EFFICIENT CONVERSION METHOD BETWEEN SUBBAND DOMAIN REPRESENTATIONS

Abdellatif BENJELLOUN TOUIMI and Adil MOUHSSINE
Speech and Sound Technologies and Processing Lab., France Telecom Research & Development
Email: {abdellatif.benjellountouimi, adil.mouhssine}@francetelecom.com

ABSTRACT

This paper presents an efficient method for the conversion between subband domain representations of different filter banks. It advantageously replaces the conventional cascade of synthesis and analysis banks leading to algorithmic delay reduction and high degree of parallelism for implementation. It is especially useful for transcoding applications between subband compressed multimedia signal formats.

1. INTRODUCTION

Despite the advancement of multimedia coding formats and the performances of transport and access networks, the concept of *Universal Multimedia Access* (UMA) is still not fully achieved [1]. One problem is the terminals heterogeneity, specifically due to the different coding formats. Transcoding is generally used to cope with this problem. Conventional transcoding consists in totally decoding the initial format and then re-encoding to the new format (figure 1). It has several drawbacks such as computational complexity, algorithmic delay and quality degradation. Intelligent transcoding has been proposed as an alternative solution: it consists in generating the new format without totally decoding the first one. If several methods have been proposed for speech and video coding formats [3], there are few works for audio [8]. Audio coders are based on subband coding and conversions between different subband domain representations are still not solved.

Several works have been developed in image and video to deal with transform coding formats conversion. In [4], a general method is developed for linear filtering and multirate processing in the transform domain combined with the conversion between different sizes transforms representations. In [2], the transposition between different sizes DCT transforms has also been proposed. These methods are restricted to non-lapped transforms. In [5], a conversion method between MDCT and DFT domains is given. The most advanced work can be found in [7], where a combination method of synthesis and analysis filter banks is presented. However, it remains specific to modulated filter banks and it is restricted to the cases where the two banks subbands numbers are multiple one to the other.

This paper presents a generic conversion method between subband domains that could be applied to any maximally-decimated filter banks without any condition on their sizes. Section 2 develops the general form of the conversion system. A representation of the system as linear periodically time-varying (LPTV) is given in section 3, based on which an efficient implementation is developed in section 4. Finally, section 5 generalizes the method to the combination of filtering and/or re-sampling with conversion.

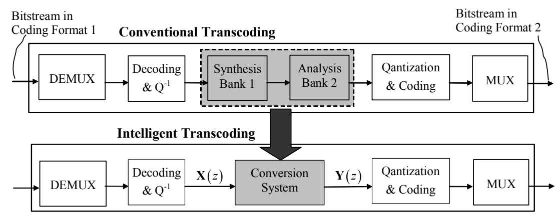


Fig. 1. Conventional and intelligent transcoding diagrams.

2. SUBBAND DOMAINS CONVERSION SYSTEM

2.1 Conversion System Formulation

Consider a L-band synthesis filter bank followed by an M-band analysis filter bank (figure 2). The two banks are maximally-decimated and are defined by the filters vectors:

$$\mathbf{f}(z)=\left[\mathrm{F}_0(z)\,\mathrm{F}_1(z)\ldots\mathrm{F}_{L-1}(z)\right] \quad (1)$$

and

$$\mathbf{h}(z)=\left[\mathrm{H}_0(z)\mathrm{H}_1(z)\ldots\mathrm{H}_{M-1}(z)\right]^{\mathrm{T}}. \quad (2)$$

The conversion problem consists in determining a direct conversion system between the subband signals vectors in the first and second filter bank domains, $\mathbf{X}(z)$ and $\mathbf{Y}(z)$ where:

$$\mathbf{X}(z)=\left[\mathrm{X}_0(z)\,\mathrm{X}_1(z)\ldots\mathrm{X}_{L-1}(z)\right]^{\mathrm{T}} \quad (3)$$

and

$$\mathbf{Y}(z)=\left[\mathrm{Y}_0(z)\,\mathrm{Y}_1(z)\ldots\mathrm{Y}_{M-1}(z)\right]^{\mathrm{T}}. \quad (4)$$

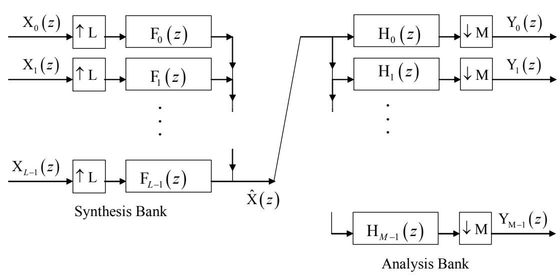


Fig. 2. Subband domains conventional conversion.

Let $K=\mathrm{lcm}(M,L)$ and $p_1=K/M$, $p_2=K/L$. Consider $\mathbf{U}(z)$ (resp. $\mathbf{V}(z)$) the vector resulting from a p_2-order (resp. p_1-order) polyphase decomposition of $\mathbf{X}(z)$ (resp. $\mathbf{Y}(z)$):

$$\mathbf{U}(z)=\left[\mathbf{X}_0^T(z),\mathbf{X}_1^T(z),\ldots\,\mathbf{X}_{p_2-1}^T(z)\right]^{\mathrm{T}}, \quad (5)$$

$$\mathbf{V}(z)=\left[\mathbf{Y}_0^T(z),\mathbf{Y}_1^T(z),\ldots\,\mathbf{Y}_{p_1-1}^T(z)\right]^{\mathrm{T}}. \quad (6)$$

Let $\mathbf{g}(z)=\mathbf{h}(z)\mathbf{f}(z)$, be the $M\times L$ product matrix of analysis and synthesis filters. The subband domain representations conversion is given by:

0-7803-9332-5/05/$20.00 ©2005 IEEE

$$\mathbf{V}(z)=\mathbf{T}(z)\mathbf{U}(z)\,. \tag{7}$$

$\mathbf{T}(z)$, the conversion matrix, of size $K\times K$, is given by:

$$\mathbf{T}(z)=\left[\mathbf{v}(z)\otimes\mathbf{g}(z)\right]\Big|_{\downarrow K}\,, \tag{8}$$

where $\mathbf{v}(z)$ is a $p_1\times p_2$ matrix defined by:

$$\mathrm{v}_{ij}(z)=z^{iM-jL}\,,\ 0\le i\le p_1-1\,,\ 0\le j\le p_2-1\,. \tag{9}$$

The operation $\otimes$ denotes the Kronecker product and $\downarrow K$ stands for factor K decimation. The proof of this formula is given in the annex. The multirate representation of the conversion system, shown in figure 3, has three consecutive blocks: serial-to-parallel conversion block, filtering matrix and parallel-to-serial conversion block.

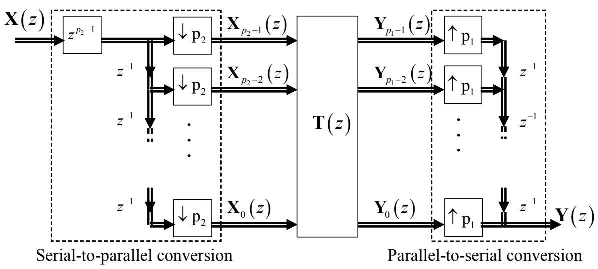


Fig. 3. Subband domains conversion system.

2.2. Minimum Delay Conversion System

In order to minimise the algorithmic delay of the conversion system, the advances/delays z^a, and z^b, $a,b\in\mathbb{Z}$, are introduced respectively at the inputs of the synthesis and the analysis banks. The formula (8) becomes:

$$\mathbf{T}(z)=\left[\left[z^{aL+b+iM-jL}\mathbf{g}(z)\right]\Big|_{\downarrow K}\right]_{\substack{0\le i\le p_1-1,\\ 0\le j\le p_2-1}}\,, \tag{10}$$

The exponents, $e_{ij}=aL+b+(iM-jL)$, $0\le i\le p_1-1$, $0\le j\le p_2-1$, vary between $e_{\min}=aL+b-K+L$ and $e_{\max}=aL+b+K-M$. The filter matrix, $\mathbf{T}(z)$, is causal iff $e_{\max}\le K-1$, i.e. $aL+b\le M-1$. While meeting this inequality constraint, a and b can be tuned to build different conversion systems with different algorithmic delays. For a and b verifying $aL+b=M-1$, the minimum delay system is obtained and the expressions of the matrix elements $\mathbf{v}(z)$ are:

$$\mathrm{v}_{ij}(z)=z^{M-1+iM-jL}\,,\ 0\le i\le p_1-1\,,\ 0\le j\le p_2-1\,. \tag{11}$$

In the following, the framework of minimum delay conversion system defined by the equations (8) and (11) is considered.

2.3. Interpretation of the Conversion Matrix

In general, for a filter, $\mathrm{S}(z)$, its K-order polyphase components can be written as [6]:

$$\mathrm{S}_l(z)=\left[z^l\,\mathrm{S}(z)\right]\Big|_{\downarrow K}\,,\ 0\le l\le K-1\,. \tag{12}$$

For $0\le e_{ij}\le K-1$, $\left[z^{e_{ij}}\mathbf{g}(z)\right]\Big|_{\downarrow K}$, is hence the e_{ij}^{th} polyphase component of $\mathbf{g}(z)$. For $e_{ij}<0$, we have $\left[z^{e_{ij}}\mathbf{g}(z)\right]\Big|_{\downarrow K}=z^{-1}\left[z^{K+e_{ij}}\mathbf{g}(z)\right]\Big|_{\downarrow K}$. As $K+e_{ij}\le K-1$ for a such case, $\left[z^{e_{ij}}\mathbf{g}(z)\right]\Big|_{\downarrow K}$ is interpreted as the $\left(K+e_{ij}\right)^{\text{th}}$ polyphase component of $\mathbf{g}(z)$ followed by a delay z^{-1}. The elements of the matrix $\mathbf{T}(z)$ can then be explicitly expressed as:

$$\mathrm{T}_{ml}(z)=\begin{cases}\mathrm{G}_{nk}^{e_{ij}}(z) & \text{, if } 0\le e_{ij}\le K-1,\\ z^{-1}\,\mathrm{G}_{nk}^{K+e_{ij}}(z) & \text{, if } e_{ij}<0,\end{cases} \tag{13}$$

for $0\le m,l\le K-1$. $\mathrm{G}_{nk}^{r}(z)$, $0\le r\le K-1$, denotes the r^{th} polyphase component of the product filter, $\mathrm{G}_{nk}(z)=\mathrm{H}_n(z)\mathrm{F}_k(z)$, $0\le n\le M-1$, $0\le k\le L-1$, corresponding to a K-order decomposition. (i,n) and (j,k) are deduced from m and l by:

$$i=\lfloor m/M\rfloor\,,\ n=m-iM \text{ and } j=\lfloor l/L\rfloor\,,\ k=l-jL\,. \tag{14}$$

3. LPTV REPRESENTATION

Before addressing the general case, two particular cases where M and L are multiple one to the other, are considered.

3.1. Particular Case $L=pM$

In this case, $(K,p_1,p_2)=(L,p,1)$. The formula (7) becomes $\mathbf{V}(z)=\mathbf{T}(z)\mathbf{X}(z)$, and the conversion matrix is of size $L\times L$ and is given by:

$$\mathbf{T}(z)=\left[\left[z^{M-1}\mathbf{g}^{\mathrm{T}}(z)\right]\Big|_{\downarrow L}\ \ \left[z^{2M-1}\mathbf{g}^{\mathrm{T}}(z)\right]\Big|_{\downarrow L}\ \ \dots\ \ \left[z^{pM-1}\mathbf{g}^{\mathrm{T}}(z)\right]\Big|_{\downarrow L}\right]^{\mathrm{T}}. \tag{15}$$

The conversion system is reduced to the filtering matrix followed by the parallel-to-serial conversion block. It can be represented as in figure 4a illustrating that the system is LPTV of period $pT_{s_2}=T_{s_1}$, characterized by a set of p transfer matrices, $\mathbf{A}_k(z)$, $0\le k\le p-1$, defined by:

$$\mathbf{A}_k(z)=\left[z^{(k+1)M-1}\mathbf{g}(z)\right]\Big|_{\downarrow L}\,,\ 0\le k\le p-1\,. \tag{16}$$

T_{s_1} and T_{s_2} denote respectively the sampling period in the first and the second subband domain. The system output, $\mathbf{Y}[n]$, at the instant nT_{s_2}, is equal to the output of filtering matrix $\mathbf{A}_k(z)$, where $k=n \bmod p$, at the instant $p\lfloor n/p\rfloor T_{s_2}=\lfloor n/p\rfloor T_{s_1}$.

3.2. Particular Case $M=pL$

In this case, $(K,p_1,p_2)=(M,1,p)$. The formula (7) becomes $\mathbf{Y}(z)=\mathbf{T}(z)\mathbf{U}(z)$, and the conversion matrix is $M\times M$ of size and is given by:

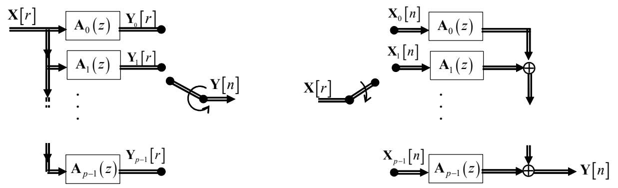
$$\mathbf{T}(z)=\left[\left[z^{pL-1}\mathbf{g}(z)\right]\Big|_{\downarrow M},\left[z^{(p-1)L-1}\mathbf{g}(z)\right]\Big|_{\downarrow M},\dots,\left[z^{L-1}\mathbf{g}(z)\right]\Big|_{\downarrow M}\right]. \tag{17}$$

The conversion system is reduced in this case to the serial-to-parallel conversion block followed by the filtering matrix. It is represented in figure 4b as an LPTV system of period $pT_{s_1}=T_{s_2}$, characterized by a set of p transfer matrices, $\mathbf{A}_k(z)$, $0\le k\le p-1$, defined by:

$$\mathbf{A}_k(z)=\left[z^{(p-k)L-1}\mathbf{g}(z)\right]\Big|_{\downarrow M}\,,\ 0\le k\le p-1\,. \tag{18}$$

The output of the conversion system, $\mathbf{Y}[n]$, at instant nT_{s_2}, is equal to the sum of the outputs of filtering matrices $\mathbf{A}_k(z)$, $0\le k\le p-1$, which have respectively as inputs

$\mathbf{X}\left[(n-1)p+k+1\right]$, at the respective instants $\left((n-1)p+k+1\right)T_{s_1}=(n-1)T_{s_2}+(k+1)T_{s_1}$.



(a) Case $L=pM$ **(b) Case** $M=pL$

Fig. 4. LPTV representations of the conversion system for the particular cases of M and L.

3.3. General Case

Taking into account the explanations given for the previous particular cases, and the general formula of $\mathbf{T}(z)$, the new structure of the proposed conversion system can be represented as in figure 5. Such structure contains p_1 LPTV sub-systems each one of period $p_2T_{s_1}$. The i^{th} sub-system of this set, $0\le i\le p_1-1$, is characterized by the following p_2 transfer matrices:

$$\mathbf{A}_{ij}(z)=\left[z^{M-1+iM-jL}\mathbf{g}(z)\right]_{\downarrow K},\ 0\le j\le p_2-1, \tag{19}$$

where the explicit expressions of theirs elements are given by:

$$\mathrm{A}_{ij,nk}(z)=\begin{cases}\mathrm{G}_{nk}^{e_{ij}}(z) & \text{, if } 0\le e_{ij}\le K-1,\\ z^{-1}\,\mathrm{G}_{nk}^{K+e_{ij}}(z) & \text{, if } e_{ij}<0,\end{cases} \tag{20}$$

for $0\le n\le M-1$ and $0\le k\le L-1$.

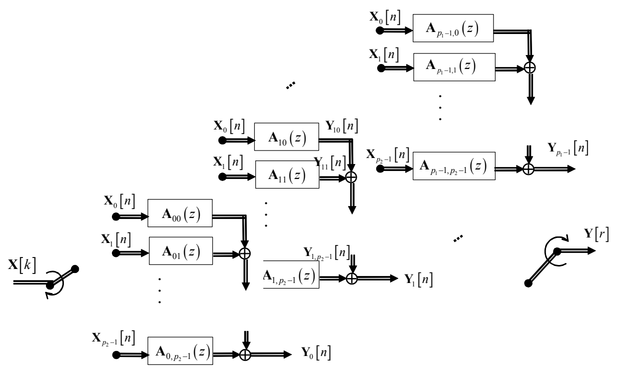


Fig. 5. LPTV representation of the conversion system for the general case.

The set of these sub-systems operates in parallel. One of their outputs is periodically chosen, with a period $p_1T_{s_2}$, as the output of the system. The global system is hence LPTV with period KT_s, where T_s is the time-domain sampling period. Indeed, we have $p_1T_{s_2}=p_2T_{s_1}=KT_s$.

The frequency of the two selector switches at the input and the output of this new structure is f_s/K. It is also the operating frequency of the transfer matrices $\mathbf{A}_{ij}(z)$. So, there is only one operating frequency equal to f_s/K instead of three. This is another advantage of the proposed conversion system.

The system output, $\mathbf{Y}[r]$, at instant rT_{s_2}, is equal to the output of the i^{th} LPTV sub-system, at instant rT_{s_2}, where $i=r\mod p_1$. The input of the system, $\mathbf{X}[k]$, at instant kT_{s_1}, is directed toward the j^{th} inputs of each of the p_1 LPTV sub-systems, where $j=k\mod p_2$.

4. LAPPED TRANSFORMS BASED IMPLEMENTATION

In this section, the filter banks are supposed to be FIR. The transfer matrices $\mathbf{A}_{ij}(z)$ can be hence written as $\mathbf{A}_{ij}(z)=\sum_{n=0}^{N-1}\mathbf{B}_{ij,n}z^{-n}$, where $\mathbf{B}_{ij,n}$ are $M\times L$ matrices. The transfer matrix $\mathbf{A}_{ij}(z)$ is associated to the $NM\times L$ transform matrix $\mathbf{B}_{ij}$ defined by:

$$\mathbf{B}_{ij}=\left[\mathbf{B}_{ij,0}^T\quad \mathbf{B}_{ij,1}^T\quad \dots\quad \mathbf{B}_{ij,N-1}^T\right]^T. \tag{21}$$

As each transfer matrix $\mathbf{A}_{ij}(z)$ contains equal length filters depending on the e_{ij} value, the matrix $\mathbf{B}_{ij}$ has the same dependency. $\mathbf{B}_{ij}$ contains null blocks and its form is given by:

$$\mathbf{B}_{ij}=\begin{cases}\left[\mathbf{B}_{ij,0}^T\quad \mathbf{B}_{ij,1}^T\quad \dots\quad \mathbf{B}_{ij,N-3}^T\quad \mathbf{B}_{ij,N-2}^T\quad \mathbf{0}_{L\times M}\right]^T & \text{, if } 0\le e_{ij}\le r_0-1,\\ \left[\mathbf{B}_{ij,0}^T\quad \mathbf{B}_{ij,1}^T\quad \dots\quad \mathbf{B}_{ij,N-3}^T\quad \mathbf{0}_{L\times M}\quad \mathbf{0}_{L\times M}\right]^T & \text{, if } r_0\le e_{ij}\le K-1,\\ \left[\mathbf{0}_{L\times M}\quad \mathbf{B}_{ij,1}^T\quad \dots\quad \mathbf{B}_{ij,N-3}^T\quad \mathbf{B}_{ij,N-2}^T\quad \mathbf{B}_{ij,N-1}^T\right]^T & \text{, if } 0\le K+e_{ij}\le r_0-1,\\ \left[\mathbf{0}_{L\times M}\quad \mathbf{B}_{ij,1}^T\quad \dots\quad \mathbf{B}_{ij,N-3}^T\quad \mathbf{B}_{ij,N-2}^T\quad \mathbf{0}_{L\times M}\right]^T & \text{, if } r_0\le K+e_{ij}\le K-1.\end{cases} \tag{22}$$

Where $N=\left\lfloor (N_1+N_2-2)/K\right\rfloor+2$, $r_0=(N_1+N_2-2)\bmod K$ and N_1, N_2 the lengths of the synthesis and analysis filters. $\mathbf{0}_{L\times M}$ denotes the $L\times M$ null matrix. It must be noticed that the third case in (22) exists only if $K+e_{\min}\le r_0-1$, i.e. $r_0\ge M+L$.

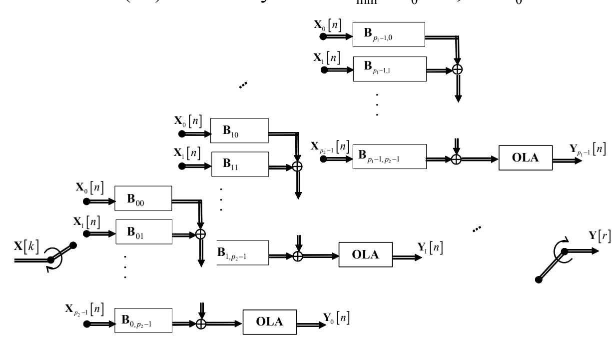


Fig. 6. Lapped Transform based implementation.

From the structure of figure 5, the figure 6 structure can be derived using lapped transforms of matrices $\mathbf{B}_{ij}$. Each new vector $\mathbf{X}[k]$ is directed to the transform matrices $\mathbf{B}_{ij}$, $0\le i\le p_1-1$, such as $j=k\bmod p_2$. For the i^{th} LPTV sub-system, the transformed vectors are summed for $0\le j\le p_2-1$ and an overlap and add (OLA) is performed to obtain $\mathbf{Y}_i[n]$. The conversion system output, $\mathbf{Y}[r]$, corresponds to the output $\mathbf{Y}_i[n]$, of the i^{th} LPTV sub-system, such as $i=r\mod p_1$. The OLA is performed on vectors of length NM with an overlapping of $(N-1)M$ elements. It is factorized for the all lapped transforms of the same LPTV sub-system.

5. FILTERING AND RE-SAMPLING COMBINED WITH CONVERSION

5.1 Conversion and Filtering Combination

In this case, a linear filtering operation by $\mathrm{S}(z)$ is performed on the synthesized signal before the new analysis bank. A similar proof (as in the annex) allows deducing that the new combined conversion/filtering system can be characterized by the same scheme of figure 3 with the new conversion matrix given by:

$$\tilde{\mathbf{T}}(z)=\left[\mathbf{v}(z)\otimes\tilde{\mathbf{g}}(z)\right]\Big|_{\downarrow K}, \tag{23}$$

where $\tilde{\mathbf{g}}(z)=\mathbf{h}(z)\mathrm{S}(z)\mathbf{f}(z)$. The only difference in the proof consists in multiplying the expression (28) of $\hat{\mathrm{X}}(z)$, by $\mathrm{S}(z)$, before the analysis bank (see annex). As in the previous sections, efficient implementation structures can be used.

5.2 Conversion and Re-sampling Combination

Here a rational re-sampling by factor Q/R, performed on the synthesized signal is considered. The conventional scheme is given in figure 7, where $\gcd(Q,R)=1$ and $\mathrm{S}_{LP}(z)$ is low-pass filter with cutoff frequency $\tilde{f}_c=\min(\pi/Q,\pi/R)$ and gain Q.

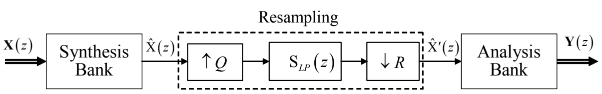


Fig. 7. Conventional combination of subband domains conversion and re-sampling.

We define $K'=\mathrm{lcm}(QL,RM)$ and $q_1=K'/RM$, $q_2=K'/QL$. The description of the new combined system is similar to sections 2-4 description by changing (K,p_1,p_2) by (K',q_1,q_2). It is characterized by the $q_1M\times q_2L$ conversion matrix:

$$\hat{\mathbf{T}}(z)=\left[\hat{\mathbf{v}}(z)\otimes\hat{\mathbf{g}}(z)\right]\Big|_{\downarrow K'}, \tag{24}$$

where $\hat{\mathbf{g}}(z)=\mathbf{h}(z^R)\mathrm{S}_{LP}(z)\mathbf{f}(z^Q)$ and $\hat{\mathbf{v}}(z)$ the matrix defined by:

$$\hat{\mathrm{v}}_{ij}(z)=z^{iRM-jQL},\ 0\le i\le q_1-1,\ 0\le j\le q_2-1. \tag{25}$$

To tune the system algorithmic delay, the same method as in section 2.2 is used, it allows replacing (25) by:

$$\hat{\mathrm{v}}_{ij}(z)=z^{aQL+bR+iRM-jQL},\ 0\le i\le q_1-1,\ 0\le j\le q_2-1. \tag{26}$$

The causality condition of $\hat{\mathbf{T}}(z)$ induces $aQL+bR\le RM-1$. The maximum advance, $c_{\max}$, to introduce in the system is such as there are solutions (a,b) in $\mathbb{Z}^2$ to the equation $aQL+bR=c_{\max}$. This occurs iff $\gcd(QL,R)$ divides $c_{\max}$. As $\gcd(Q,R)=1$, then $\gcd(QL,R)=\gcd(L,R)$. The minimal delay system is then obtained for:

$$c_{\max}=\max\left\{n\in\mathbb{N}\text{ such as } n\le RM-1 \text{ and } \gcd(L,R)\text{ divides } n\right\}. \tag{27}$$

6. CONCLUSION

By integrating the mathematical operations of synthesis and analysis banks, a generic method for conversion between different subband domain representations has been proposed to reduce algorithmic delay and enhance the parallelism degree of implementation. The computation efficiency of the proposed method compared to the conventional can be further enhanced for specific banks like cosine modulated ones by using fast algorithms as generally used. Future works should hence focus on developing fast algorithms for these specific banks. The case of time-varying filter banks has also to be studied.

7. ANNEX: PROOF OF CONVERSION MATRIX FORM

In figure 2, the output of the synthesis bank can be written as:

$$\hat{\mathrm{X}}(z)=\mathbf{f}(z)\left[\mathbf{X}(z)\right]\Big|_{\uparrow L}=\mathbf{f}(z)\mathbf{X}(z^L). \tag{28}$$

Then, after the filter bank analysis, we have:

$$\mathbf{Y}(z)=\left[\mathbf{h}(z)\mathbf{f}(z)\mathbf{X}(z^L)\right]\Big|_{\downarrow M}=\left[\mathbf{g}(z)\mathbf{X}(z^L)\right]\Big|_{\downarrow M}. \tag{29}$$

The p_2-order polyphase decomposition of $\mathbf{X}(z)$, results in:

$$\mathbf{X}(z)=\sum_{j=0}^{p_2-1}z^{-j}\mathbf{X}_j\left(z^{p_2}\right). \tag{30}$$

Hence, (29) becomes:

$$\mathbf{Y}(z)=\sum_{j=0}^{p_2-1}\left[z^{-jL}\mathbf{g}(z)\mathbf{X}_j\left(z^{p_2L}\right)\right]\Big|_{\downarrow M}. \tag{31}$$

As $K=p_2L=p_1M$, then :

$$\mathbf{Y}(z)=\sum_{j=0}^{p_2-1}\left[z^{-jL}\mathbf{g}(z)\right]\Big|_{\downarrow M}\mathbf{X}_j\left(z^{p_1}\right). \tag{32}$$

In other hand, the i^{th} polyphase component corresponding to a p_1-order decomposition of $\mathbf{Y}(z)$ is given by [6]:

$$\mathbf{Y}_i(z)=\left[z^i\mathbf{Y}(z)\right]\Big|_{\downarrow p_1},\ 0\le i\le p_1-1. \tag{33}$$

By using the formula (32), we obtain:

$$\mathbf{Y}_i(z)=\sum_{j=0}^{p_2-1}\left[z^i\left[z^{-jL}\mathbf{g}(z)\right]\Big|_{\downarrow M}\mathbf{X}_j\left(z^{p_1}\right)\right]\Big|_{\downarrow p_1},\ 0\le i\le p_1-1, \tag{34}$$

i.e.

$$\mathbf{Y}_i(z)=\sum_{j=0}^{p_2-1}\left[z^{iM-jL}\mathbf{g}(z)\right]\Big|_{\downarrow p_1M}\mathbf{X}_j(z). \tag{35}$$

Finally, we deduce the following formula which implies directly the expressions (7) and (8):

$$\mathbf{Y}_i(z)=\sum_{j=0}^{p_2-1}\left[z^{iM-jL}\mathbf{g}(z)\right]\Big|_{\downarrow K}\mathbf{X}_j(z),\ 0\le i\le p_1-1. \tag{36}$$

8. REFERENCES

[1] "Special issue on Universal Multimedia Access," *IEEE Signal Processing Magazine*, Vol. 20, No. 2, March 2003.

[2] A. N. Skodras "Direct Transform to Transform Computation" *IEEE Signal Processing Letters*, Vol. 6, No. 8, August 1999.

[3] A. Vetro, Ch. Christopoulos, and H. Sun, "Video Transcoding Architecture and Techniques: An Overview," *IEEE Signal Processing Magazine*, Vol. 20, No. 2, March 2003.

[4] J.-B. Lee and A. Eleftheriadis, "2-D Transform-Domain Resolution Translation," *IEEE Trans. on Circuit and Systems for Video Technology*, Vol. 10, No. 5, August 2000.

[5] M. M. Goodwin, "Efficient System and Method for Converting between Different Transform-Domain Signal Representations," U.S Patent 2003/0093282, May 2003.

[6] P. P. Vaidyanathan, *Multirate Systems and Filter Banks.* Prentice Hall, Englewood Cliffs, NJ, 1993.

[7] R. Sande, A. Balasubramanian, "An Efficient VLSI/FPGA Architecture for Combining an Analysis Filter Bank following a Synthesis Filter Bank," *IEEE International Symposium on Circuits and Systems*, Vancouver, BC, Canada, May 2004.

[8] R. F. Gavin and M. V. Woodward, "Method of and Apparatus for Converting an Audio Signal between Data Compression Formats," U.S Patent 2003/014241, January 2003.