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Cloud Transmission: System Performance and Application Scenarios

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***Abstract*— Cloud Transmission (Cloud Txn) System is a flexible multi-layer system that uses spectrum overlay technology to simultaneously deliver multiple program streams with different characteristics and robustness for different services (mobile TV, HDTV and UHDTV) in one RF channel. The transmitted signal is formed by superimposing a number of independent signals at desired power levels, to form a multi-layered signal. The signals of different layers can have different coding, bit rate, and robustness. For the top layer, system parameters are chosen to provide very robust transmission that can be used for high speed mobile broadcasting service to portable devices. The bit rate is traded for more powerful error correction coding and robustness so that the Signal to Noise Ratio (SNR) threshold at the receiver is a negative value in the range of -2 to -3 dB. The top layer is designed to withstand combined noise, co-channel interference and multipath distortion power levels higher than the desired signal power. The lower-layer signal can be DVB-T2 signal or other newly designed system to deliver HDTV/UHDTV to fixed receivers. The system concept is open to technological advances that might come in the future: all new technologies, BICM/Non Uuniform-QAM, rotated constellations, Time Frequency Slicing or MIMO techniques can be implemented in the Cloud Txn lower (high data) rate layer. The main focus of this paper is to thoroughly describe the performance of this newly presented Cloud Transmission broadcasting system.**

***Index Terms*—Cloud Transmission, LDPC, MBMS, Single Frequency Network, Spectrum Re-Use Friendly System, Terrestrial Broadcasting.**

I. INTRODUCTION

EFFICIENT use of the spectrum is one of the engineering research areas that has driven more efforts during the last two decades. First with the analogue to digital transition and the adoption of standards developed during 90's, and later with the development of second generation broadcast standards during the first ten years of the XXI century, this topic has become more and more relevant to broadcasting. At the same time, other communication sectors have increased the pressure for further spectrum attributions to broadband wireless access [1], which is another catalyst that has fostered broadcasting technology developments for efficient use of spectrum.

Another trend related to the objective of higher spectral efficiency materializes on global movements towards harmonization of broadcast technologies worldwide. The consortium FOBTV (Future of Broadcast Television), founded by most of the relevant broadcast regulatory bodies in Asia, Europe and America, is the most remarkable initiative in this area [2]. In 2013, the ATSC 3.0 call for proposals has been issued [3][4] with a roadmap designed to have a finalized standard in 2015, with tight requirements for an efficient use of the broadcasting bands and better worldwide compatibility.

One of the potential candidate technologies to address the highlighted challenges faced by the next generation broadcasting system is a technique called Cloud Transmission (Cloud Txn) proposed in [5], which employs a flexible ultra-robust coding and modulation scheme based on LDPC codes.

The Cloud Txn system allows the delivery of multiple layers on the same broadcast channel (spectrum overlay), where each layer is associated with its own injection power level, and lower-layer signals are recovered by means of signal cancellation techniques. This feature provides a wide range of possibilities for flexible use of the RF channel, enabling the broadcaster to mix different services with independent and differentiated robustness. Inserting a second data stream below a desired signal has been implemented before in the legacy ATSC DTV system [6][7], which is called hierarchical spectrum re-use or spectrum overlay technique. One of the beauties of the Cloud Transmission system is the implementation simplicity. The additional computation power requirements for the second layer are OFDM mapping and subtraction.

The use of hierarchical structure for delivering multiple streams is not new in broadcasting and has been proposed previously. Nevertheless, none of the existing proposals allows all streams (layers) to transmit using 100% of the time

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and 100% of the RF channel bandwidth. In comparison to Time Division Multiplex (TDM) system (ATSC mobile), frequency division multiplex (FDM) system (ISDB-T), or combined TDM and FDM system (DVB-T2), which either transmit data in part of the time or part of the RF channel bandwidth, the Cloud Txn system has the advantage on the total aggregated data rate and better time-frequency diversity.

The spectrum efficiency of the Cloud Txn broadcasting system depends to a great extent on the degree of robustness against co-channel interference and noise, especially for the top layer signal. It needs to perform well at very low SNR conditions, even in the negative SNR range. Lately, the LDPC codes have drawn a lot of attention due to their Shannon-limit-approaching performance codes over AWGN channels [8], an asymptotically better performance than turbo codes, parallelizable decoding, self-error-detection capability by syndrome check, etc.

In addition to the error correction capability, the other challenge that a new generation system must face is the quality of reception for mobile receivers. In particular, a sizable interleaver is required to deal with different speeds which may range from low mobility scenarios, corresponding to pedestrian users, to very fast time-varying scenarios, such as highway reception. Among these, low mobility scenarios are the worst case due to the existence of deep fading with long duration, which can only be overcome by a long time interleaver [9].

This paper presents detailed performance evaluation results of the Cloud Transmission System and it is organized as follows. In Section II the Cloud Txn error correction structure is presented. Then, Section III explains the main concepts related to the hierarchical spectrum reuse and Section IV discusses the receiver implementation aspects. Section V presents the results obtained from the simulations that have been carried out for the system evaluation and Section VI reports the results obtained from system feasibility tests. Section VII presents the main conclusions of this paper.

II. Error Correction

Cloud Txn system proposes the use of a two-dimensional LDPC-RS error correction code structure to provide extremely robust detection performance (Fig. 1). A newly designed quarter-rate QC-LDPC code for the cloud transmission system was introduced in [10], which is a raptor-like rate compatible LDPC code. One of the main features of this code is that it can be easily shortened from R = 1/4 code to higher rate codes, while keeping relatively good performance. For instance, by truncating 50% and 83.3% of the quarter-rate mother Parity Check Matrix (PCM), rate R = 1/3 and 1/2 codes can be easily formed with reduced decoding complexity 28% and 56%, respectively. It should be noted that usually 80% of the broadcasting coverage areas could have SNR values 5 dB above the minimum required threshold. This means that 80% of locations do not need full error correction capabilities that are designed for the lowest SNR, where the receivers can take advantage of the shortening capability of the LDPC code to achieve better power efficiency, i.e., longer battery life. Furthermore, as it has been specially designed to work under very low SNR scenarios, it outperforms the DVB-T2/S2 LDPC codes at low coding rate range.

In Fig. 1, both LDPC and RS code are linear systematic codes, where the rate-1/4 LDPC encoding is performed vertically and the RS code implemented horizontally. The RS code rate should be in the range of 1% to 10%. The advantage of a 2-dimensional error correction structure is that it is equivalent to a concatenated error correction code so that the RS code can eliminate the possible error floor created by LDPC code. In [11] the authors explained in depth all the concepts mentioned in this section.

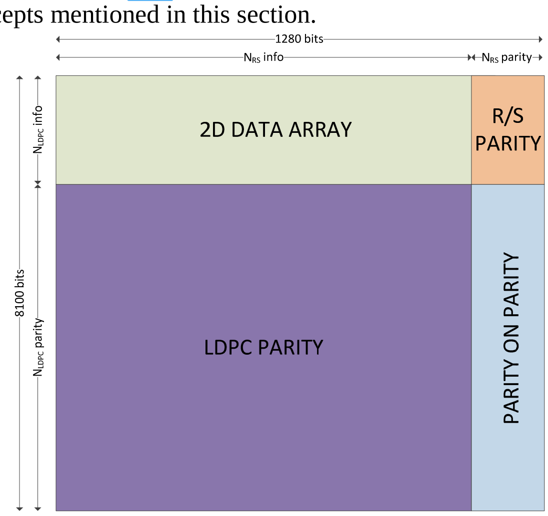

Fig. 1. A 2-Dimensional LDPC-RS code for Cloud Transmission

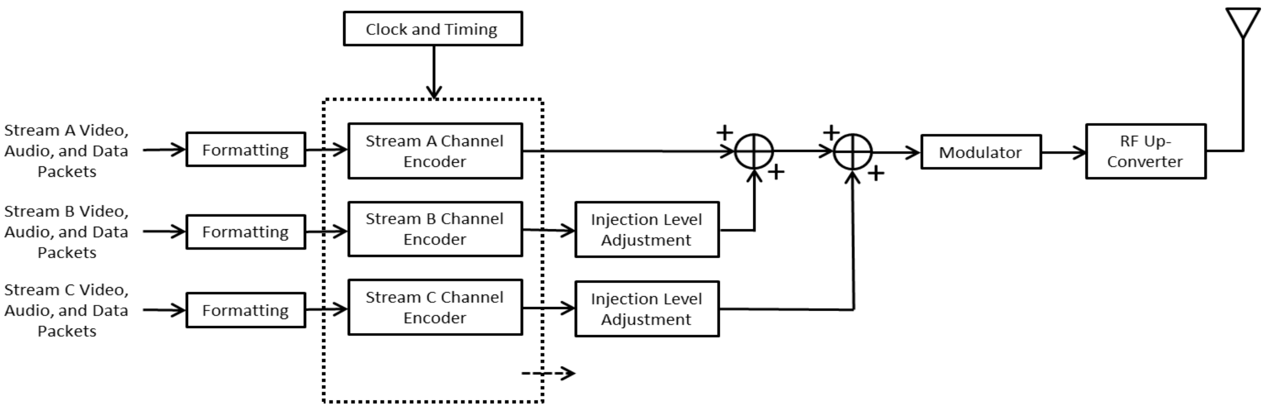

Fig. 2. Multilayer Hierarchical System transmitter.

III. HIERARCHICAL SPECTRUM REUSE

A straightforward limitation of a very robust system working on negative SNR values is the net bitrate capacity. Because of the robustness provided by the product code, the Cloud Txn system allows the use of hierarchical spectrum re-use, or spectrum overlay technology. With this approach, it is possible to inject on the same channel a second digital stream (Stream B), where Stream B could be a DVB-T2 signal or some other signal formats [12]. It should be noted that in principle, there is not any restriction for the second layer choice. Nevertheless, if the second layer is based on OFDM, with the same FFT size, symbol period and pilot pattern as the high-layer, the receiver implementation will simplify significantly. In this work, it is assumed that the second layer (Stream B) is DVB-T2 signal, which has the same RF channel bandwidth, is frequency locked, and clock synchronized with the upper layer Cloud Txn signal (Stream A).

The spectrum efficiency of the Stream B can be around 2 to 4 bit/s/Hz with an SNR threshold of 6-14 dB depending on the selected DVB-T2 mode [13]. The combined multi-layer system spectrum efficiency will be about 2.5 - 4.5 bit/s/Hz. For a 6 MHz TV band, the total expected data rates are in the range of 15 to 33 Mbps, with about 2-3 Mbps very robust data for mobile service and the rest for fixed multiple HDTV services or even UHDTV-4k service if HEVC coders are used [14]. It should be mentioned that injection levels between data streams are flexible, as well as the modulation and channel coding applied on each data stream for different reception robustness requirements. Fig. 2 shows a Cloud Transmission system diagram. At the transmitter, the signals of different streams are superimposed with specific injection levels, after being separately formatted and encoded. A third data Stream C can be further injected at e.g., 5 dB, below the Stream B. In this case, Stream C has also the same RF channel bandwidth as that of the other streams (A and B), and is frequency locked and clock synchronized with the other layers.

A. Operation Modes and System Capacity

The system capacity is a function of the channel bandwidth (8, 7 or 6 MHz depending on the band and ITU-R Region). The bitrate will be a function of the number of layers and the configuration of each layer (mode). If the lower layer of the system is based on DVB-T2, we can assume the system parameters and associated values as described in Table II, while the Cloud Txn layer performance is depicted in Table I.

Three channel models are considered: Additive White Gaussian Noise (AWGN) channel; Single 0 dB Echo channel at 90% of Guard Interval delay; and TU-6 channel [15]. In all cases, perfect channel estimation is assumed. For multipath channels (0 dB echo channel and TU-6) there will be an additional SNR degradation in the range from 0.5 to 1 dB. This margin will account for non-ideal channel estimation, implementation margin, and other distortions, as stated on DVB-T2 implementation guidelines [16]. It should be noted that the code rate stands only for the LDPC code rate, while all the minimum receiving thresholds are for a bit error rate (BER) of 10^{-7} at the product decoder output.

SNR thresholds in Tables I and II apply to each signal independently. If the SNR values are referenced to the whole signal (upper plus lower layer), the upper layer threshold should be corrected with noise degradation created by the second layer. Also, the lower layer threshold should be corrected with the cancellation noise and the injection level as described in next subsection. In Table I, the Cloud Txn upper layer is used with the Pilot Patter PP2 as defined for DVB-T2, a Guard Interval ratio of 1/32, and a 6 MHz channel. In Table II, the DVB-T2 system is configured with Guard Interval ratio of 1/128, and the Pilot Pattern PP7.

The optimal combination of parameters will strongly depend on the use case and type of service being delivered. At this point, a configuration with two layers is proposed as an example of the potential application of the system, with one layer targeting very robust reception (i.e. indoor/outdoor portable, mobile), and a second layer conveying HD services to fixed receivers. The C/N requirement is close to –2 dB for the upper layer and ranges from 6 to 25 dB for the lower layer.

TABLE I
CLOUD TXN OR MOBILE LAYER (UPPER LAYER)

Modulation	Code Rate	Channel	$(C/N)_{min}$	Data Rate
QPSK	R=1/4	AWGN	-3.4 dB	2.3 Mbps
		0 dB Echo	-2.8 dB	
		TU-6 (150 Hz)	-2.2 dB	
	R=1/3	AWGN	-1.6 dB	3.0 Mbps
		0 dB Echo	0 dB	
		TU-6 (150 Hz)	-0.5 dB	
	R = 1/2	AWGN	0 dB	4.5 Mbps
		0 dB Echo	1.5 dB	
		TU-6 (150 Hz)	1.0 dB	

TABLE II
HIGH DATA RATE LAYER (LOWER LAYER)

Modulation	Code Rate	Channel	(C/N)min	Data Rate
16QAM	R = 1/2	AWGN	6.2 dB	11.2 Mbps
		0 dB Echo	10.9 dB	
	R = 3/5	AWGN	7.6 dB	13.5 Mbps
		0 dB Echo	12.7 dB	
	R = 2/3	AWGN	8.9 dB	15.0 Mbps
		0 dB Echo	14.4 dB	
64QAM	R = 1/2	AWGN	10.5 dB	16.8 Mbps
		0 dB Echo	16.0 dB	
	R = 3/5	AWGN	12.3 dB	20.2 Mbps
		0 dB Echo	18.0 dB	
	R = 2/3	AWGN	13.6 dB	22.5 Mbps
		0 dB Echo	19.7 dB	
256QAM	R = 1/2	AWGN	14.4 dB	22.5 Mbps
	R = 3/5	AWGN	16.7 dB	27.0 Mbps
	R = 2/3	AWGN	18.1 dB	30.1 Mbps

B. Injection Levels and Inter-Layer Interference

When the system is working in a multilayer hierarchical transmission, with two or more layers transmitted within the same RF channel, inter-layer interferences will appear. The lower layer signal will act as interference to the upper layer, which will reduce its noise tolerance capacity. Meanwhile, assuming a fixed total transmission power, adding the lower layer signal will also reduce the transmission power of the higher layer. Therefore, there is a two-fold impact from the lower layer signal to the upper layer signal: reducing the transmission power and acting as noise interference.

Usually, the SNR is calculated referenced to the total received signal power. The correction factor K that accounts for the noise injected by the lower layer is given by (1), and the new carrier to noise ratio is given by (2), where Δ is the injection level:

$$K(dB) = 10\log_{10}\left[1-10^{\frac{\left(\left(\frac{C}{N}\right)_{UL}^{min\_theor}+\Delta\right)}{10}}\right] \qquad (1)$$

$$\left(\frac{C}{N}\right)_{UL}^{min\_real} = \left(\frac{C}{N}\right)_{UL}^{min\_theor} - K \qquad (2)$$

The overall power is now reduced also as a function of the injection difference (Δ). Assuming the upper layer SNR is -3 dB, and the injection level is 5 dB, the total signal power can be calculated as 1.2 dB above upper layer signal (see Fig. 3). This is the total received signal power and should be the 0 dB power level reference in the receiver SNR calculation. The effective noise level for the upper layer system will be the upper layer noise threshold minus the lower layer injection level. Therefore, the effective SNR for the upper layer system referenced to the total received signal power will be the total signal power minus the effective noise power.

The required lower layer SNR in an overlay configuration ($SNR_{LL\_OVERLAY}$) is calculated as:

$$SNR_{LL\_OVERLAY} = SNR_{LL} - \Delta - C \qquad (3)$$

where SNR_{LL} is the original lower layer signal SNR (standalone SNR, without any Cloud Txn configuration as in Table II), Δ is the injection level, and C is the power correction factor due to the fact that the transmitter distributes the nominal output power between the Upper and Lower Layers. Table III provides a two-layer overlay system with threshold values and capacities. Threshold SNR values are required minimum ratios considering the overall signal power.

Additionally, the upper leyer interference to the lower layer system due to cancellation errors should be considered. The level of this interference depends on the performance of the cancellation algorithm and will be developed in following sections.

C. *Comparison with Other Multi-Layered Systems*

There are other systems that have considered two components on the transmitted signal in the form of hierarchical transmission. DVB-T [16] and DVB-NGH have working modes based on hierarchical modulations, which enable two layers of the same information message to be transmitted with different robustness. Nevertheless, the Cloud Txn system has a different approach for multi-layer signal transmission if compared to the DVB family of standards.

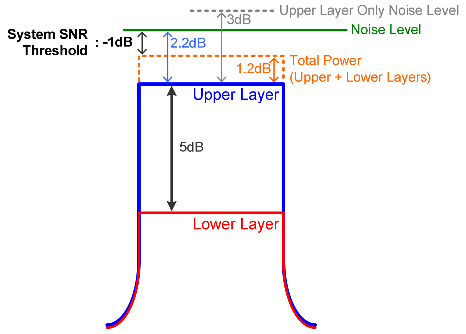

Fig. 1. Calculation of Inter-Layer Interference of a 2-Layer System

TABLE III
TWO LAYERS SYSTEM INJECTION RATIOS AND REQUIRED SNR CALCULATIONS (6 MHZ CHANNEL) AWGN CONDITIONS

Upper layer only	Injection level	UL Min. SNR	Lower layer only	LL Min. SNR
SNR = -3.4dB 2.25 Mbps R = 1/4QPSK	-3 dB	-0.5 dB	SNR=6.2 dB 11.2 Mbps R = 1/2 16QAM	11.0 dB
	-4 dB	-1 dB		11.7 dB
	-5 dB	-1.5 dB		12.4 dB
SNR = -3.4dB 2.25 Mbps R = 1/4 QPSK	-3 dB	-0.5 dB	SNR=13.4dB 22.5 Mbps R = 2/3 64QAM	18.2 dB
	-4 dB	-1 dB		18.9 dB
	-5 dB	-1.5 dB		19.6 dB
SNR = -3.4dB 2.25 Mbps R = 1/4 QPSK	-3 dB	-0.5 dB	SNR=18.1dB 30 Mbps R= 2/3 256QAM	22.9 dB
	-4 dB	-1 dB		23.6 dB
	-5 dB	-1.5 dB		24.3 dB

Unlike hierarchical modulation in DVB systems, where only QPSK modulation can be used for high priority bit stream, and only 16QAM/64QAM for low priority bit stream, the Cloud Txn allows any modulation on any layer, where modulation schemes among layers are independent. In DVB-T systems, the injection levels of different layers are fixed values, whereas in the Cloud Txn system, the injection levels are much more flexible. Furthermore, the Cloud Txn system can have more than two transmission layers, with an ultra-robust upper layer (with a negative value of SNR threshold) for mobile/pedestrian service to handheld devices.

Another option related to the DVB family could be to integrate the NGH signal within the DVBT-T2 FEF frames, in which case robust mobile services and fixed services are carried in one channel using TDM. On the other hand, the Cloud Txn system delivers mobile services and fixed services in different layers using spectrum overlay technology. The advantage is the 100% reuse of the TV channel in both time and frequency. Table IV provides a comparison of the configuration modes of DVB and Cloud Txn system.

IV. RECEIVER IMPLEMENTATION ASPECTS

A. *Receiver Complexity*

The Cloud Txn receiver is not much more complicated than

TABLE IV
CLOUD TXN VS NGH+T2

6 MHz RF Channel								
Cloud Txn			NGH 50% Time		NGH 33.3% Time		NGH 25% Time	
Upper layer	Data Rate	SNR	Data rate	SNR	Data rate	SNR	Data rate	SNR
	2.4 Mbps QPSK 1/4	-0.5 dB	2.4 Mbps QPSK 2/5	-0.2 dB	2.4 Mbps QPSK 2/3	3 dB	2.4 Mbps QPSK 4/5	5 dB
Lower layer with -4 dB injection			DVB-T2 50% Time		DVB-T2 66.7% Time		DVB-T2 75% Time	
Low-rate	11 Mbps 16QAM 1/2	10 dB	11 Mbps 64QAM 2/3	13.6 dB	11 Mbps QPSK 3/4	10 dB	11 Mbps 16QAM 3/5	9 dB
Mid-rate	17 Mbps 16QAM 3/4	14 dB	17 Mbps 256QAM 3/4	20 dB	17 Mbps 64QAM 3/4	15 dB	17 Mbps 64QAM 2/3	13.6 dB
High-rate	25 Mbps 64QAM 3/4	19 dB	19 Mbps 256QAM 5/6	22 dB	25 Mbps 256QAM 5/6	22 dB	25 Mbps 256QAM 3/4	20 dB

a regular DTV receiver. Fig. 4 displays a Cloud Txn reception diagram illustrating that many components of the receiver are shared by all layers. These include: the RF front-end (tuner), IF system and AGC, carrier recovery, time synchronization, and equalization. For an OFDM modulation system, for simplicity, all layers should use the same size of FFT, same guard interval length and same in-band pilots. On the other hand, different modulation schemes can be applied on different layers or even on different data carriers in the same layer. The Physical Layer Pipe (PLP) concept used in the DVB-T2 system can also be applied on each layer. Actually, the multi-layer approach is equivalent to a layered PLP.

For a Cloud Txn receiver that is designed to receive only the mobile (top) layer signal, the receiver system can be really simple. Only Stream A decoder is required, without the need of other stream decoders and re-modulations (Fig. 4). This single-layer receiver is very simple, energy efficient and can be easily integrated into portable and handheld devices. On the other hand, for a Cloud Txn receiver that can decode the high-data rate lower layer, the first step is to correctly decode the upper layer, re-modulate the decoded data, and then cancel it from the received signal. Once the upper layer has been removed, the decoding of the second layer signal can proceed.

From Fig. 4 it can be seen that for each additional layer decoding capability, a re-modulation/cancellation path and a decoding block is needed; while, the equalization and synchronization blocks will work for all layers.

The accuracy of the signal cancellation process is closely related to the channel estimation accuracy.

B. *Channel Estimation*

Channel estimation is critical for signal detection of the Cloud Txn system, firstly to decode the upper layer under very challenging conditions, and afterwards to perform accurate signal cancellation.

1) Pilot Aided Channel Estimation

To decode the Cloud Txn signal (upper-layer), pilot-aided channel estimation is performed using the in-band pilots. For reference and simulation purposes, the DVB-T2 scattered pilot pattern PP2 is used (see Fig. 5). The pilot spacing is 1/12 and shifted by 6 sub-carriers over two OFDM symbols. The pilots are spread in both time and frequency-domain to cope with both time-selectivity and frequency-selectivity of the wireless mobile channels.

This configuration is appropriate for channel estimation in mobile conditions. Channel estimation for the pilot structure shown in Fig. 5 can be efficiently performed with a

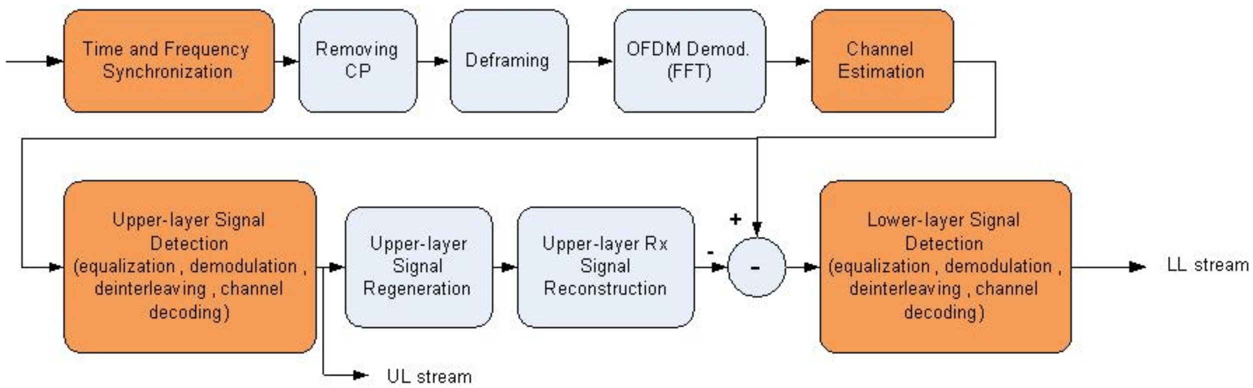

Fig. 2. Signal Detection of Multiple-Layer Cloud Txn System.

concatenation of frequency-domain channel estimation and time-domain noise filtering.

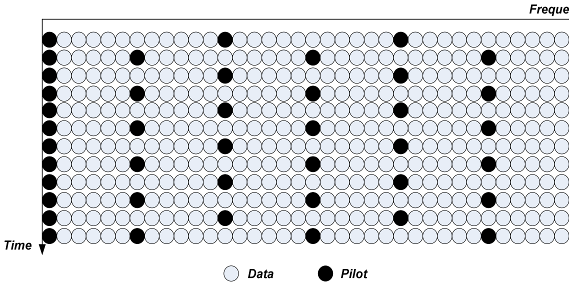

Fig. 3. Pilot pattern distribution in both time and frequency domain

In order to perform frequency-domain channel estimation, the receiver first obtains the Least Square (LS) estimates on the pilots:

$$\tilde{H}_P\left(n,k\right)=\frac{Y_P\left(n,k\right)}{X_P\left(n,k\right)} \tag{4}$$

where $X_p(n, k)$ and $Y_p(n, k)$ are the transmitted and received pilot symbols in the kth sub-carrier of the nth OFDM symbol. With the channel estimation on the pilot sub-carriers, frequency-domain interpolation is performed to obtain the channel estimations on the data sub-carriers. This can be performed using many different interpolation techniques proposed in the literature [17]. Some popular interpolation techniques are: linear, cubic-spline and DFT-Interpolation [18].

Linear interpolation has the lowest complexity, but it provides the poorest performance for channels with high frequency-selectivity, i.e., channels with large delay spread. For such channels, cubic-spline interpolation is more accurate and the most commonly used piecewise-polynomial interpolation method, with reasonable complexity.

The good performance of cubic-spline interpolation in frequency-selective channels is achieved by approximating the channel frequency response as a third-order polynomial. However, for very challenging channels with very large delay spread, such as 0 dB single-echo channels or SFN channels, this approximation is no longer accurate and therefore could generate large estimation error. This effect was confirmed by simulation results. Nevertheless, a noise-filtering can be used to reduce the estimation noise generated from the cubic-spline interpolator, assuming that the channel delay spread is shorter than the OFDM guard interval.

Finally, DFT-Interpolation performs an accurate interpolation using the *sinc()* function without making any assumptions on the frequency-domain channel response. To perform DFT-Interpolation, the time-domain channel response is first obtained by an IDFT operation as,

$$\tilde{h}_P = IDFT\left\{\tilde{H}_P\right\} \tag{5}$$

where $\tilde{h}_P$ is a vector of length M_P, M_P being the number of pilots in one OFDM symbol.

The interpolation is performed as:

$$\begin{gathered}\tilde{h}=[\tilde{h}_P\ 0\ 0\cdots 0]\\ and\\ \tilde{H}=DFT\left\{\tilde{h}\right\}\end{gathered} \tag{6}$$

where $\tilde{h}_P$ is expanded to a N-length vector, $\tilde{h}$, by appending zeros and perform an N-point FFT. When the channel delay spread is smaller than the number of pilots, the DFT-Interpolation provides accurate interpolation.

Up to now, the explained channel estimation methods use the LS channel estimate on the pilot sub-carriers obtained using(4). More accurate estimates on the pilot sub-carriers can be obtained using the MMSE estimator described in [19]. However, this requires much higher complexity. For the considered Cloud Txn system parameters, it will be shown from the simulation results that using the LS channel estimates on the pilots given by (4) already provides performance very close to limit, even for very challenging channel conditions. This suggests that using the highly complicated MMSE estimator on pilots is not necessary at least for the considered system parameters.

2) *Time-Domain Wiener Filtering*

With the channel estimates obtained by frequency-domain channel estimation, time-domain Wiener filtering can be used to further improve the channel estimation accuracy. For the kth sub-carrier in the nth OFDM symbol, a 2A-tap time-domain Wiener interpolator is performed as,

$$\hat{H}(n,k)=\sum_{m=-A,m\neq 0}^{A} u_m\cdot\tilde{H}(n-m,k) \tag{7}$$

where u_m are the Wiener filter coefficients. Coefficients of the Wiener filter in (7) are calculated as,

$$u=[u_{-A},u_{-A+1},\cdots,u_{-1},u_1,\cdots,u_A]=R^{-1}\cdot p \tag{8}$$

and R is the time-domain correlation matrix for the fading process (of this sub-carrier) whose entries are given by,

$$\begin{aligned}\left[R\right]_{m,n}&=E\left\{H\left(m,k\right)H^*\left(n,k\right)\right\}+\sigma^2\delta_{m,n}\\ &=R_H\left(0,m-n\right)+\sigma^2\cdot\delta\left(m-n\right)\end{aligned} \tag{9}$$

In *(9)*, $R_H(\Delta n,\Delta k)$ is the space-time space-frequency correlation function of the mobile channel, σ^2 is the noise variance, and $-A \le m, n \le A$.

The vector p in (8) is calculated as,

$$p=\left[R_H\left(A,0\right),\ \cdots\ ,R_H\left(1,0\right),R_H\left(-1,0\right),\ \cdots\ ,R_H\left(-A,0\right)\right] \tag{10}$$

and the 2D correlation function, $R_H(\Delta n,\Delta k)$, is defined as,

$$R_H\left(\Delta n,\Delta k\right)=E\left\{H\left(n+\Delta n,k+\Delta k\right)H^*\left(n,k\right)\right\} \tag{11}$$

where n is the time-index (OFDM symbol index) and k is the frequency index (sub-carrier index).

For mobile channels with a classic U-shape Doppler spectrum, the 2D correlation function is calculated as,

$$C(\Delta n,\Delta k)=\frac{J_0(2\pi f_d \Delta_n T_s/N)}{1+j2\pi\Delta_k f_u \sigma_d} \quad (12)$$

where σ_d is the delay spread (in second), f_d is the maximum Doppler shift, T_s is the OFDM symbol duration, and f_u is the OFDM sub-carrier spacing.

3) Decision-Directed Channel estimation

As it will be seen in the next section, to achieve good signal cancellation, it is critical to obtain accurate estimate of the channel gain, *H(k)*. To decode the upper layer signal, an estimate of the channel gain *H(k)* has already been obtained with pilot-aided channel estimation. The accuracy of this channel estimate is sufficient for the upper layer signal detection due to the strong error correction coding. However, the knowledge of the transmitted upper layer signal at the cancellation stages allows implementing the decision-directed (DD) channel estimation techniques to pursue good signal cancellation performance.

For each OFDM symbol, the receiver first obtains the LS channel estimates on each sub-carrier as,

$$\tilde{H}(n,k)=\frac{Y(n,k)}{X_U(n,k)} \quad (13)$$

With the LS channel estimation, more accurate estimates can be obtained by applying different frequency-domain smoothing filters, including but not limited to, the MMSE and SVD algorithms, the DFT-filtering, and the Wiener filtering[17]-[19].

Among these techniques, the decision-directed MMSE channel estimator is very complex to implement, since a matrix multiplication is required to decode each OFDM symbol. DFT-filtering and Wiener filtering are both practical techniques with relative low complexity.

To further reduce the estimation noise, the output of the frequency-domain channel estimator can be processed by a time-domain Wiener filtering as in pilot-aided (PA) methods.

C. Equalization

For the proposed Cloud Txn system with OFDM modulation, since the frequency-domain signal in the sub-carriers is regular modulus signal, a zero-forcing single-tap equalizer should provide very good performance.

D. Doppler Noise

Doppler effects impact the mobile reception, especially when the receiver is moving at high speed. In OFDM systems, Doppler effects cause Inter-Carrier-Interference (ICI) [20]-[23], which could severely degrade the reception performance. Therefore, there is a trend to use small size of FFT in the OFDM modulation (large subcarrier spacing) to reduce the impact of Doppler effects in mobile applications, while using larger size FFT for fixed reception systems.

Large sized FFT will lead to smaller OFDM sub-carrier (or sub-channel) spacing, which is more sensitive to ICI caused by Doppler effects. However, our results show that the proposed Cloud Transmission Layer is very robust to the Doppler effects.

Fig. 6 provides the BER performance under TU-6 channel [24][25] with different receiver FFT sizes, 2k to 16k. The impacts due to FFT size difference are negligible, assuming that system synchronization can be maintained. Fig. 7 depicts the ICI component of an OFDM modulated signal for a Doppler rate of 150 Hz in a TU-6 channel with FFT sizes of 2k, 4k, 8k and 16k. The transmitted signal has been coded using a R = 1/4 LDPC matrix (QPSK modulation, GI = 1/16, PP2), and the TU-6 channel noise injection level is set at SNR = -1 dB (or 1.2 dB below the TU-6 threshold of -2.2 dB).

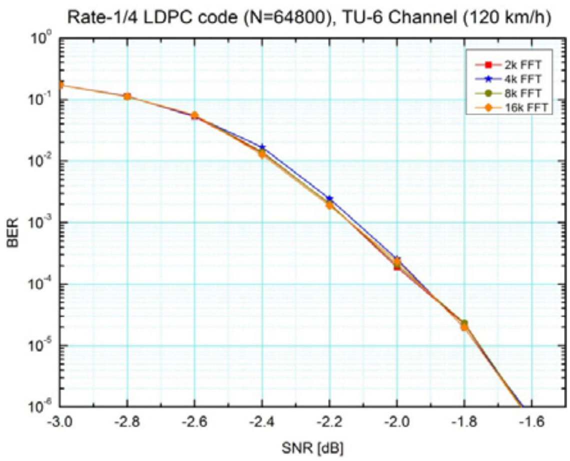

Fig. 4. Cloud Txn Layer BER for TU-6 Channel (150 Hz Doppler Rate) with Difference FFT Sizes.

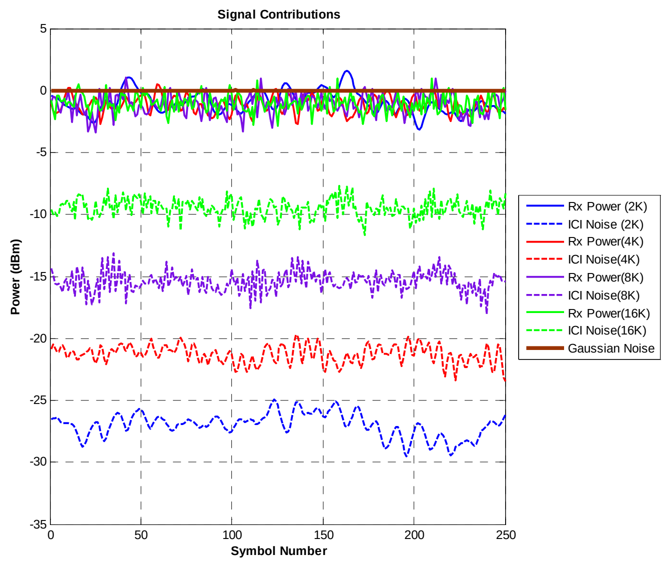

Fig. 5. OFDM Inter-Carrier Interference noise (FFT 2k, 4k, 8k and 16k, 150 Hz Doppler).

It can be noted that the relationship between the FFT size and the dynamic range of the ICI variability over consecutive symbols is inversely proportional. In addition, the ICI noise (blue-dashed curve) of 2k FFT system is more than 27 dB below the signal power. Comparing ICI noises for 2k, 4k, 8k and 16k FFT, there is about 6 dB ICI noise increase when doubling the FFT size [20][21]. Larger size of FFT results in closer carrier spacing, which increases the ICI. In case of the 8k FFT, ICI is about 16 dB below the signal power shown in Fig. 7, more than 18 dB below the TU-6 noise threshold. In consequence, the impact of Doppler on the receiver performance should be negligible (less than 0.07 dB). When using 16k FFT, the ICI noise, by calculation, is about 12 dB below the TU-6 noise threshold, which causes 0.26 dB degradation and is still a very small impact. (Note: 150 Hz Doppler shift is equivalent to a travelling speed of 280 km/h with 575 MHz channel (CH-31) – sufficient for High-Speed Train). To sum up, a stronger code will lead to less ICI degradation, whereas a weaker code will suffer more degradation.

Using larger sized FFT might have other advantages such as a reduced guard interval percentage and higher data throughput. Also, for the same pilot and data carrier ratio, larger FFT means that the distance between adjacent pilots is smaller in Hz. This will improve the channel estimation accuracy, which will lead to better signal cancellation and better system performance.

E. Upper Layer Cancellation: Spectrum Domain Cancellation

As shown in Fig. 5, for a multiple-layer system, to decode the lower-layer signal, the receiver needs to cancel the higher-power Cloud Txn layer (upper layer) first. The receiver first performs channel estimation and signal detection of the upper layer signal. The transmitted upper layer signal is then. Considering that the required SNR for lower layer decoding is much higher than what is required for error-free decoding of the upper layer, it is reasonable to assume that the upper layer signal can be perfectly reconstructed.

In the multiple-layer Cloud Txn system, for any OFDM symbol, the received signal in the *k*th sub-carrier can be expressed as:

$$Y(k) = H(k)\left(\Delta X_{UL}(k) + X_{LL}(k)\right) + W(k) \qquad (14)$$

where $X_U(k)$ and $X_L(k)$ are the upper and lower layer transmitted symbols in the *k*th sub-carrier, Δ is the injection level referenced to lower layer, and $H(k)$ is the channel gain.

To decode the lower layer signal, $X_L(k)$, the signal cancellation has to be applied as,

$$Y_L(k) = Y(k) - \Delta \cdot \hat{H}(k) X_{UL}(k) = X_{LL}(k) + W(k) \qquad (15)$$

where $\hat{H}(k)$ is the estimation of the channel gain.

F. System Latency

When decoding multi-layer signals, the decoding latency is an important factor to be considered. One of the disadvantages of a digital broadcasting system, in comparison to analog TV system, is the channel changing delay. Although most of the delay is caused by video decoding, extra caution is needed to control the channel decoding delay. This is especially important for a multi-layered Cloud Txn system, where delays from each layer will accumulate. For example, a mobile broadcasting system typically has a data interleaver of about 1 *sec* to be able to sustain the signal fading experienced in mobile reception environments. This means that the receiver has to wait for 1 *sec* for the interleaver to buffer up to start the decoding. After that, re-modulation should not introduce much delay. In this case, to decode the second layer signal, a Cloud Txn receiver needs to wait for 1 s*ec* for the first layer signal decoding and additional time for the second layer interleaver to buffer up.

However, it should be pointed out that the decoding SRN threshold for the second layer is likely to be at least 10 dB higher than the first layer. Under this condition, if there is sufficient SNR to support the second layer decoding, the first layer should have a 10+ dB SNR margin and should not need a strong error correction code. For example, assuming the first layer FEC coding rate R = 1/4 (i.e., 25% are information bits and 75% are parity bits), when there is 5 dB additional SNR margin in the received signal, rate-1/2 code is sufficient to decode the signal successfully. This means that only the 25% information bits and another 25% parity bits are required to form an rate-1/2 code to decode the signal. The remaining 50% parity bits are not needed. Therefore, with a specially designed interleaver structure and the Raptor-like rate compatible LDPC code [6],[26] which can be easily truncated (cut the parity bits) into higher rate code, the decoding delay can be reduced by 50% in this example.

As a summary, if there is sufficient SNR to decode the second layer signal, the first layer decoding delay can at least be cut by 50%. It should be considered that in DTV broadcast environments, within the coverage area, at least 80% of the locations will receive the signal with at least a 5 dB SNR margin. Thus, in most of the receiving locations, the channel decoding latency can be significantly reduced.

V. Simulation Results

This section provides in depth analysis of the performance of Cloud Txn system. First of all, the upper layer robustness against very challenging noise and fading environments is analyzed in terms of the BER curves. Next, performance of the previously explained signal cancellation is reported. Finally, the two layered system behavior is presented. Table V summarizes the channel estimation algorithms considered in this paper. It should be noted that these channel estimators have low complexity because they are based on cubic-spline interpolation and/or DFT process (which can be efficiently implemented using FFT).

TABLE V
Channel estimation Methods

Name	Description
FD-Cubic	Frequency-domain cubic-spline interpolation with noise-filtering
2D-Cubic	Frequency-domain cubic-spline interpolation with noise-filtering and time-domain Wiener filtering
FD-DFT	Frequency-domain DFT-interpolation
2D-DFT	Frequency-domain DFT-interpolation and time-domain Wiener filtering
DD-DFTF	Decision-directed channel estimation with DFT-filtering and Time-Domain Wiener Filtering

A. *Upper Layer Performance*

In all the figures in this subsection, the dashed lines represent the LDPC ouput BER, whereas the solid lines account for the error at the LDPC decoder input, i.e., uncoded BER. In this section, a time-domain Wiener interpolator with 10 taps is used, i.e., A=5 in (7).

1) *TU-6 Mobile Scenario*

In this section, we show the simulaton performance over TU-6 channel condition. Fig. 8 and Fig. 9 display the BER performance of Cloud Txn upper layer detection for a transmission signal with rate-1/4 LDPC code and QPSK modulation. Each figure contains curves for different channel estimators as presented in Table V. In addition to the performance curves for the four channel estimators, the performance of an ideal receiver with perfect channel knowledge is obtained and plotted in this figure. This curve provides a reference on how well the channel estimators perform when compared to an ideal receiver. Fig. 8 depicts the performance for static channels, while in Fig. 9 the performance curves for a receiver moving at 280 km/h are plotted.

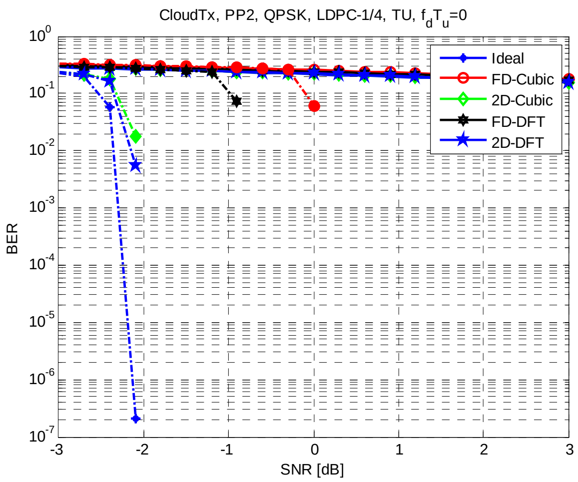

Fig. 6. Cloud Txn BER Performance under a TU-6 Channel, Stationary reception

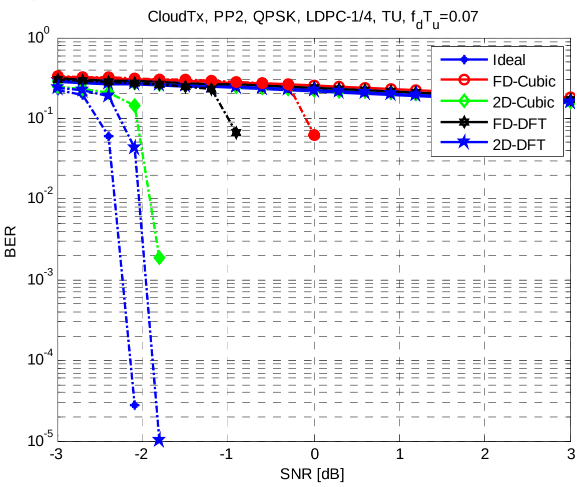

Fig. 7. Cloud Txn BER performance under a TU-6 Channel, Receiver Speed v = 280 km/h

As can be seen, both figures show that, using practical two-dimensional (2D) channel estimators (2D-DFT and 2D-Cubic), the performance is very close to that of an ideal receiver with perfect channel knowledge. The performance gap is lower than 0.5 dB. However, using only frequency-domain channel estimator suffers at least a 1.2 dB performance loss.

2) *0 dB Echo*

The 0dB echo wireless channel presents a very challenging channel condition for broadcast signal detection, especially when the echo delay is long. Simulations were performed assuming a worst-case echo channel condition with a single echo as strong as the main signal (0 dB echo), and an echo delay close to 90% of the guard interval. In addition, there is a 0.1 Hz frequency shift between them Fig. 10 shows the performance for the different channel estimation techniques for stationary receivers. Using frequency-domain DFT-Interpolation with time-domain Wiener filtering (2D-DFT), the maximum gap with respect to the ideal case is just about 0.6 dB. In Fig. 11, similar observations are made for a Cloud Txn receiver moving at 280 km/h.

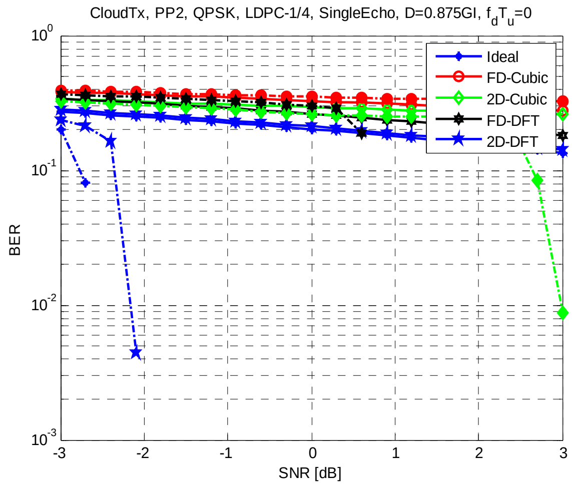

Fig. 8. Cloud Txn BER Performance, 0 dB Single Echo Channel, Stationary Reception.

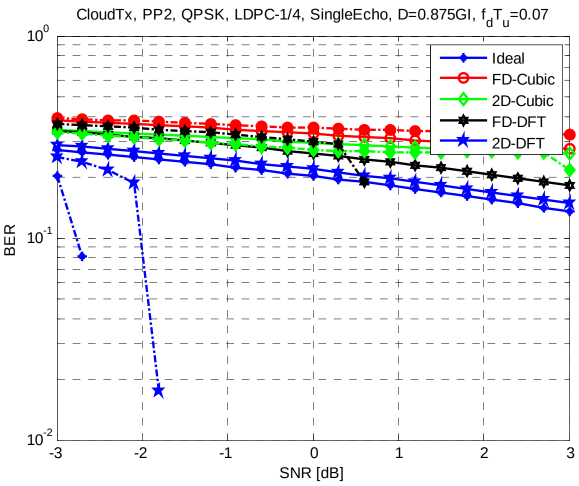

Fig. 9. Cloud Txn BER Performance, 0 dB Single Echo Channel, Receiver Speed v = 280 km/h

A slightly larger gap of 0.9 dB is observed between the performance obtained with 2D-DFT and that of the ideal receiver. It is important to keep in mind that the test channel is a worst-case scenario with both extremely strong echo and large delay. The simulation results show that with a properly designed low-complexity 2D channel estimator, the Cloud Txn transmission provides very robust performance in extremely challenging mobile fading channels.

B. Signal Cancellation Performance

In this subsection simulations are performed to evaluate the performance of signal cancellation with both pilot-aided (PA) and decision-directed (DD) channel estimation algorithms. In the simulations, a Cloud Txn system with two layers is assumed. The upper layer is the Cloud Txn signal with rate-1/4 LDPC code and QPSK modulation. A lower layer signal with 256-QAM and rate-2/3 LDPC code is injected with a power level –5dB lower than the upper layer signal.

Prior to LL decoding, the system already knows that the UL has been successfully decoded; indeed, the receiver can easily obtain the transmitted UL stream. This provides us with a chance to use another channel estimation technique: Decision Directed Channel estimation (DD). This section presents test results with both PA and DD channel estimators under the most representative channel models.

Channel estimation accuracy is analyzed using the mean square error (MSE) of the estimate, and the performance of the signal cancellation is characterized by the normalized mean square error (NMSE) referenced to the upper layer signal. The NMSE is essentially the power ratio of the cancellation residual errors referenced to the upper layer signal. Signal power is calculated as total received signal power, i.e., the main received signal power plus all multipath signal powers. In the following figures, the performance of PA based signal cancellation and that of DD based channel estimation are compared assuming an SNR of 10 dB for the LL signal.

In each figure, the upper subplot shows the NMSE versus OFDM symbol index. To simplify the simulation, a time-domain 40-tap Wiener filtering is performed over a block of 840 OFDM symbols. This causes the first and last few symbols having higher NMSE because there are not enough adjacent symbols to perform Wiener filtering. These symbols should be ignored since in reality, the time-domain Wiener filtering is performed continuously. The lower subplot shows the MSE versus sub-carrier (or sub-channel) index. It is observed that the sub-carriers close to the edges of the spectrum show higher MSE. This is due to the nature of the estimation algorithms, where the sub-carriers close to the edge have less correlation information to carry out the estimation.

Fig. 12 and Fig. 13 present the channel estimation MSE and NMSE of the residual of upper layer signal after the signal cancellation in 0 dB single echo channels with different echo delays. For the lower layer signal, stationary reception is assumed.

Fig. 12 shows the channel estimation methods for a short delay spread channel (D=1/4·GI). It is observed that 2D-Cubic performs better than 2D-DFT. Furthermore, DD-DFT provides 2 dB performance gain compared to 2D-CUBIC in terms of signal cancellation.

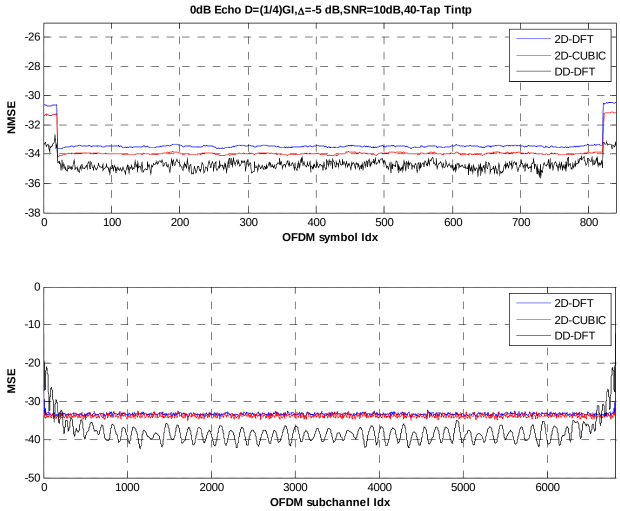

Fig. 10. Signal Cancellation Performance, 0 dB Echo (D=1/4GI), Stationary Reception, 40-tap Wiener filter, SNR_{LL}=10 dB.

In Fig. 13, we present the signal interference cancellation performance of a Cloud Txn receiver when the second transmitted signal arrives nearby the guard interval end (D=7/8·GI), which results in a channel with high frequency selectivity. It is clearly observed that for this challenging channel, the 2D-CUBIC suffers significant performance loss, whereas 2D-DFT keeps very good cancellation performance. Indeed, its performance is very close to that offered by the DD-DFT. From now on, the 2D-CUBIC method will no longer be considered, as it is not good enough to deal with the worst 0dB echo scenario.

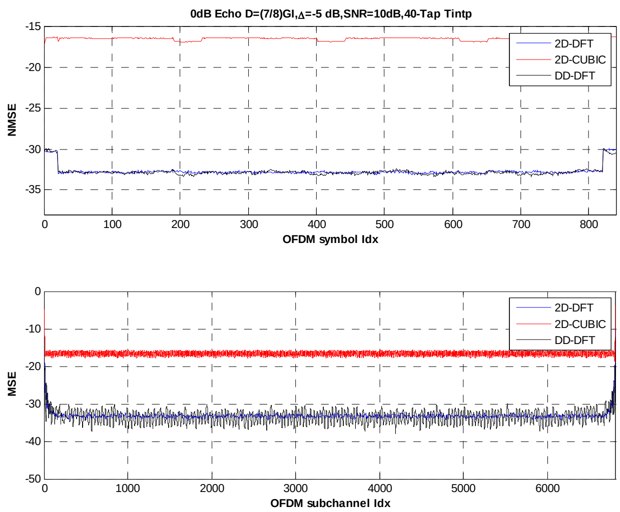

Fig. 11. Signal Cancellation Performance, 0 dB Echo (D=7/8GI), Stationary Reception, 40-tap Wiener filter, SNR_{LL}=10 dB..

An additional interesting test is the evaluation of the impact of the time-domain interpolator length on the system performance. When looking closely at the NMSE vs OFDM symbol index curves, it can be observed that the center part is 3 dB better than the two ends. This is because the interpolator on the center OFDM symbols has twice as many taps as the end symbols. For static channels, the Wiener interpolator applied to the center OFDM symbols is essentially a moving average window. For comparison, the Wiener filter length has been doubled to 80-tap for the D=7/8·GI case and performance is shown in Fig. 14. As expected, a 3 dB performance gain appears due to the use of a filter twice as long.

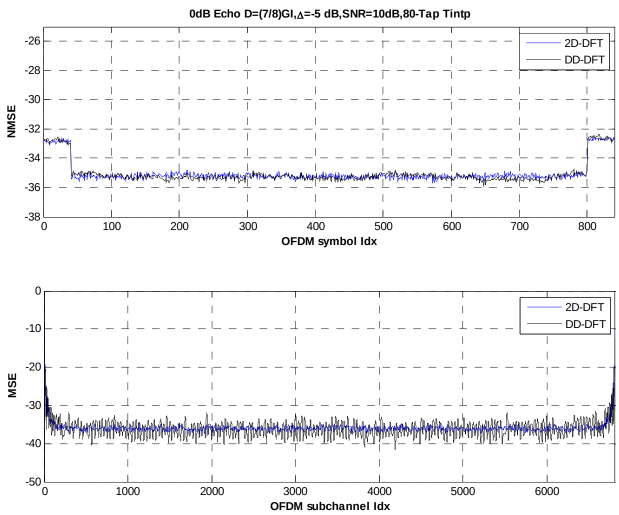

Fig. 12. Signal Cancellation Performance, 0dB Echo (D=7/8·GI), Stationary Reception, 80-tap Wiener filter, SNR_{LL}=10 dB.

Finally, the impact of the AWGN at the receiver is analyzed. When the LL signal is designed for very high data-rate, a high SNR is required for reliable detection, therefore, the PA-based channel estimation would have an advantage because the DD has a fixed SNR which is inversely-proportional to the LL signal injection level. Thus, in Fig. 15, with an SNR of 20 dB for the LL signal, the performance of the PA-based signal cancellation and that of the DD-based signal cancellation are compared. As expected, due to the higher SNR on the pilots, 2D-DFT provides better performance than DD-DFT.

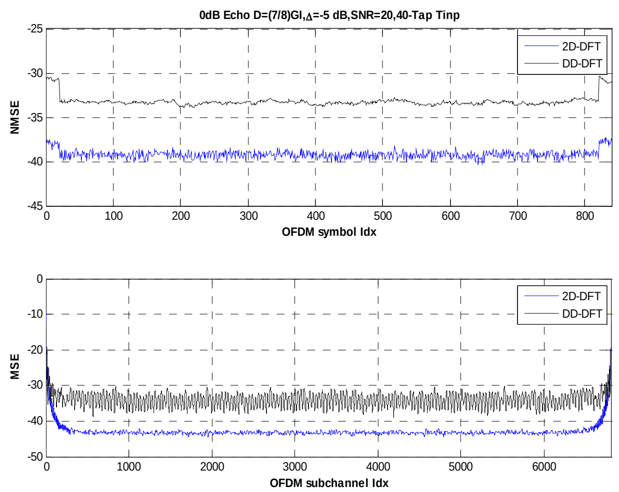

Fig. 13. Signal Cancellation Performance, 0dB Echo (D=7/8·GI), Stationary Reception, 40-tap Wiener filter, SNR_{LL}=20 dB.

It can be concluded that a long time-domain moving average window is a good solution to obtain good signal cancellation. Furthermore, pilot-aided methods have an advantage over decision directed methods when the SNR for the LL signal is high, for example, SNR=20 dB.

C. *Lower Layer System Performance: DVB-T2 Use Case*

This section provides simulation results on a two layered Cloud Txn system. In this simulation, the LL signal is a DVB-T2 signal which is configured to offer high data rate services, i.e. high modulation order and low code rates. More precisely, two different configurations for the LL have been considered: one targeting services up to 23 Mbps (64QAM, CR=2/3) and the other one up to 30.1 Mbps (256QAM, CR=2/3), which may suffice for a UHDTV service. The proposed configurations apply to a use case where the LL is delivered targeting directional roof top antennas (usually associated to AWGN and Rice channel models) [13].

The rest of the parameters are common to the upper layer. The injection factor is 5 dB, which must offer a trade-off between the possible impact of the lower layer on the upper layer performance and the inherent degradation of the lower layer due to the injection range.

1) *AWGN*

The first analyzed model is the Gaussian channel, where possible channel estimation error should be smaller. In Fig. 16 the BER performance for the different LL configurations is shown. The plot on the left represents the UL layer performance for the two LL configurations, providing nearly the same result. The figure on the right presents the performance for the two different LL services. It is important to note that the minimum receiving thresholds show consistency with the expected results presented in Section III. Indeed, there is just about a 0.4 dB loss due to the implementation cost.

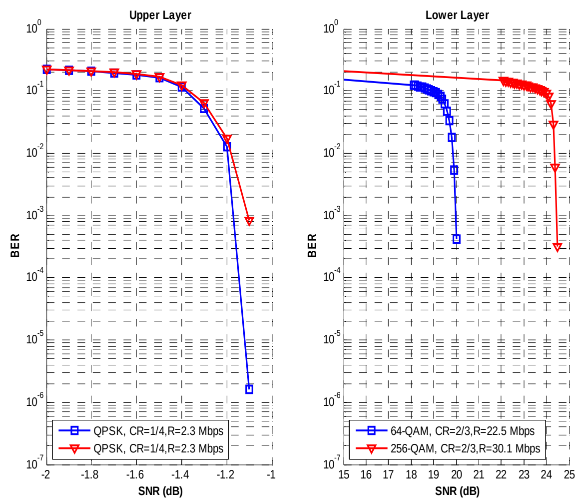

Fig. 14. Upper layer and lower layer BER performance for an AWGN channel when the spectrum overlay technique is applied.

2) RICE

In Fig. 17, the results for a Ricean channel are presented. This type of channel matches well the roof-top antenna scenario, composed of a LOS path and some weak reflected components leading to multipath energy well below the LOS contribution.

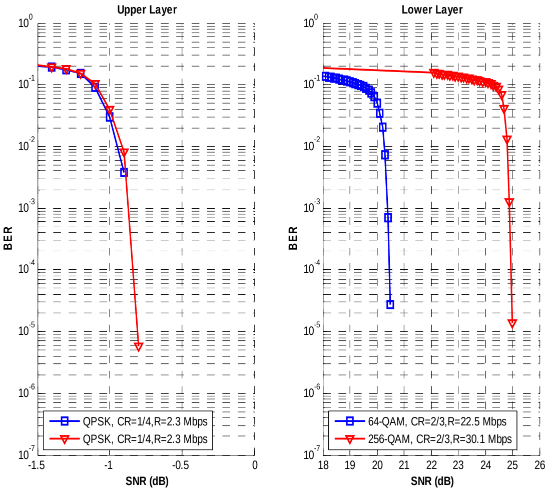

Fig. 15. Upper layer and lower layer BER performance for an RICE channel when the spectrum overlay technique is applied.

The plot on the left represents the UL layer performance for both configurations, which is 0.3 dB worse than in the AWGN scenario. Similarly, the right subplot curves, representing the LL layer performance, show the same tendency as in the AWGN case plus 0.3 dB penalty due to channel fading.

VI. System Feasibility Tests

The aim of these laboratory tests is to produce real Cloud Txn signals that can be fed into channel emulation hardware and thus to obtain system performance closer to practical scenarios using real equipment. The available results apply to a Gaussian channel for 6 MHz signals.

A. Test Methodology

The test methodology is based on a complete Cloud Txn transmission and reception system. This system is a mixture of software and hardware components. The signal is generated offline and reproduced by Vector Signal Generators that can also apply different propagation channel profiles and different white noise levels to the RF signal. The RF signal is fed using a cable to the receiver hardware. The experiments are performed using different combinations of Cloud Txn signals, propagation channels and increasing levels of noise in order to obtain a list of SNR vs BER values. These values are used to plot the BER curves and to determine the system thresholds. The receiver is built on a Vector Signal Analyzer and a DVB-T2 SDR software platform. The synchronization, carrier recovery, channel estimation and equalization algorithms were optimized for DVB-T2 and thus there might be room for further improvement [27].

The subsequent figures show results from both laboratory tests and simulations for performance comparison and evaluation of the implementation margin. The presented values have been obtained with noise injection steps of 0.1 dB in simulations and 0.25 in laboratory tests. Plots that end before converging to 10^{-7} are cases where the immediate higher SNR value provides error free transmission.

B. Cloud Txn Single Layer Results

The first results are for a Cloud Txn single layer. The system is configured in 8K mode, with QPSK, rate-¼ LDPC, and a GI of 1/32. The pilot pattern is DVB-T2 PP2, i.e., scattered pilot with 1/12 pilot/symbol in frequency-domain, with a 6 subchannel shift between adjacent OFDM symbols. This Cloud Txn layer provides a bitrate of 2.25 Mbps. Fig. 18 provides the results obtained for a Gaussian channel (AWGN).

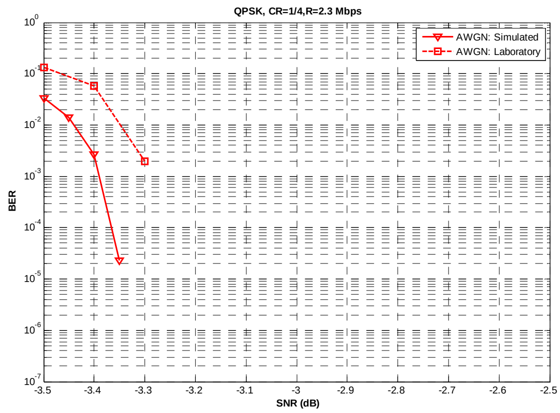

Fig. 16. Simulated vs laboratory BER Curves for a single layer system

C. Two Layer Results: Upper Layer Cloud Txn, Lower Layer DVB-T2

Following the same methodology, a two layer Cloud Txn system has been evaluated in the lab. The configuration consists of a Cloud layer signal (8K, QPSK, R=1/4, 2.25 Mbps) as the upper layer and a DVB–T2 (8K, 256QAM, R = 2/3, 30 Mbps) as the lower layer, with an injection factor (Δ) of 5 dB.

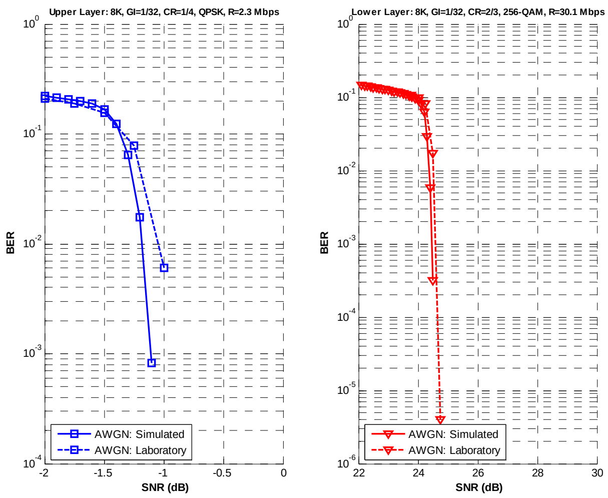

Fig. 17. Laboratory BER Curves for a two layer system (Upper layer Cloud Txn, Lower Layer DVB-T2, Injection rate 5 dB).

This configuration targets a high bit rate for the second layer, while providing the same robustness for the upper layer. Fig. 19 shows the obtained BER curves.

The laboratory results are almost identical to the simulation results. This demonstrates that performing signal cancellation algorithm to achieve multilayered Cloud Txn transmission is feasible in the realistic environments, with very reasonable complexity.

Finally, the system thresholds for the two-layer case are summarized in Table VI.

TABLE VI
PERFORMANCE SUMMARY

Name	Description	AWGN
Matlab Simulations (System threshold SNR, dB)	UL	-1.0
	LL	24.6
Laboratory Tests (System threshold SNR, dB)	UL	-1.0
	LL	25.0

VII. CONCLUSION

Cloud Transmission is proposed for future broadcasting system to achieve robust mobile reception, large mobile TV coverage and more efficient use of the spectrum. The key of this system is to use a robust error protection based on LDPC, which enables successful demodulation and decoding even with negative SNRs. High spectrum efficiency and flexibility is achieved by using a spectrum overlay technique that enables simultaneously transmitting multiple signal streams within the same broadcast channel, taking advantage of signal cancellation schemes. The cancellation schemes are based on the robustness of the LDPC code plus well known channel estimation techniques. The incremental complexity of the receiver is in this respect minimal due to the fact that most of the receiver blocks are shared by both the upper and lower layers.

This paper contains detailed results of the system performance evaluation, which are obtained from both extensive simulations and laboratory tests.

It is proven that the Cloud Txn system can accommodate scalable bit rates of more than 30 Mbps in a 6 MHz channel. The SNR requirements have been analyzed and presented. A comparison with other broadcast systems is performed and shows that the Cloud Transmission scheme provides a more efficient use of the spectrum.

Moreover, the multilayer system performance has been evaluated for different propagation channels. Specifically, results for Gaussian, Rice, Rayleigh, 0dB Echo SFN and TU6 channels have been analyzed assuming perfect channel estimation. In most of the cases, the simulated minimum SNR requirements are very close to the theoretical values.

Next, Practical channel estimation algorithms are implemented and tested for evaluating their impact on the Cloud Txn performance. The simulation results demonstrate that a properly design 2D channel estimator provides very robust performance in challenging mobile fading channels.

We also investigated frequency-domain signal cancellation technique and studied the impact of the channel estimation schemes on the cancellation performance. For small delay spreads (less than 1/4 GI), pilot-added Cubic-Spline interpolation gives good results. For delay spread values higher than 50% of the guard interval, DFT interpolation is better than Cubic-Spline. Decision Directed (DD) channel estimation is always the best in all the studied cases (3 to 4 dB better). DD can use fewer pilots, thus, it can increase the data rate by about 5% and it is worthwhile in terms of complexity reduction. For high-data rate lower layer (> 20 Mbps for UHDTV) signal which requires high SNR, since reception is likely to be fixed, time average on impulse response is an efficient method to increase the accuracy.

Finally, this paper has presented the performance of a first laboratory prototype of the Cloud Txn system, based on a mixed Software/Hardware architecture. The laboratory results for single-layer Cloud Transmission have proven that the implementation losses are lower than 0.5 dB. In addition, a multilayer Cloud Txn system has also been evaluated, with a configuration that provides a data rate close to 33 Mbps, using 256QAM for the lower layer. The results also show little loss as compared to simulations.

The system is undergoing further tests in laboratory as well we in the field to evaluate its performance in a wider variety of reception conditions.

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