLUME + GAIN_ADJ + 20 - MIN_VOL1 + 63 'Volume_2 control below
                                  ' loudness control or
        Else
          VOL2_LF = LDN_STRT + GAIN_ADJ - MIN_VOL1 + 63 'VOL2_xy constant
        End If                    '    within loudness control range
      End If
    Else

      VOL2_LF = -OUTP_ATT + 63              'load only output attenuation offset
    End If
  End If
End If

                                'copy VOL2_LF value into all
VOL2_RF = VOL2_LF                   '     remaining output registers
VOL2_LR = VOL2_LF
VOL2_RR = VOL2_LF                   'End "volume contribution"



'Start "Insert balance contribution into VOL2_xy registers"
'---------------------------------------------------------
```

```
                                        '? balance right ?
If BALANCE > 0 Then                     '(balance right means left attenuation)
                                        '? last step (mute) ?
 If BALANCE > (MAX_BAL - 1) Then
  VOL2_LF = 3                           'mute left
  VOL2_LR = 3
 Else
  VOL2_LF = VOL2_LF - BALANCE           'add balance attenuation left
  VOL2_LR = VOL2_LR - BALANCE
 End If
Else                                    'balance left means right
                                        '  attenuation (center no att.)
                                        '? last step (mute) ?
 If BALANCE < (-MAX_BAL + 1) Then
  VOL2_RF = 3                           'mute right
  VOL2_RR = 3
 Else
  VOL2_RF = VOL2_RF + BALANCE           'add balance attenuation right
  VOL2_RR = VOL2_RR + BALANCE
 End If
End If                                  'End "balance contribution"


'Start "Insert fader contribution into VOL2_xy registers"
'---------------------------------------------------------
                                        '? fader front ?
If FADER > 0 Then                       '(fader front means rear attenuation)
                                        '? last step (mute) ?
 If FADER > (MAX_BAL - 1) Then          'mute rear
  VOL2_LR = 3
  VOL2_RR = 3
 Else

  VOL2_LR = VOL2_LR - FADER             'add fader attenuation rear
  VOL2_RR = VOL2_RR - FADER
 End If
Else                                    'fader rear means front
                                        '  attenuation (center no att.)
                                        '? last step (mute) ?
 If FADER < (-MAX_BAL + 1) Then
  VOL2_LF = 3                           'mute front
  VOL2_RF = 3
 Else
  VOL2_LF = VOL2_LF + FADER             'add fader attenuation front
  VOL2_RF = VOL2_RF + FADER
 End If
End If                                  'End "fader contribution"


'Start "Check VOLU2_xy register and set mute if out of range"
'-------------------------------------------------------------
                                        '? VOL2_xy out of range ?
If VOL2_LF < 3 Then
 VOL2_LF = 3                            'set mute LF
End If

If VOL2_RF < 3 Then 'set mute RF
 VOL2_RF = 3                            'set mute RF
End If

If VOL2_LR < 3 Then
 VOL2_LR = 3                            'set mute LR
End If
```

```
If VOL2_RR < 3 Then
  VOL2_RR = 3                          'set mute RR
End If
                              'End "check range"


  DISPL_REG
End Sub
```

CASP Tone / Volume control, simplified Blockdiagram

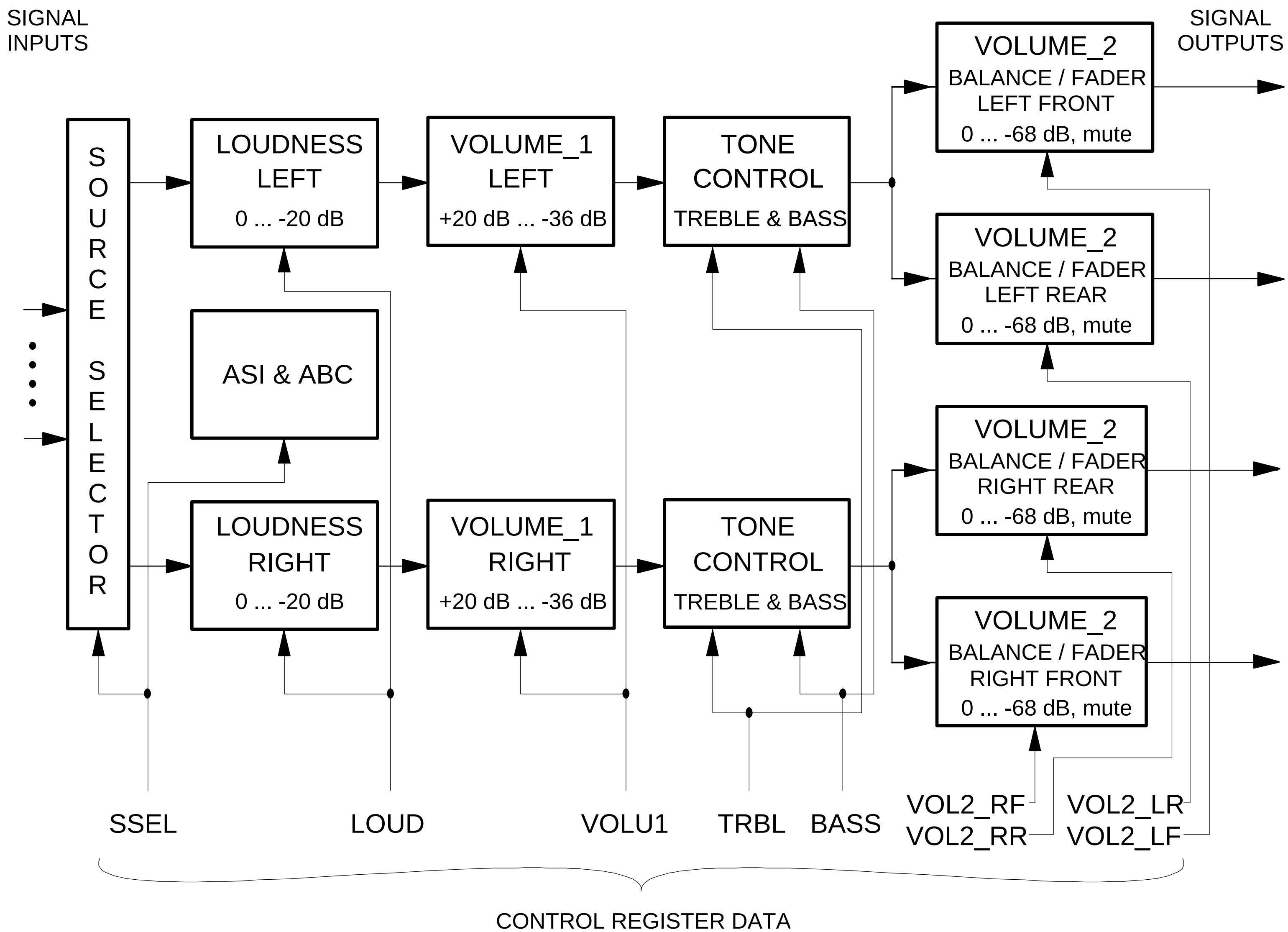
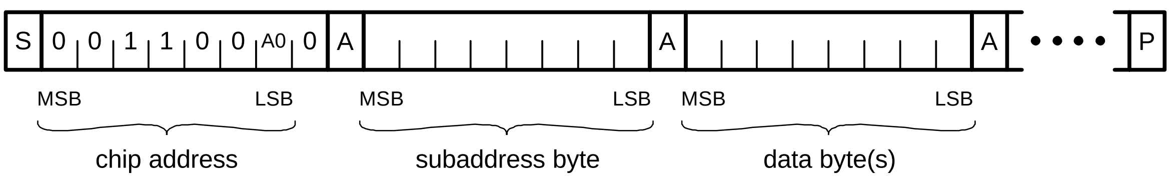


Fig.1

Structure of the CASP I²C Bus Protocol

WRITE MODE:



A0 = defined by address pin ADR

subaddress byte:
(to be used only for write)

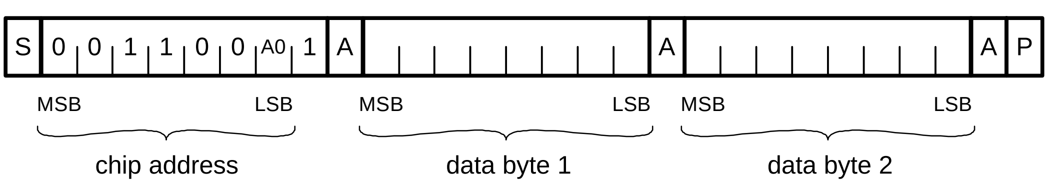
auto-incrmnt. AIOF	bus output BOUT	0	0	subaddress SAD3	SAD2	SAD1	SAD0

AIOF: 0 = autoincrement on
1 = autoincrement off

BOUT: 0 = bus output disabled
1 = bus output enabled

SAD3 .... SAD0 = subaddress for data bytes (write)

READ MODE:



no subaddress byte is used for read mode!

S = start condition
A = acknowledge
P = stop condition

Fig.2

Data Registers for CASP Tone / Volume Control

(subaddr.) register name	data description							
(4) SSEL	ASI / ABC speed					main source selector		
	ASI1	ASI0	1	1	1	MSS2	MSS1	MSS0
(5) BASS	bass filter mode			bass control				
	BSYC	0	BSYB	BAS4	BAS3	BAS2	BAS1	BAS0
(6) TRBL	test mode				treble control			
	HSTM	0	0	0	TRE3	TRE2	TRE1	TRE0
(7) LOUD	loudn. switch			loudness attenuation				
	LOFF	0	0	LSN4	LSN3	LSN2	LSN1	LSN0
(8) VOLU1	audio mute		volume_1 control					
	AMUT	0	VOL5	VOL4	VOL3	VOL2	VOL1	VOL0
(9) VOL2_LF	chime add. left		left front volume_2, balance, and fader control					
	CHML	0	VLF5	VLF4	VLF3	VLF2	VLF1	VLF0
(A) VOL2_RF	chime add. right		right front volume_2, balance, and fader control					
	CHMR	0	VRF5	VRF4	VRF3	VRF2	VRF1	VRF0
(B) VOL2_LR			left rear volume_2, balance, and fader control					
	0	0	VLR5	VLR4	VLR3	VLR2	VLR1	VLR0
(C) VOL2_RR			right rear volume_2, balance, and fader control					
	0	0	VRR5	VRR4	VRR3	VRR2	VRR1	VRR0

Fig.3

Software Control of the CASP Tone / Volume Part

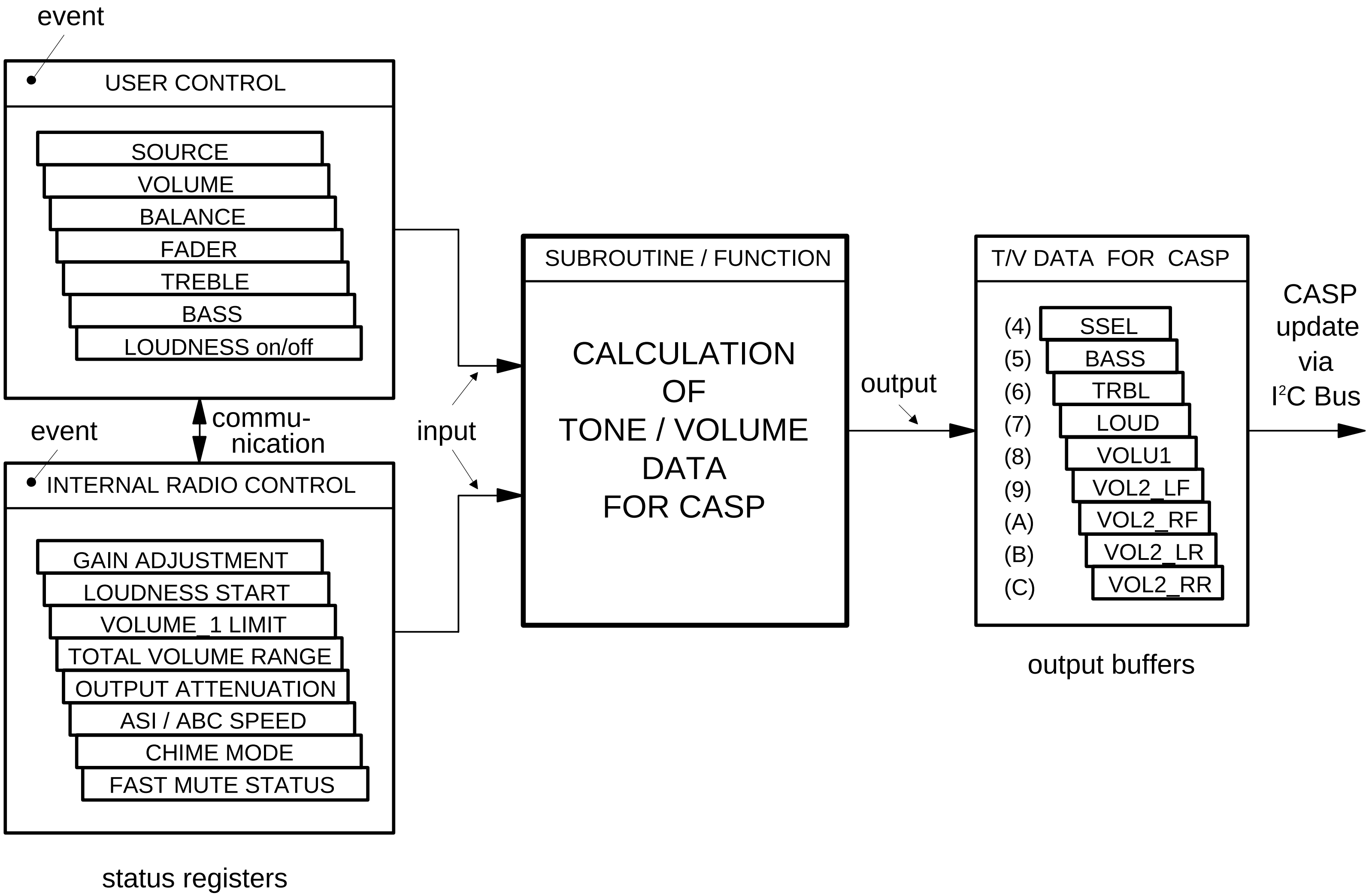


Fig.4

Principle of CASP Volume Control

- Influence of the Parameter "Gain Adjust" (GAIN_ADJ) -

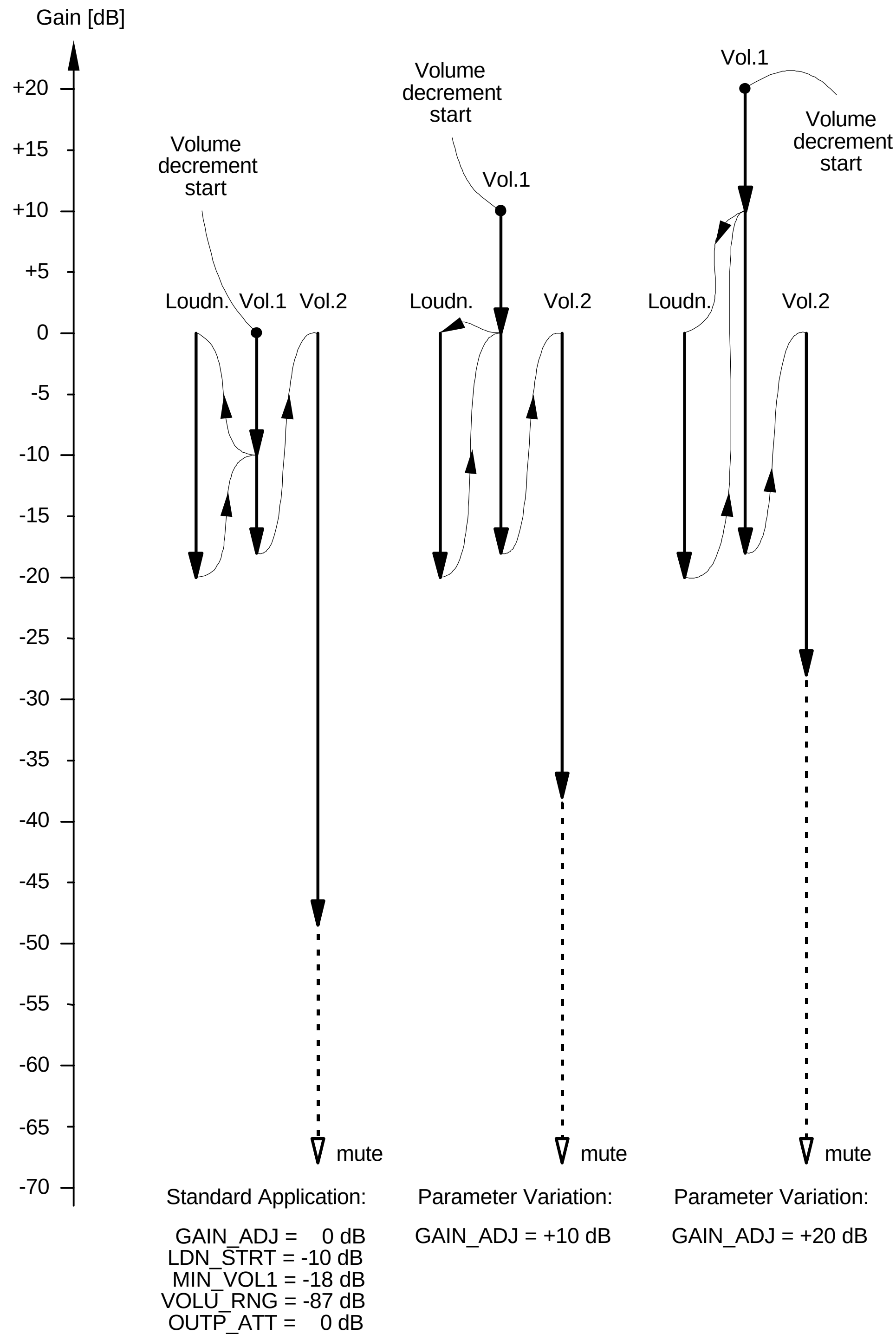


Fig.5

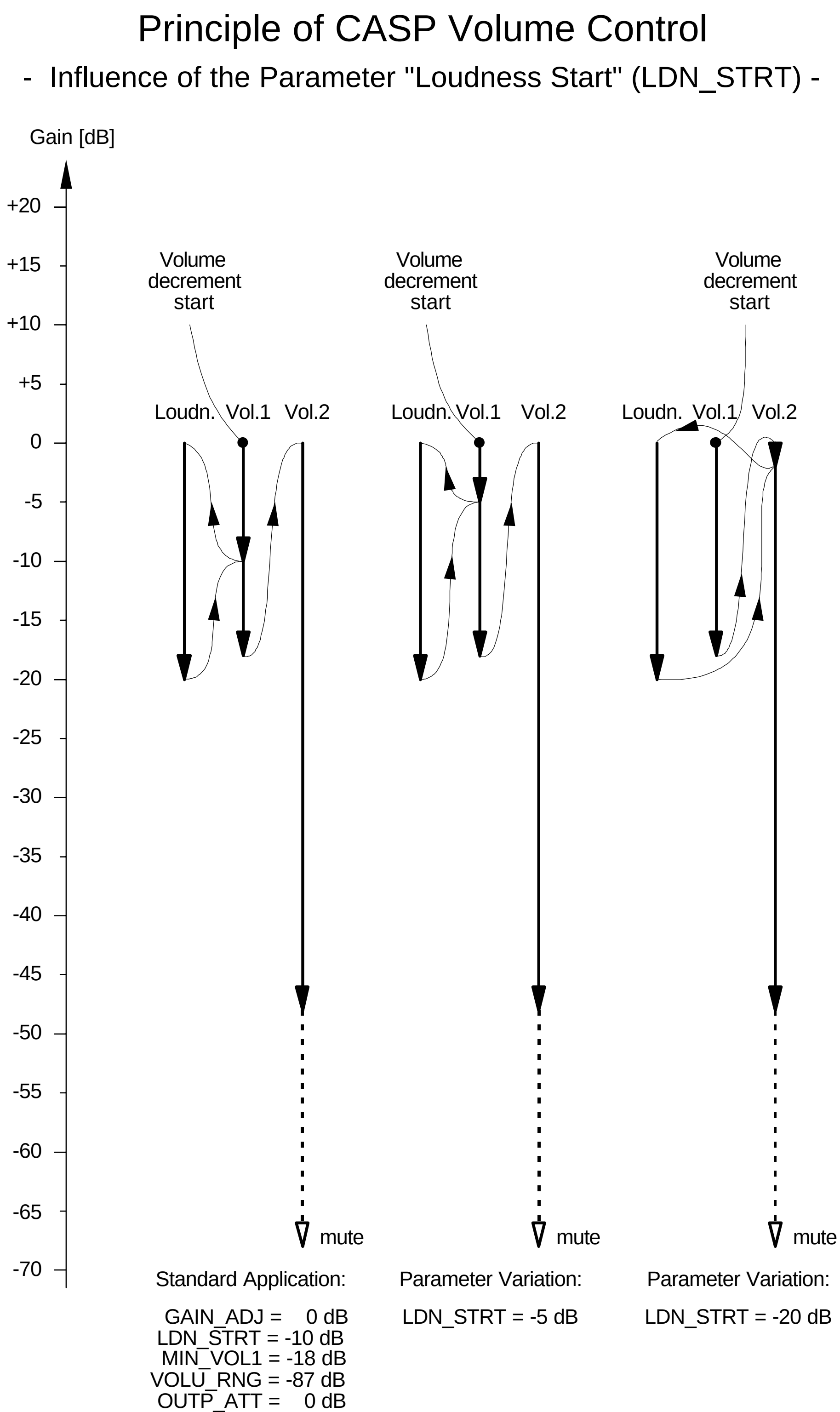

Principle of CASP Volume Control
- Influence of the Parameter "Loudness Start" (LDN_STRT) -
Gain [dB]
+20
+15
+10
+5
0
-5
-10
-15
-20
-25
-30
-35
-40
-45
-50
-55
-60
-65
-70
Volume decrement start
Volume decrement start
Volume decrement start
Loudn. Vol.1 Vol.2
Loudn. Vol.1 Vol.2
Loudn. Vol.1 Vol.2
mute
mute
mute
Standard Application:
GAIN_ADJ = 0 dB
LDN_STRT = -10 dB
MIN_VOL1 = -18 dB
VOLU_RNG = -87 dB
OUTP_ATT = 0 dB
Parameter Variation:
LDN_STRT = -5 dB
Parameter Variation:
LDN_STRT = -20 dB

Fig.6

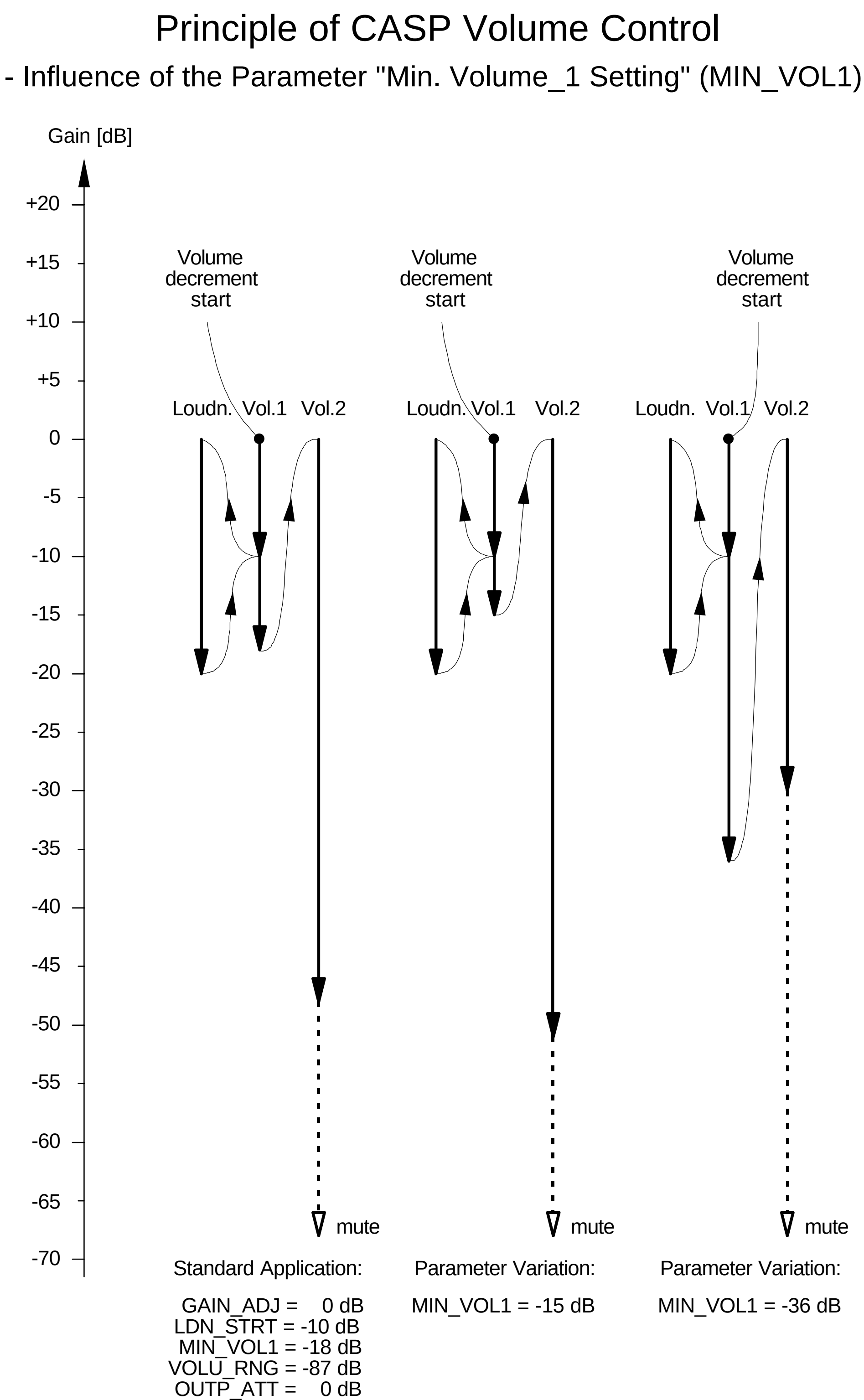

Principle of CASP Volume Control
- Influence of the Parameter "Min. Volume_1 Setting" (MIN_VOL1)
Gain [dB]
+20
+15
+10
+5
0
-5
-10
-15
-20
-25
-30
-35
-40
-45
-50
-55
-60
-65
-70
Volume decrement start
Volume decrement start
Volume decrement start
Loudn. Vol.1 Vol.2
Loudn. Vol.1 Vol.2
Loudn. Vol.1 Vol.2
mute
mute
mute
Standard Application:
GAIN_ADJ = 0 dB
LDN_STRT = -10 dB
MIN_VOL1 = -18 dB
VOLU_RNG = -87 dB
OUTP_ATT = 0 dB
Parameter Variation:
MIN_VOL1 = -15 dB
Parameter Variation:
MIN_VOL1 = -36 dB

Fig.7

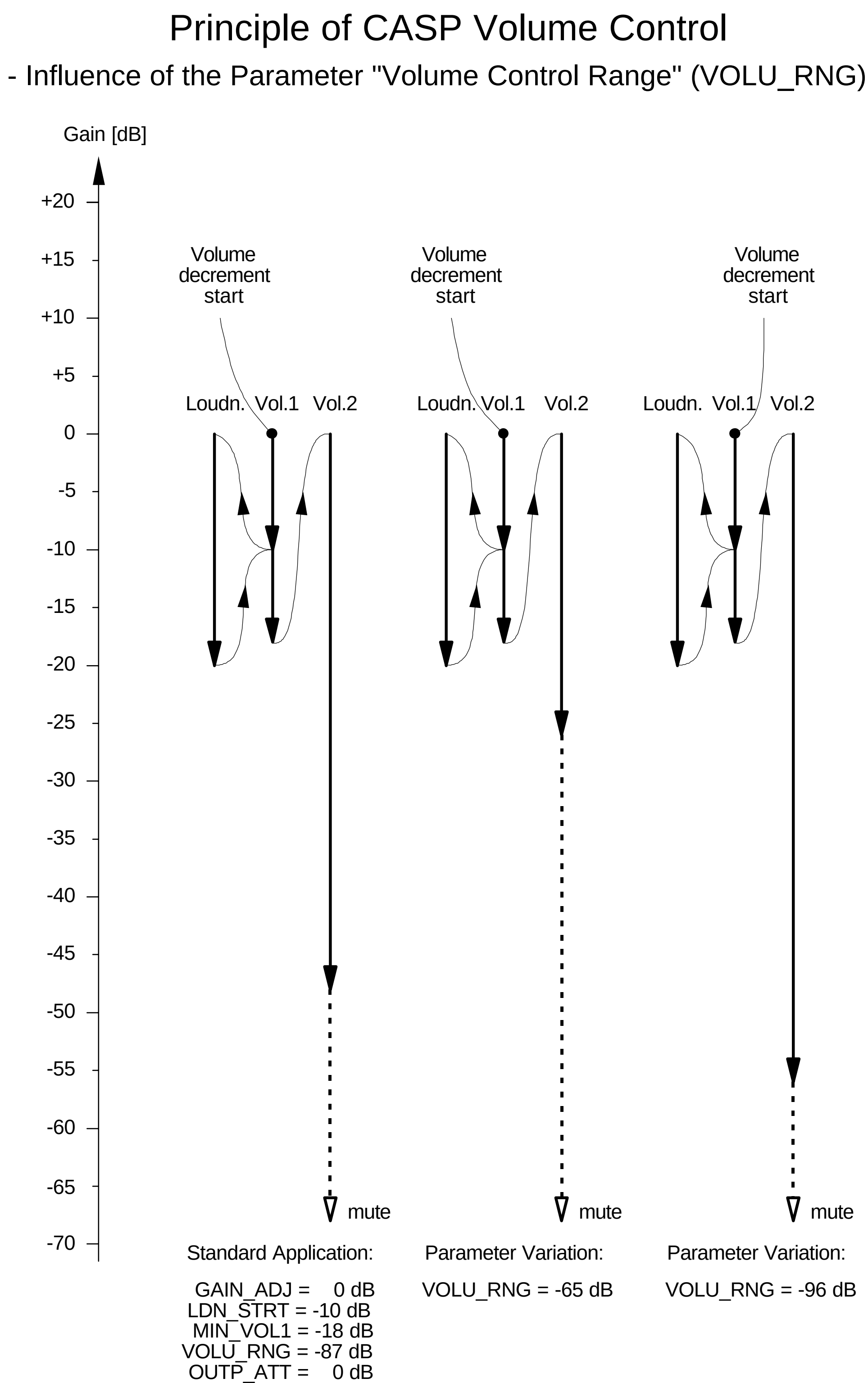

Principle of CASP Volume Control
- Influence of the Parameter "Volume Control Range" (VOLU_RNG)
Gain [dB]
+20
+15
+10
+5
0
-5
-10
-15
-20
-25
-30
-35
-40
-45
-50
-55
-60
-65
-70
Volume decrement start
Volume decrement start
Volume decrement start
Loudn. Vol.1 Vol.2
Loudn. Vol.1 Vol.2
Loudn. Vol.1 Vol.2
mute
mute
mute
Standard Application:
GAIN_ADJ = 0 dB
LDN_STRT = -10 dB
MIN_VOL1 = -18 dB
VOLU_RNG = -87 dB
OUTP_ATT = 0 dB
Parameter Variation:
VOLU_RNG = -65 dB
Parameter Variation:
VOLU_RNG = -96 dB

Fig.8

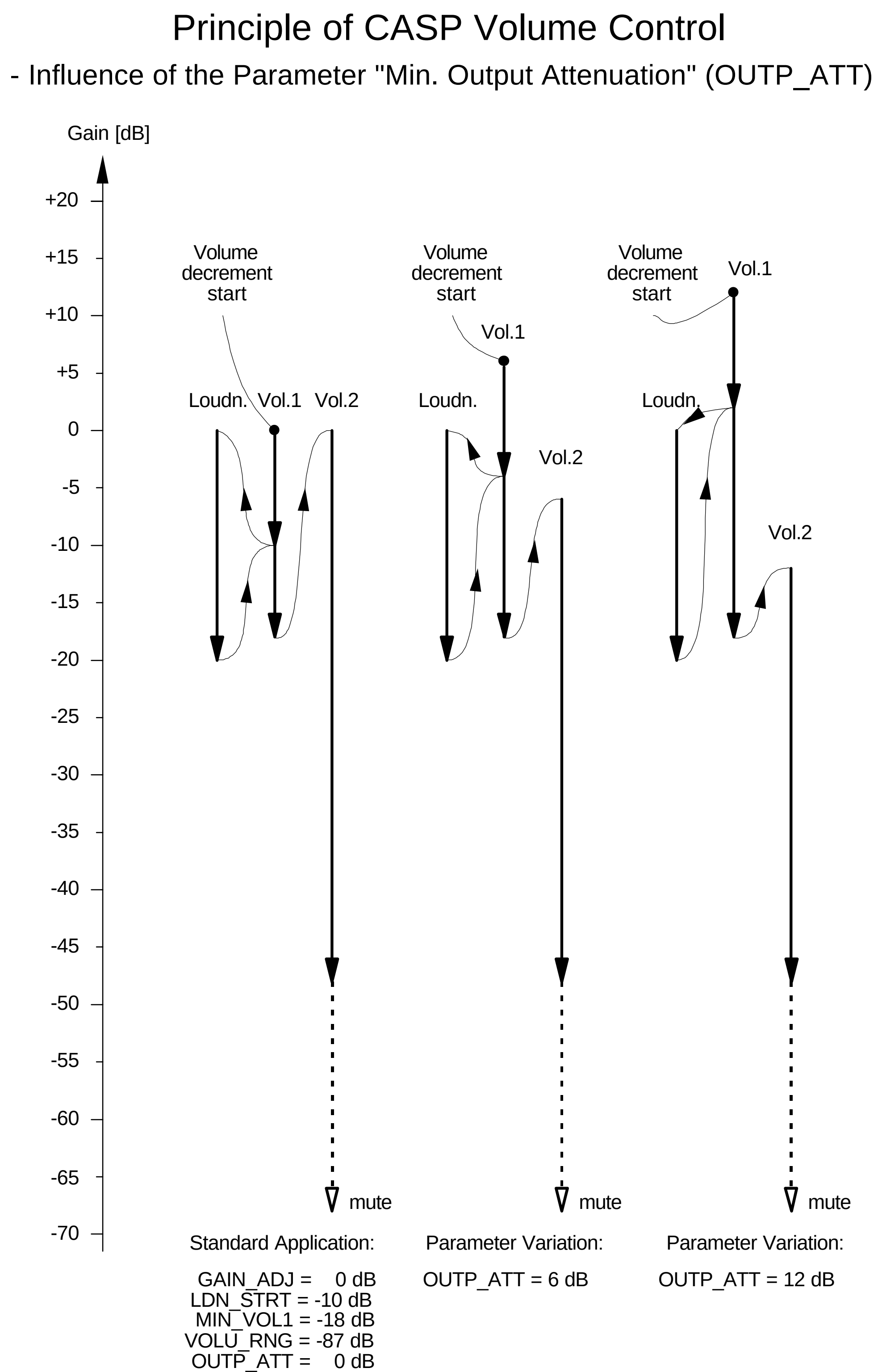

Principle of CASP Volume Control
- Influence of the Parameter "Min. Output Attenuation" (OUTP_ATT)
Gain [dB]
+20
+15
+10
+5
0
-5
-10
-15
-20
-25
-30
-35
-40
-45
-50
-55
-60
-65
-70
Volume decrement start
Loudn.
Vol.1
Vol.2
mute
Volume decrement start
Vol.1
Loudn.
Vol.2
mute
Volume decrement start
Vol.1
Loudn.
Vol.2
mute
Standard Application:
GAIN_ADJ = 0 dB
LDN_STRT = -10 dB
MIN_VOL1 = -18 dB
VOLU_RNG = -87 dB
OUTP_ATT = 0 dB
Parameter Variation:
OUTP_ATT = 6 dB
Parameter Variation:
OUTP_ATT = 12 dB

Fig.9

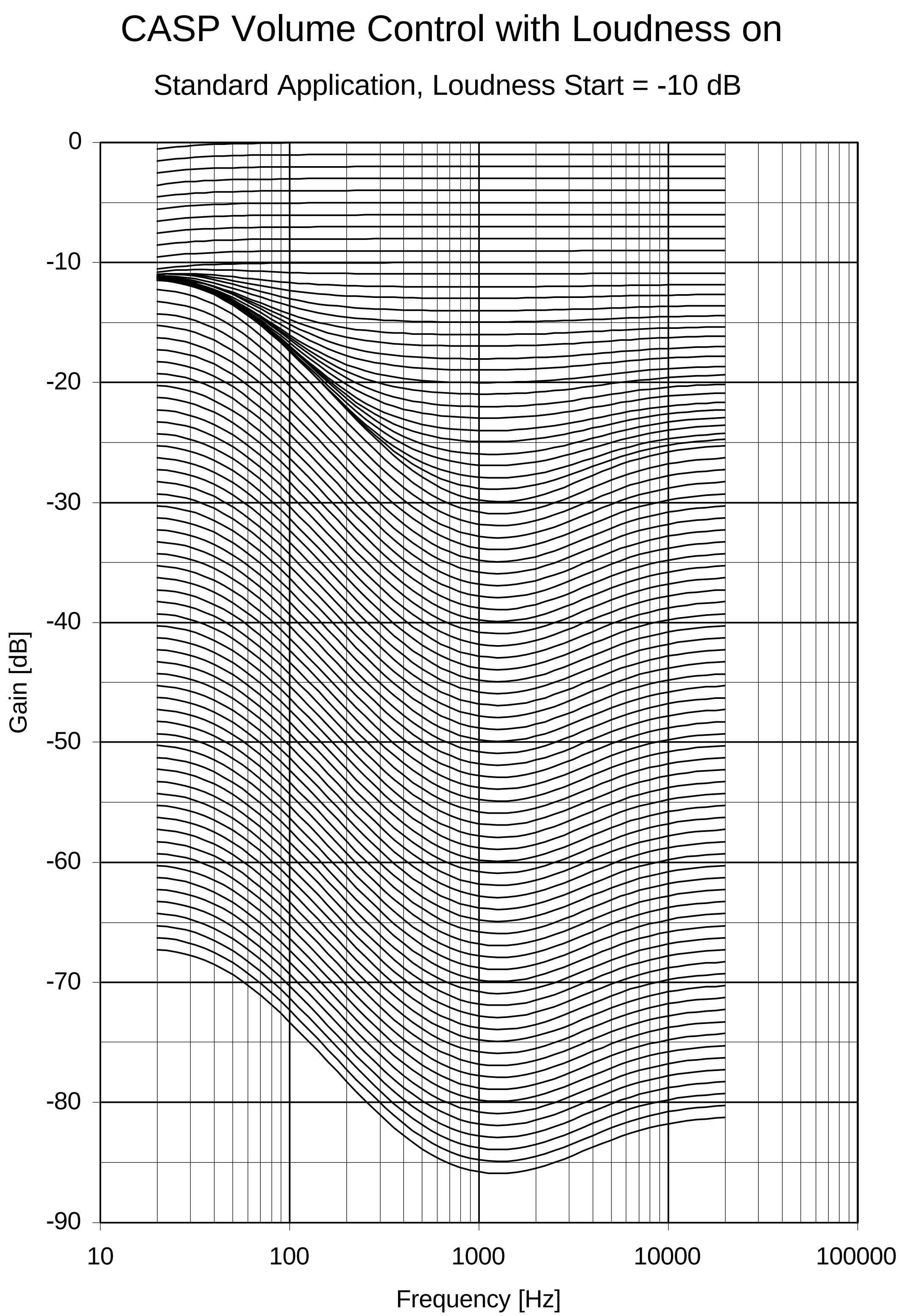
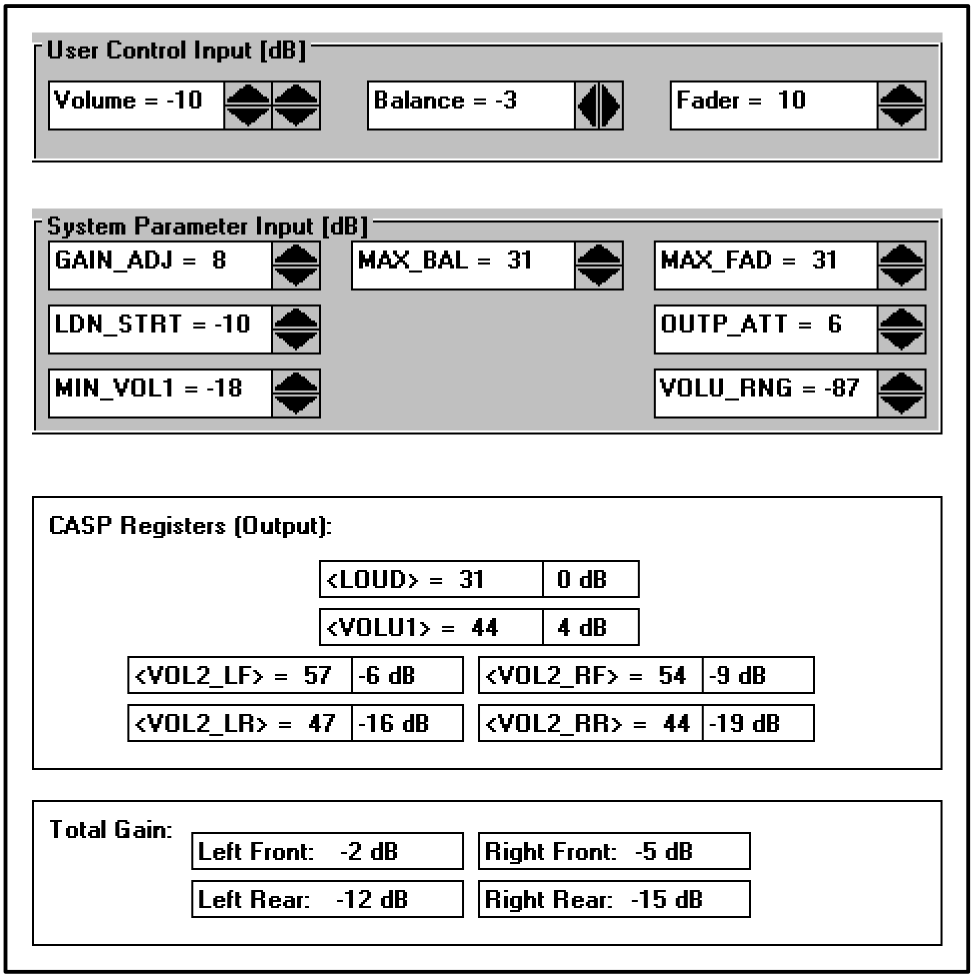

CASP Volume Control with Loudness on
Standard Application, Loudness Start = -10 dB
Gain [dB]
0
-10
-20
-30
-40
-50
-60
-70
-80
-90
10
100
1000
10000
100000
Frequency [Hz]

Fig.10



Input and Output Parameters for
CASP Volume, Balance, and Fader Control

Fig.11