

## **7: An Alternative Signal Set for Second-Order Surround Sound Systems**

It was established in Chapter 4 that the definition of second-order B-format in terms of spherical harmonics necessarily involves some information loss unless the sound field is such as to satisfy the Helmholtz equation; a condition which can not be assumed to hold for an arbitrary sound field.

So far as first-order B-format is concerned, it makes no difference whether the signal set is considered to be defined in terms of spherical harmonics or directional derivatives, since the same set of signals is obtained in each case. It has indeed recently been noted that the performance of ambisonic decoders may be sub-optimal when replaying material recorded in strongly reactive environments [26], because ambisonic decoders are also based on plane wave assumptions; however, that is an issue relating to the use of the signals. Regardless of assumptions made in the design of ambisonic decoders, first-order B-format represents the sound pressure and the three independent first-order derivatives, and no information is in fact lost if we make plane wave assumptions during the design of the microphone.

It may reasonably be considered undesirable that the definition of a system should inherently involve information loss, unless this loss has been shown to be unimportant. In the case of second-order ambisonic systems, there does not seem to be general awareness that any information loss exists, and so it has not been subjected to experiment. There are, therefore, grounds for considering an alternative signal set which does not discard information .

Instead of defining the signals in terms of spherical harmonic polar patterns, this alternative format, which the author presently refers to as  $\mathcal{K}$ -format (“zheh-format”), is defined in terms of the generally independent partial derivatives. Since we consider here only second-order systems, a total of ten signals are required. These signals are defined in such a way as to maintain the maximum possible similarity with B-format.

The second-order  $\mathcal{K}$ -format, or  $\mathcal{K}_2$ -format, signal set is defined as follows:

$$q_0 = \frac{1}{\sqrt{2}} p \tag{7.1a}$$

$$q_x = c \int \frac{\partial p}{\partial x} dt \tag{7.1b}$$

$$Q_y = c \int \frac{\partial p}{\partial y} dt \quad (7.1c)$$

$$Q_z = c \int \frac{\partial p}{\partial z} dt \quad (7.1d)$$

$$Q_{x^2} = c^2 \iint \frac{\partial^2 p}{\partial x^2} dt dt \quad (7.1e)$$

$$Q_{xy} = 2c^2 \iint \frac{\partial^2 p}{\partial x \partial y} dt dt \quad (7.1f)$$

$$Q_{y^2} = c^2 \iint \frac{\partial^2 p}{\partial y^2} dt dt \quad (7.1g)$$

$$Q_{xz} = 2c^2 \iint \frac{\partial^2 p}{\partial x \partial z} dt dt \quad (7.1h)$$

$$Q_{yz} = 2c^2 \iint \frac{\partial^2 p}{\partial y \partial z} dt dt \quad (7.1i)$$

$$Q_{z^2} = c^2 \iint \frac{\partial^2 p}{\partial z^2} dt dt \quad (7.1j)$$

It will be noted that the zeroth-order and first-order signals here are identical to the  $W$ ,  $X$ ,  $Y$  and  $Z$  signals of B-format; this is not surprising in view of the preceding discussion. The relative scaling of the pressure signal is retained to enhance compatibility. Furthermore, the  $\mathcal{K}_2$ -format signals  $Q_{xz}$ ,  $Q_{yz}$ , and  $Q_{xy}$  are equivalent to the second-order B-format signals  $S$ ,  $T$  and  $V$  respectively.

Clearly, if a recording is made using these ten signals, then the B-format signals that would have been recorded can subsequently be derived; seven of the signals are identical, and the remaining two second-order B-format signals are linear combinations of  $\mathcal{K}_2$ -format signals:

$$R = \frac{1}{2} (3Q_{z^2} - \sqrt{2}Q_0) \quad (7.2a)$$

$$U = Q_{x^2} - Q_{y^2} \quad (7.2b)$$

It is not surprising that  $\mathcal{K}$ -format is entirely backward compatible with B-format, since information that is retained can always be discarded at a later stage.

By means of a suitable linear transformation, the  $\mathcal{K}_2$ -format signals may be converted into a signal set consisting of second-order B-format together with one additional signal. Selection

of a suitable transformation amounts to defining the additional signal in terms of  $\mathcal{K}$ -format signals; denoting this signal  $B$ , the simplest option is to set

$$B = \mathcal{U}_{x^2} \quad (7.3)$$

The transformation between the two signal sets is then defined by equations (7.2) and (7.3), together with the equivalences between the remaining signals. Conversion back to  $\mathcal{K}_2$ -format requires  $\mathcal{U}_{x^2}$ ,  $\mathcal{U}_{y^2}$  and  $\mathcal{U}_{z^2}$  to be recovered from  $R$ ,  $U$  and  $B$ :

$$\mathcal{U}_{x^2} = B \quad (7.4a)$$

$$\mathcal{U}_{y^2} = B - U \quad (7.4b)$$

$$\mathcal{U}_{z^2} = \frac{1}{3}(2R + \sqrt{2}W) \quad (7.4c)$$

It is expected that it will be possible to derive the ten  $\mathcal{K}_2$ -format signals using the second-order soundfield microphone array. This can obviously be done for the seven signals which correspond directly to B-format signals. It is assumed that it can also be done for the remaining three signals, again using techniques of the Blumlein difference type, but no attempt has yet been made to determine the combination of capsule outputs which should be employed to achieve this.

It is by no means certain that this approach will offer a substantial advantage. It may be that relatively little perceptually useful information is lost as a consequence of the B-format assumptions. Furthermore, no consideration has been given as to how a decoder might employ the extra information retained to advantage. The author proposes this alternative signal format as a possibility which may be worthy of investigation, not as a solution to a problem which has been definitely been found to exist.

## **8: Conclusions & Further Work**

### 8.1: Conclusions

Chapter 6 of this thesis contains information required to facilitate the design and construction of a second-order soundfield microphone; the A-B matrix coefficients given in table 6.1 are of primary importance, while the frequency response functions derived in section 6.4 will assist in the design of suitable spaced-to-coincident conversion filters.

As well as permitting the design of a second-order soundfield microphone, this thesis presents material which enables its expected performance to be characterised. The description of the proximity effect for second-order pressure gradient microphones is significant in this respect, as is the analysis of the presence of unwanted spherical harmonic components in the derived B-format signals (which shows that the zeroth-order and first-order B-format signals obtained from a second-order soundfield microphone are better approximations to the theoretical ideals than are the signals obtained using a first-order soundfield microphone).

It has been shown that most of the facilities provided by the first-order soundfield microphone for the manipulation of the encoded sound field may similarly be implemented with a second-order soundfield microphone; however, it is not possible to extend the dominance transformation to work with second-order signals.

The analysis of second-order pressure gradient microphones in Chapter 3 is of value independently of its relevance to the second-order soundfield microphone, since a general theoretical treatment of second-order microphones has not previously been available, and since the methods employed may be extended in an obvious and logical way to describe pressure gradient microphones of any order. Section 3.6 demonstrates that aspects of the operation of the second-order soundfield microphone may be understood in terms of the Blumlein difference technique. The more directional polar responses which may be obtained by using second-order microphones motivate the possible use of the second-order soundfield microphone as a tool for the synthesis of arbitrary coincident arrays of second-order microphones.

In Chapter 4, it was shown that the assumptions on which the B-format signal specification are based may result in loss of information. The alternative signal set described in Chapter 7

does not depend on these assumptions, and can therefore preserve this information; it is however completely backward compatible with B-format.

## 8.2: Further Work

### 8.2.1: Effects of Variations in Capsule Parameters

Investigation of the effects of variations in the parameters between the individual microphone capsules will be of interest. It has previously been noted that the overall responsivity  $G$  and the polar pattern constant  $a'$  may vary in a substantially known way as a function of frequency; this can be accounted for in the design of the non-coincidence compensation filters. However, these parameters may also be subject to random differences between capsules. Additionally, both the positions and the orientations of the capsules may be subject to error.

It is simple to devise a model which allows the effects of variations in the responsivity, polar pattern and distance from the centre of the array to be studied. It appears to be rather more difficult to include angular mislocation or incorrect orientation; a model involving these has not yet been devised. Taking the FU capsule as an example, we may write for the output signal

$$v_{FU} = G(1 + e_{FU}^G) \left[ (a' + D_{FU}^{a'}) + (1 - a' - D_{FU}^{a'}) \hat{\mathbf{u}}_{FU} \cdot \hat{\mathbf{d}} \right] e^{jkr(1 + e_{FU}^r) \hat{\mathbf{u}}_{FU} \cdot \hat{\mathbf{d}}} \quad (8.1)$$

where  $e_{FU}^G$  is the relative error in the responsivity,  $D_{FU}^{a'}$  is the absolute error in the normalised polar pattern constant, and  $e_{FU}^r$  is the relative error in the distance from the array centre.

Using this model, previous calculations may be repeated to generate expressions for the spherical harmonic components of each signal when capsule parameter errors are present. Since the simplifications that result from the assumption that all capsules have identical parameters are no longer available, it may be appreciated that the resulting expressions are rather long and unwieldy.

It is hoped that this will lead to an understanding of how sensitive the performance of the second-order soundfield microphone is to capsule parameter variations.

It may also be of interest to apply this same analysis to the first-order soundfield microphone, and thereby to investigate the questions of whether the performance of the second-order soundfield microphone is more or less robust in the presence of capsule parameter variations.

### *8.2.2: Design of Control Unit*

If the second-order soundfield microphone is to be useful as a tool for the synthesis of arbitrary arrays of coincident second-order microphones, then a suitable control unit will be required. Design of such a unit presents some considerable difficulty. A system which (for example) requires the user to know the Laplace series coefficients for each desired polar pattern is unlikely to gain widespread acceptance. It is not clear how the desired functionality can be provided in such a way that the underlying theory is transparent to the user.

Such issues do not arise in connection with the use of the microphone for ambisonic B-format recording; there are no conceptual difficulties with the provision of controls for such facilities as inversion or side-fire / end-fire switching, which indeed are present on the control units for existing first-order soundfield microphones.

### *8.2.3: Other Areas for Future Research*

Since the dominance transformation can not be applied to second-order B-format signals, and dominance is considered to be a useful facility for the manipulation of first-order B-format signals, it would be desirable to devise some transformation having similar properties which can be applied to B-format signals of any order.

It is hoped that the full  $\mathcal{K}_2$ -format signal set can be obtained directly from the second-order soundfield microphone capsule array; it has not yet been established how this may be achieved.