**3GPP TSG-RAN WG4 Meeting#99-e R4-2110609**

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**Agenda item: 6.3.2.3.3**

**Source: ZTE Corporation**

**Title: TP to TS 38.176-1: Annex G and H: In-channel TX test**

**Document for:** **Approval**

1. Introduction

In the past RAN4#98e meetings, work split has been agreed among companies, therefore in this contribution, we want to share the draft TP for Annex G and H for In-channel Tx test for IAB.

1. Reference

[1] R4-2103856 WF on IAB conformance specification work split and drafting guidelines, approved.

1. Annex

<Start of TP>

# Annex G[(normative)]: In-channel TX tests for IAB-DU

The Annex H in TS 38.141-1 [2] applies to FR1 IAB-DU.

# Annex H[(normative)]: In-channel TX tests for IAB-MT

# H.0 Applicability

FR1 IAB-MT EVM can be determined by the process according to following alternatives:

Alternative 1: Annex E in TS 38.521-1 [23]. Only CP-OFDM waveform of PUSCH is measured for IAB-MT or

Alternative 2: from Annex H.1 to Annex H.7.

# H.1 General

The in-channel TX test enables the measurement of all relevant parameters that describe the in-channel quality of the output signal of the TX under test in a single measurement process.

The parameters describing the in-channel quality of a transmitter, however, are not necessarily independent. The algorithm chosen for description inside this annex places particular emphasis on the exclusion of all interdependencies among the parameters.

# H.2 Basic principles

The process is based on the comparison of the actual output signal of the TX under test, received by an ideal receiver, with an ideal signal, that is generated by the measuring equipment and represents an ideal error free received signal. All signals are represented as equivalent (generally complex) baseband signals.

The description below uses numbers and illustrations as examples only. These numbers are taken from a TDD frame structure with normal CP length, 30 kHz SCS and a transmission bandwidth configuration of 100 MHz (*N*RB = 273). The application of the text below, however, is not restricted to this parameter set.

## H.2.1 Output signal of the TX under test

The output signal of the TX under test is acquired by the measuring equipment and stored for further processsing. It is sampled at a sampling rate which is the product of the SCS and the *FFT size*, and it is named . The *FFT size* is determined by the transmission bandwidth in table 6.5.3.5-2 for 15 kHz SCS, table 6.5.3.5-3 for 30 kHz SCS and table 6.5.3.5-4 for 60 kHz SCS. In the time domain, it comprises at least 10 ms. It is modelled as a signal with the following parameters:

- demodulated data content,

- carrier frequency,

- amplitude and phase for each subcarrier.

For the example in the annex, the *FFT size* is 4096 based on table 6.5.3.5-3. The sampling rate of 122.88 Msps is the product of the *FFT size* and SCS.

## H.2.2 Ideal signal

Two types of ideal signals are defined:

The first ideal signal is constructed by the measuring equipment according to the relevant TX specifications, using the following parameters:

- demodulated data content,

- nominal carrier frequency,

- nominal amplitude and phase for each subcarrier.

It is represented as a sequence of samples at the sampling rate determined from annex H.2.1 in the time domain. The structure of the signal is described in the test models.

The second ideal signal is constructed by the measuring equipment according to the relevant TX specifications, using the following parameters:

- nominal demodulation reference signals (all other modulation symbols are set to 0 V),

- nominal carrier frequency,

- nominal amplitude and phase for each applicable subcarrier,

- nominal timing.

It is represented as a sequence of samples at the sampling rate determined from annex H.2.1 in the time domain.

## H.2.3 Measurement results

The measurement results, achieved by the in-channel TX test are the following:

- Carrier frequency error

- EVM

- Resource element TX power

- OFDM symbol TX power (OSTP)

Other side results are: residual amplitude- and phase response of the TX chain after equalisation.

## H.2.4 Measurement points

The resource element TX power is measured after the FFT box as described in figure H.2.4-1. The EVM shall be measured at the point after the FFT and a zero-forcing (ZF) equalizer in the receiver, as depicted for FR1 in figure H.2.4-1. The FFT window of *FFT size* samples out of (*FFT size* + cyclic prefix length) samples in the time domain is selected in the "Remove CP" box. The *FFT size* and the cyclic prefix length are obtained from table 6.5.3.5-2 for 15 kHz SCS, table 6.5.3.5-3 for 30 kHz SCS and table 6.5.3.5-4 for 60 kHz SCS.

In one subframe, there are two symbols with the length of the cyclic prefix larger than the values listed in tables 6.5.3.5-2, 6.5.3.5-3 and 6.5.3.5-4. Table H.2.4-1 lists the slot number and the symbol number and the formula how to compute the length of cyclic prefix for those two symbols according to the sampling rate.

Table H.2.4-1: Slot number and symbol number identifying the longer CP length for normal CP

|  |  |  |  |
| --- | --- | --- | --- |
| SCS (kHz) | # slots in subframe | Symbol # and slot # with longer CP | Longer CP length |
| 15 | 1 | (symbol 0, slot 0) (symbol 7, slot 0) | CP length + *FFT size* / 128 |
| 30 | 2 | (symbol 0, slot 0)(symbol 0, slot 1) | CP length + *FFT size* / 64 |
| 60 | 4 | (symbol 0, slot 0)(symbol 0, slot 2) | CP length + *FFT size* / 32 |

For the example used in the annex, the "Remove CP" box selects 4096 samples out of 4384 samples. Symbol 0 has 64 more samples in the cyclic prefix than the other 13 symbols in the slot (the longer CP length = 352).



Figure H.2.4-1: Reference point for FR1 EVM measurements

# H.3 Pre-FFT minimization process

Sample Timing, Carrier Frequencyin are varied in order to minimise the difference between and , after the amplitude ratio of and has been scaled. Best fit (minimum difference) is achieved when the RMS difference value between and is an absolute minimum.

The carrier frequency variation is the measurement result: carrier frequency error.

From the acquired samples, one value of carrier frequency error can be derived.

Note 1: The minimisation process, to derive the RF error can be supported by post-FFT operations. However the minimisation process defined in the pre-FFT domain comprises all acquired samples (i.e. it does not exclude the samples in-between the FFT widths and it does not exclude the bandwidth outside the transmission bandwidth configuration).

Note 2: The algorithm would allow to derive carrier frequency error and sample frequency error of the TX under test separately. However there are no requirements for sample frequency error. Hence the algorithm models the RF and the sample frequency commonly (not independently). It returns one error and does not distinguish between both.

After this process, the samples are called .

# H.4 Timing of the FFT window

The FFT window length is *FFT size* samples per OFDM symbol.

The position in time for the FFT shall be determined.

In an ideal signal, the FFT may start at any instant within the cyclic prefix without causing an error. The TX filter, however, reduces the window. The EVM requirements shall be met within a window *W* < CP. There are three different instants for FFT:

- Centre of the reduced window, called ,

- , and

- .

The value of EVM window length *W* is obtained from tables 6.5.3.5-2 for 15 kHz SCS, 6.5.3.5-3 for 30 kHz SCS and 6.5.3.5-4 for 60 kHz SCS and the transmission bandwidth.

The IAB-MT shall transmit a signal according to the test models intended for EVM. The demodulation reference signal of the second ideal signal shall be used to find the centre of the FFT window.

The timing of the measured signal is determined in the pre-FFT domain as follows, using and :

1. The measured signal is delay spread by the TX filter. Hence the distinct borders between the OFDM symbols and between data and CP are also spread and the timing is not obvious.

2. In the ideal signal , the timing is known.

Correlation between bullet (1) and (2) will result in a correlation peak. The meaning of the correlation peak is approximately the "impulse response" of the TX filter.

3. The meaning of "impulse response" assumes that the autocorrelation of the ideal signal is a Dirac peak and that the correlation between the ideal signal and the data in the measured signal is 0. The correlation peak, (the highest, or in case of more than one highest, the earliest) indicates the timing in the measured signal.

The number of samples used for FFT is reduced compared to . This subset of samples is called .

From the acquired samples one timing can be derived.

The timing of the centre is determined according to the cyclic prefix length of the OFDM symbols. For normal CP, there are two values for in a 1 ms period:

- = length of cylic prefix / 2,

- = Longer CP length - length of cylic prefix / 2,

Where the length of cyclic prefix is obtained from table 6.5.3.5-2 for 15 kHz SCS, table 6.5.3.5-3 for 30 kHz SCS and table 6.5.3.5-4 for 60 kHz SCS, and the longer CP length is obtained from table H.2.4-1.

As per the example values:

- = 144 within the CP of length 288 for OFDM symbols 1 to 13 of a slot,

- = 208= 352 - 144) within the CP of length 352 for OFDM symbol 0 of a slot.

# H.5 Resource element TX power

Perform FFT on with the FFT window timing . The result is called . The RE TX power (RETP) is then defined as:

Where SCS is the subcarrier spacing in Hz.

From RETP the OFDM Symbol TX power (OSTP) is derived as follows:

Where the summation accumulates RETP values of all *Nsym* OFDM symbols that carry PUSCH and not containing PUCCH, SRS or PRACH within a slot.

From the acquired samples, values for each OSTP can be obtained and averaged where for TDD, is the number of slots with uplink symbols in a 10 ms measurement interval and is computed according to the values in table 4.9.2.2-1.

For the example used in the annex, and .

# H.6 Post-FFT equalisation

Perform FFTs on , one for each OFDM symbol within 10 ms measurement interval with the FFT window timing to produce an array of samples, in the time axis *t* by *FFT size* in the frequency axis *f*.

For the example in the annex, 280 FFTs are performed on . The result is an array of samples, 280 in the time axis by 4096 in the frequency axis.

The equalizer coefficients and are determined as follows:

1. Calculate the complex ratios (amplitude and phase) of the post-FFT acquired signal and the post-FFT ideal signal for each demodulation reference signal, over 10 ms measurement interval. This process creates a set of complex ratios:

2. Perform time averaging at each demodulation reference signal subcarrier of the complex ratios, the time-averaging length is 10 ms measurement interval. Prior to the averaging of the phases an unwrap operation must be performed according to the following definition:

- The unwrap operation corrects the radian phase angles of by adding multiples of 2 \* π when absolute phase jumps between consecutive time instances are greater than or equal to the jump tolerance of π radians.

- This process creates an average amplitude and phase for each demodulation reference signal subcarrier (i.e. every second subcarrier).

and

Where *N* is the number of demodulation reference signals time-domain locations from for each demodulation reference signal subcarrier *f*.

3. The equalizer coefficients for amplitude and phase and at the demodulation reference signal subcarriers are obtained by computing the moving average in the frequency domain of the time-averaged demodulation reference signal subcarriers. The moving average window size is 19 and averaging is over the DM-RS subcarriers in the allocated RBs. For DM-RS subcarriers at or near the edge of the channel, or when the number of available DM-RS subcarriers within a set of contiguously allocated RBs is smaller than the moving average window size, the window size is reduced accordingly as per figure H.6-1.

4. Perform linear interpolation from the equalizer coefficients and to compute coefficients, for each subcarrier.

Figure H.6-1: Reference subcarrier smoothing in the frequency domain

# H.7 EVM

### H.7.0 General

For EVM create two sets of , according to the timing and , using the equalizer coefficients from H.6.

The equivalent ideal samples are calculated from (annex H.2.2) and are called .

The EVM is the difference between the ideal signal and the equalized measured signal.

Where:

- T is the set of symbols with the considered modulation scheme being active within the slot,

- is the set of subcarriers within the resource blocks with the considered modulation scheme being active in symbol *t*,

- is the ideal signal reconstructed by the measurement equipment in accordance with relevant test models,

- is the equalized signal under test.

NOTE: Although the basic unit of measurement is one slot, the equalizer is calculated over the entire 10 ms measurement interval to reduce the impact of noise in the reference signals.

## H.7.1 Averaged EVM (TDD)

EVM is averaged over all allocated uplink resource blocks with the considered modulation scheme in the frequency domain, and a minimum of slots where is the number of slots in a 10 ms measurement interval.

For TDD, let be the number of slots with uplink symbols within a 10 ms measurement interval, the averaging in the time domain can be calculated from slots of different 10 ms measurement intervals and should have a minimum of slots averaging length where is the number of slots in a 10 ms measurement interval.

- is derived by: Square the EVM results in each 10 ms measurement interval. Sum the squares, divide the sum by the number of EVM relevant locations, square-root the quotient (RMS).

- Where is the number of resource blocks with the considered modulation scheme in slot *i*.

- The is calculated, using the maximum of at the window *W* extremities. Thus is calculated using and is calculated using (*l* and *h*, low and high; where low is the timing and and high is the timing ).

- In order to unite at least slots, consider the minimum integer number of 10 ms measurement intervals, where is determined by.

and for 15 kHz SCS, for 30 kHz SCS and for 60 kHz SCS normal CP.

- Unite by RMS.

<End of TP>