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Title: **PSS/MMS High-Rate Audio Selection Test and Processing Plan Version 2.0**

Source: **TSG SA WG4**

Document for: Information

Agenda Item: 7.4.1

Contents

0 Status of document

V 0.7: Agreed version
   Remaining issues:

   Identify and/or confirm entities:
   - o Listening labs
   - o Host lab and mirror host lab
   - o Selection entity
   - o Entity providing vote collection template
   - o Global analysis lab
   - o Volunteers providing processing scripts and tools

   - o Volunteers providing processing scripts and tools
       Assignment of entities by correspondence

   List contact persons for all involved entities

0 Status of document

1   Introduction

This document contains the complete set of experimental designs for the selection phase of the PSS/MMS high-rate audio codec selection.

Various types of music material will be used in the testing process.

A summary of the experiments to be conducted can be found in table 1-1. Please note that experiments at 24 kbit/s in both stereo and mono are conducted in the course of the low-rate selection test. The selection process in the high-rate range will re-use the results obtained in those experiments.

Table 1-1: Experiments for high-rate codec selection

Exp.	Operational mode	#Codecs in test	# reference codecs	#Anchors in test	#References	#items	Total
1	32 kbps, stereo	4(use case A and use case B encoder)	2, incl. RealAudio @ 32 kbit/s stereo	2	1	12	108
2	48 kbps, stereo	4(use case A and use case B encoder)	2, incl. RealAudio @ 48 kbit/s stereo	2	1	12	108
3	32 kbps, stereo, 1% and 3% random frame loss	4	2	2	1	12	108

Section 2 provides a list of candidate and reference codecs used in the experiments.

Section 3 provides a list of reference documents and contact names for any question related to the test plan.

Section 5 gives an overview of the audio material required for the testing.

Sections 6 contains the test plans for each experiment.

The specification of the processing functions of the audio material is included in section 7

Annex A contains English language examples of instructions for the listening subjects for the MUSHRA test methodology used in the experiments.

Annex B presents the filename convention.

1.1   Responsibilities

The processing will be done by the host-lab (T-Systems), reproduced by the mirror host lab (Audio Research Labs) and verified by the candidates. The funding for the Selection and the characterization tests will be equally shared between the proponents.

T-Systems will serve as host laboratory and Audio Research Labs will serve as the mirror host laboratory. The host laboratory will have the following responsibilities:
• Receive candidate testing material after the call for the material and distribute it to the material selection entity.
• Prepare testing and training material based on the indications from the material selection entity.
• Receive executables of the candidate codecs from the codec proponents.
• Receive executables of the reference codecs.
• Generate FER pattern file after receipt of the random generator seed.
• Process reference, anchor and codec conditions (including re-sampling to sampling frequency of original material).
• Cross-check the processing between the host laboratories and the codec proponents: provide the processed material to the codec proponents who cross-check.
• Assemble the final distribution of the processed material to the listening laboratories, which includes blinding of the material.

The selection experiments for each candidate codec are run by two listening laboratories in parallel, as shown in Table 1-2.

Table 1-2: Allocation of sub-experiments to the Listening Laboratories

Exp. LL ID	Lab1 TS	Lab2 NT	Lab3 FT	Lab4 DY	Lab5 NK	Lab6 ER	Total Per Exp.	Size
1	X			X			2	
2		x			x		2	
3			x			x	2	
Totals:	1	1	1	1	1	1	6	

(Legend: T-Systems (TS), NTT-AT (NT), France Telecom R&D (FT), Dynastat (DY), Nokia (NK), Ericsson (ER)

Each listening lab will be asked to provide a full report of the experiments performed. The test results will be included in spreadsheets prepared for that purpose. Any deviations from the specifications contained in this document will be documented along with the results.

The test results will then be combined by the Global Analysis Laboratory (Audio Research Labs) and presented to SA4.

1.2   Contact Names

2   Codecs, Anchors and References

2.1   Candidate and Reference codecs

The following table provides an overview of the candidate codec participating in the PSS/MMS high-rate audio selection tests.

#	Codec name	Providing Organization(s)	Contact
1	aacPlus	Coding Technologies/ NEC	kunz@codingtechnologies.com frederic.gabin@nectech.fr
2	CT	Coding Technologies	kunz@codingtechnologies.com

The reference codecs are listed in the following table.

#	Codec name	Providing Organization(s)	Contact
3	AAC	Fraunhofer	Bernhard.grill@iis.fraunhofer.de Johannes.hilpert@iis.fraunhofer.de
4	RealAudio	RealNetworks	Greg Conklin gregc@real.com Jeremy Worley jworley@real.com

2.2   Anchors and references

Besides the items encoded with the candidate and reference codecs, anchor and reference items will be included in the tests. Their purpose is to normalize the tests and to make them more comparable across different realizations.
In the experiments, two anchors will be used with lowpass filtered original signal. Also included is the uncoded original signal, once as open and once as hidden reference.

#	Type	Specification	Channel type
1	Anchor	3.5 kHz Lowpass	Mono, stereo at 48 kbit/s
2	Anchor	7.0 kHz Lowpass	Mono, stereo at 48 kbit/s
6	Hidden Reference	Original signal	Mono/Stereo
7	Open Reference	Original signal	Mono/Stereo

3 References and Conventions

3.1 Reference Documents

[I] RECOMMENDATION ITU-R BS.1534   Method for the subjective assessment of intermediate quality level of coding systems

4   Test Material

The test material will be composed following a similar approach as in MPEG audio codec standardization. First, a call will be sent out for stereo test material according to a number of generic audio signal categories. Then, an independent selection entity <tbd> will identify 12 critical items, which are representative for assumed typical application scenarios such as music streaming, video soundtracks, sports commentary, commercials, news, out of the categories to be used in the experiments.

Original material shall always be in stereo and sampled at 48kHz.

The selection panel will identify, based on material coded by the candidate and reference codecs, a set of critical test items covering all the following generic audio signal categories with a focus on the music categories:

- Pop, with and/or without vocals
- Classic, with and/or without vocals
- Single instruments
- a capella vocals, solo and/or choir
- Mixed speech and music
- Speech with and/or without background noise

The approximate lengths in time of the items will be 10s at a maximum.

4.1   Material selection panel

The selection entity is formed by the following organizations:
- Organizations TBA

All 3GPP members are invited to submit test material to the host lab. The submitting organization shall assign the items to the above-mentioned audio signal categories. The host lab will blind the material and provide it to the material selection entity.
This ensures the selection is based on items whose origin is not revealed to the selection entity.
The host lab will further maintain and report to SA4 a list indicating the number of proposed items per submitting organization.
In case the submitted material is insufficient/inadequate to conduct the tests, the selection entity will add the missing test items.
The selection entity will provide SA4 with a report about the selection process.

4.2   Training material

Limited material will be used in the training phase in which the subjects familiarize with the testing methodology and environment.
The training will be conducted with four sound items. These items will be identified by the selection entity and shall not be re-used in the blind grading phase. The training phase shall be executed as a separate short MUSHRA session.

5   Information relevant to all Experiments

5.1   General Technical Notes

Any and all deviations from the specifications contained in this document must be documented and submitted to TSG-SA-WG4 along with the experimental results.

For all experiments, subjects should be seated in a quiet environment; 30dBA Hoth Spectrum (as defined by ITU-T, Recommendation P.800, Annex A, section A.1.1.2.2.1 Room Noise, with table A.1 and Figure A.1) measured at the head position of the subject. This will help ensure consistency between the different subjects in the same laboratory as well as across the different laboratories in which these experiments will be performed.

The test stimuli will be presented to the subjects over headphones meeting one of the following requirements:

1)   Binaural listening using closed-back, supra-aural headphones;

2)   Binaural listening using open-back, circum-aural headphones.

5.2   Testing methodology

The testing is carried out according to MUSHRA methodology [1], which is suitable for evaluation of intermediate audio quality and gives accurate and reliable results.
The labs carrying out the testing should have experience with the MUSHRA method from earlier exercises.
The MUSHRA test method applied here uses the original unprocessed material with full bandwidth as the reference signal (which is also used as a hidden reference), a number of hidden anchors, the conditions of the codec under test as well as the reference conditions with which the codec under test is to be compared.

5.3   Error Patterns and Error Conditions

Error conditions will be applied in experiment 5 according to document S4-030433.

5.4   Training phase

Prior to the actual testing a training phase is carried out in which the test subjects are familiarized with testing methodology and environment. The training is done following the same MUSHRA methodology as the actual test, though limited to four trials.

The training is based on the same codec, anchor and reference conditions as the blind grading phase.

5.5   Selection of subjects

The selection of subjects follows the guidelines given in [1].
In particular, it is recommended that experienced listeners should be used. These listeners should have some experience in listening to sound in a critical way. Such listeners will give a more reliable result more quickly than non-experienced listeners.

5.5.1   Screening of subjects

There is sometimes a reason for introducing a rejection technique either before (pre-screening) or after (post-screening) the real test. In some cases both types of rejections might be used. Here, rejection is a process where all judgements from a particular subject are omitted.
Any type of rejection technique, not carefully analysed and applied, may lead to a biased result. It is thus extremely important that, whenever elimination of data has been made, the test report clearly describes the criterion applied.

5.5.1.1   Pre-screening of subjects
The listening panel should be composed of experienced listeners, in other words, people who understand and have been properly trained in the described method of subjective quality evaluation. These listeners should:

- have experience in listening to sound in a critical way;
- have normal hearing (ISO Standard 389 should be used as a guideline).

The training procedure might be used as a tool for pre-screening.

5.5.1.2 Post-screening of subjects

Post-screening methods can be roughly separated into at least two classes:
- one is based on the ability of the subject to make consistent repeated gradings;
- the other relies on inconsistencies of an individual grading compared with the mean result of all subjects for a given item.

It is recommended to look to the individual spread and to the deviation from the mean grading of all subjects.

The aim of this is to get a fair assessment of the quality of the test items.

If few subjects use either extreme end of the scale (excellent, bad) and the majority are concentrated at another point on the scale, these subjects could be recognized as outliers and might be rejected.

Due to the fact that "intermediate quality" is tested, a subject should be able to identify the coded version very easily and therefore find a grade which is in the range of the majority of the subjects. Subjects with grades at the upper end of the scale are likely to be less critical and subjects who have grades only at the lowest end of the scale are likely to be too critical. By rejecting these extreme subjects a more realistic quality assessment is expected.

The methods are primarily used to eliminate subjects who cannot make the appropriate discriminations. The application of a post-screening method may clarify the tendencies in a test result. However, bearing in mind the variability of subjects' sensitivities to different artefacts, caution should be exercised.

Taking into account the size of the listening panel used throughout the experiments, the effects of any individual subject's grades is low and so the need to reject a subject's data is greatly diminished.

6   Description of listening tests

6.1   Test at 32 kbit/s stereo

Main Codec Conditions

Candidates	4	2 candidates, use case A and use case B encoder each
Error Conditions		No errors
Applications		32 kbit/s stereo

References:

Codec references	1	AAC-LC
Other references	1	RealAudio, for information purposes only
Open Reference	1	Original signal
Hidden Reference	1	Original signal
Anchors	2	3.5 kHz and 7 kHz band-limited original signal

Common Conditions

Number of test items	12	
Listening Level		To be chosen by subject
Listeners	15	Experienced listeners
Presentation randomizations	15	One per listener
Rating Scale		Continuous quality scale
Replications	2	The experiment covers all candidate codecs, 2 independent test sites
Listening System		Binaural high-quality headphones
Listening Environment		Room Noise: Hoth Spectrum at 30dBA (as defined by ITU-T, Recommendation P.800, Annex A, section A.1.1.2.2.1 Room Noise, with table A.1 and Figure A.1)

6.2   Test at 48 Kbit/s stereo

Main Codec Conditions

Candidates	4	2 candidates, use case A and use case B encoder each
Error Conditions		No errors
Applications		48 kbit/s stereo

References:

Codec references	1	AAC-LC
Other references	1	RealAudio, for information purposes only
Open Reference	1	Original signal
Hidden Reference	1	Original signal
Anchors	2	3.5 kHz and 7 kHz band-limited original signal

Common Conditions

Number of test items	12	
Listening Level		To be chosen by subject
Listeners	15	Experienced listeners
Presentation randomizations	15	One per listener
Rating Scale		Continuous quality scale
Replications	2	The experiment covers all candidate codecs, 2 independent test sites
Listening System		Binaural high-quality headphones
Listening Environment		Room Noise: Hoth Spectrum at 30dBA (as defined by ITU-T, Recommendation P.800, Annex A, section A.1.1.2.2.1 Room Noise, with table A.1 and Figure A.1)

6.3 Test at 32 Kbit/s stereo with frame loss

Main Codec Conditions

Candidates	2	Use case A encoder
Error Conditions	2	1% and 3% frame loss, random
Applications		32 kbit/s stereo

References:

Codec references	2	AAC-LC at 1% and 3% frame loss, random
Other references		
Open Reference	1	Original signal
Hidden Reference	1	Original signal
Anchors	3	3.5 kHz and 7 kHz band-limited original signal

Common Conditions

Number of test items	12	
Listening Level		To be chosen by subject
Listeners	15	Experienced listeners
Presentation randomizations	15	One per listener
Rating Scale		Continuous quality scale
Replications	2	The experiment covers all candidate codecs, 2 independent test sites
Listening System		Binaural high-quality headphones
Listening Environment		Room Noise: Hoth Spectrum at 30dBA (as defined by ITU-T, Recommendation P.800, Annex A, section A.1.1.2.2.1 Room Noise, with table A.1 and Figure A.1)

6.4 Material

See section 4.

6.5 Experimental Design

See section 5.2

6.6 Opinion Scale

The question asked of the subject will be a continuous Listening Quality Scale ranging from 0 to 100. The intervals 0 to 20 correspond to BAD, 20 to 40 to POOR, 40-60 to FAIR, 60 to 80 to GOOD, and 80 to 100 to EXCELLENT.

6.7 Processing

Processing is specified in section 7.

6.8 Duration of the Experiment

The duration of the experiment per subject depends on the number of trials and on the number of items per trial. However, it can be assumed that each vote requires listening to the respective item, the open reference and two (quality-wise) neighboring items 2 times. With an assumed average length per item of 7.5 s, the test will consume a listening time per subject of:

#trial * #hidden items/trial * (1+1+2) * #re-listenings * length/item.

For the blind grading phase in each of the sub-experiments this accounts to

12 * 9 * 4 * 2 * 7.5s = ~1.8 hours per subject.

For the training phase the number of re-listenings is assumed to be 1. For each of the sub-experiments this accounts to

4 * 9 * 4 * 1 * 7.5s = ~20 min per subject,

In order to avoid listener fatigue, sufficient breaks are required between the trials.

The experiments can be carried out with several subjects in parallel provided that a corresponding number of proper listening facilities are available.

6.9 Votes Per Condition

The number of votes per conditions is identical to the number of subjects per sub-experiment.

6.10 Randomizations

Each listener will be presented with the sound items in an individual random presentation order. Also the order of the trials will be random per individual.

The statistical significance of the results obtained in the sub-experiments with frame erasures may suffer from the effect that – due to different frame sizes of the candidate codecs – the frame erasures become effective at different points in time. The subjective relevancy of the frame erasure will thus not only depend on the frame error mitigation procedure of the codec candidates but also on the signal characteristic of the sound item at the erased frames.

In order to statistically mitigate this latter effect as much as possible there will be a randomization of the error patterns across the various listeners and the various listening labs. I.e. for each individual listening different error patterns will be used.

7 Processing

7.1 Main processing

Common for the processing of all conditions is that it will be done with concatenated material. To that purpose, the complete material including training material will be concatenated into one single file. This file will then be processed through the respective candidate or reference codecs, or the specific anchor signal processing will be applied. After processing and proper compensation of the processing delay, the material will be split up again into separate items.

7.1.1 Concatenation of material

The concatenation will be done using an executable script. The script will take as input all items of the training material and the test material. The order in the concatenation is such that the training material will precede the test material. Output of the concatenation script will be a single file containing the concatenated material and a time-file comprising name identifiers and time markers of the individual items. The time-file will be used after processing for split-up of the material to the individual items.

The concatenation script will be provided by: Ericsson

7.1.2 Processing of anchor conditions

The creation of anchor conditions should be performed using the "resamp-audio" tool of the AFSP library.

The program is freely available on the Internet, a copy (including usage assistance, if necessary) will be provided to the host lab by Coding Technologies.

Usage examples: Assuming a stereo .wav file sampled at 48kHz as the input signal, the anchor signals are created by the following commandlines:
- 3.5kHz anchor signal: ResampAudio –i 1 –f cutoff=0.0729167 <input-file> <output-file.wav>
- 7kHz anchor signal: ResampAudio –i 1 –f cutoff=0.145833 <input-file> <output-file.wav>

7.1.3 Candidate and reference codec processing

The concatenated material file will be created by the hostlab. After processing, the encoded material as well as the concatenated original files will be delivered to candidates for cross-checking. The candidates make sure that possible codec delay is properly compensated. The procedure of how the delay was compensated need to be reported as part of the respective candidate codec descriptions. The analogue processing procedure will be applied to the reference codecs.

7.1.3.1 Impaired channel processing

The host lab does the impaired channel processing with a FER pattern file identifying correct and erroneous frames. The host lab will produce that file using a script provided by:

Volunteering organization tbd.

The input to the script is a seed to the random generator included in the script. It will be provided by theETSI Secretary, Paolo Usai after candidate submission.

The format of the FER pattern file is ASCII text with one line per frame. A '0' (zero) in a line indicates a correct frame, a '1' (one) indicates a frame erasure.

The length of the file will be sufficient to cover the complete concatenated training and test material, assuming a minimum frame length of 10 ms. Note, that depending on the actual frame size of the individual candidate codecs, the FER patter file will not be applied in its entire length.

In order to realize the randomization of the error patterns across each individual listening (see paragraph 7.9), before application the error pattern file will be circularly shifted with individual time offsets. The time offset (in frames) for listener *s*, lab *p* is calculated as follows:

$offs = \mod(s \cdot 12345 + p \cdot 31415, totlen)$, where *totlen* is the total length of the FER pattern file (in frames).

The circular shift will be done using an executable script with the command line syntax:

shiftcirc –<offs> <inpat> <outpat>

The script will be provided by volunteering organization <tbd>.

7.2   Post-processing

The post-processing comprises the steps of up-sampling to the original sampling frequency of 48 kHz and split-up of the concatenated material.

7.2.1   Up-sampling

Resampling to the original sampling frequency of 48 kHz will be done using "resamp-audio" tool from the AFSP library.
The program is freely available on the Internet, a copy (including usage assistance, if necessary) will be provided to the host lab by Coding Technologies.

In order to avoid possible signal clipping distortions introduced after re-sampling during the conversion of the internal floating-point to the 16-bit integer representation of the wav-file, before this conversion the re-sampled signal is attenuated by applying a constant factor of 0.93 (≈ -0.3 dB).
Note, in order to ensure that this factor is consistently applied to all codec, anchor and reference conditions, the program must always be applied, even if actually no re-sampling is necessary since the sampling rate of the respective signal is already 48 kHz.

Usage example: To attenuate and resample a .wav file of an arbitrary sampling rate to 48 kHz sampling rate, the following commandline should be used:
-    ResampAudio –s 48000 -g 0.93 <input-file.wav> <output-file.wav>

7.2.2   Split-up of processed material

The split-up of the processed material to partial sound files will be done using a program which is functionally inverse to the concatenation procedure described in section 8.2.1. The program will take the concatenated and processed (and delay-compensated) file along with the time-file as input. On time intervals of length $2N$ around the time markers n_i indicated by the time-file it will first apply a roll-off/roll-on window with Hanning characteristic:

$$x_w(n) = x(n) \cdot \sin^2\left(\frac{\pi}{2N}(n - n_i)\right)$$

and then split-up the file at the time markers n_i. N corresponds to a roll-off/roll-on time of 10 ms. I.e. $N = 0.01 * f_s$ where f_s is the sampling frequency of the file. Partial sound files will be generated with names according to the name identifiers comprised in the time-file and a processing tag, identifying the kind of processing.

The split-up program will have the following command-line syntax:
splitup <infile> <processing_tag> <timefile>

The input file format is wav if the name has the extension .wav. If it has the extension .raw, then a headerless 1-channel (mono) 16 bit PCM file sampled at 48kHz is assumed.
The processing tag is an arbitrary string which is added to the name identifiers given in the time-file.

The generated output files are of .wav format.
The program will be provided by Ericsson.

7.3   Blinding of material

The purpose of blinding is to unveil the identity of the different codec, reference and anchor conditions. The blinding is done using a script that maps the de-concatenated items onto files with names unveiling the actual condition. The mapping shall vary across the different items.

The blinding of the material is made by the host lab, based on a script created by the host lab. The host lab is responsible to keep the mapping confidential.

Annex A: Listeners Instructions

The following is an example of the type of instructions that should be given to or read to the subjects in order to instruct them on how to perform the test.

Instructions to be given to subjects

1   Training phase

The first step in the listening tests is to become familiar with the testing process. This phase is called a training phase and it precedes the formal evaluation or blind grading phase.

The purpose of the training phase is to allow you, as an evaluator, to learn how to use the test equipment and the grading scale.

In the training phase you will make a short test similar to the one you will make in the blind grading phase of the test.

No grades given during the training phase will be taken into account in the actual tests.

2   Blind grading phase

The purpose of the blind grading phase is to invite you to assign your grades using the quality scale. Your grades should reflect your subjective judgement of the quality level for each of the sound excerpts presented to you. Each trial will contain <x> signals to be graded. Each of the items is approximately 5 to 10 s long. You should listen to the reference and all the test conditions by clicking on the respective buttons. In a first step it is recommended to browse through all signals within each trial in order to get a coarse impression of the offered quality range. Then you may listen more carefully and start to rank the signals. You may listen to the signals in any order, any number of times. Use the slider for each signal to indicate your opinion of its quality. When you are satisfied with your grading of all signals you should click on the "register scores" button at the bottom of the screen.


You will use the quality scale as given in Fig. 1 when assigning your grades.

The grading scale is continuous from "excellent" to "bad". A grade of 0 corresponds to the bottom of the "bad" category, while a grade of 100 corresponds to the top of the "excellent" category.

In evaluating the sound excerpts, please note that you should not necessarily give a grade in the "bad" category to the sound excerpt with the lowest quality in the test. However one or more excerpts must be given a grade of 100 because the unprocessed reference signal is included as one of the excerpts to be graded.

You should not discuss your personal interpretation or your gradings with the other subjects at any time during the test.

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An example of the user interface used in the blind grading phase
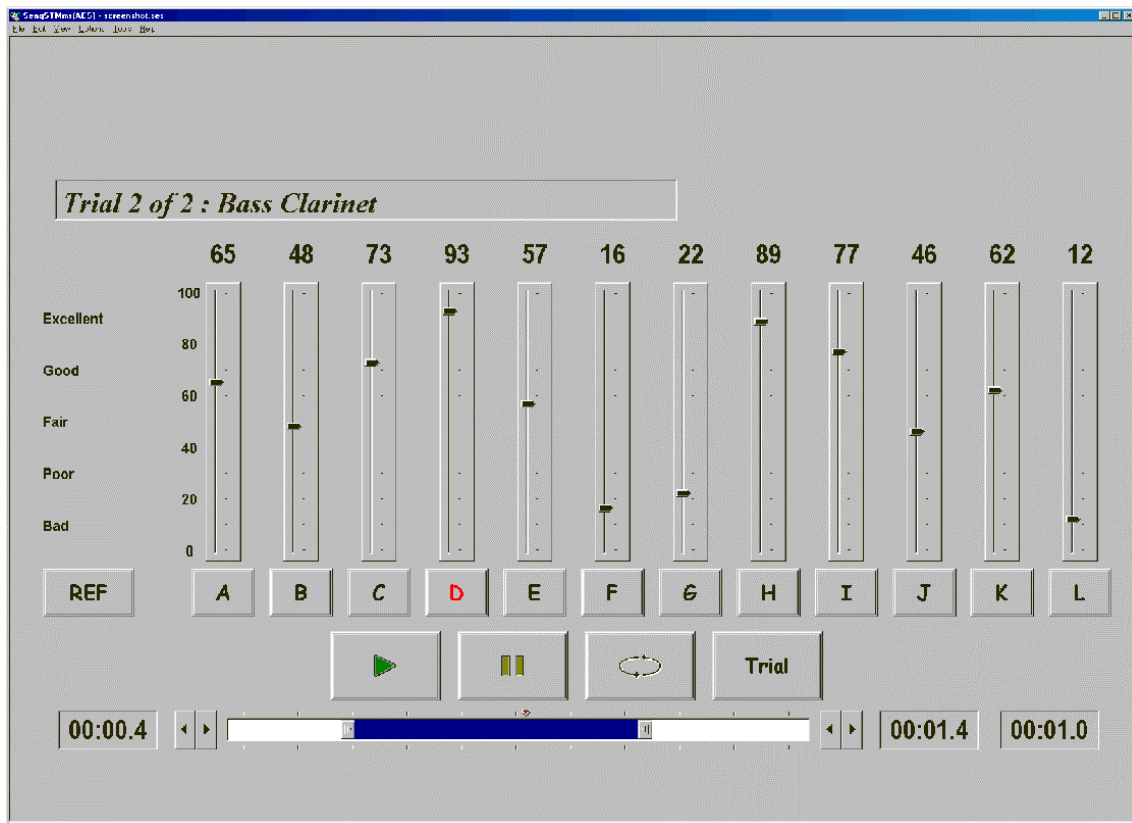
1534-04

Annex B: File formats, Naming conventions, Directory structures, Platforms

B-1   File formats

Unless specified differently all signal files are of wav format using a linear 16-bit PCM sample representation. The sampling rate prior to pre-processing and after post-processing is 48 kHz.

B-2   Computer platform

The processing will be done in a Cygwin environment under Windows.

B-3   Directory structure

```
Root
|
-.- org            Original (unprocessed) material
 .- preproc        Pre-processed material
 .- ep             Error pattern and program to generate it
 .- bin            Programs and scripts required for pre-processing
                   and error pattern generation
 |
 .- cand_1         Directory for candidate codec 1
 |  |
 |  .- processed   processed material
 |  .- bin         programs and scripts required for processing
 |  .- tmp         intermediate processing results
 |
 .- cand_2         Directory for candidate codec 2
 |  |
 |  .- processed   processed material
 |  .- bin         programs and scripts required for processing
 |  .- tmp         intermediate processing results
 |
.- ref             Reference codecs, anchors and hidden references
 |  |
 |  .- processed   processed material
 |  .- bin         programs and scripts required for processing
 |  .- tmp         intermediate processing results
 |
 .- output         Processed and blinded material per sub-experiment
    |
    .- exp_1
    .- exp_2
    .- exp_3
```

.

B-4   File naming

1. Unprocessed original signals:

<signal_category>_<item_no>_org.wav

where

- <signal_category> is one out of {p, c, si, v, sm, sp} (for Pop, Classic, Single instruments, a capella vocals, Mixed speech and music, Speech)
- <item_no> is a number identifying the item out of the respective signal category.

2. Pre-processed original signals after concatenation:

all_cat.wav

The corresponding time-file comprising the segmentation information is:

all_cat.tim

3. Material after main processing:

all_cat_<codec_id>_<exp_id>.wav

where

o <codec_id> is one out of
   o {cand_1, cand_2, AAC, RN, lp3500, lp7000, hidref, opref} for the stereo experiments below 48kbit/s,
   o {cand_1, cand_2, AAC, RN, lp3500, lp7000, hidref, opref} for the stereo experiments at 48kbit/s.
o <exp_id> is one out of {1, 2, 3}

4. Processed material after de-concatenation:

<signal_category>_<item_no>_<codec_id>_<exp_id>.wav

5. Processed material after blinding:

<signal_category>_<item_no>_<exp_id>_<sub_exp>_<cond_id>.wav

where <cond_id> is one out of

   o {1-7} for the stereo experiments below 48kbit/s,
   o {1-8} for the stereo experiments at 48kbit/s.