Universal Radio Communication Tester R&S CMU200

Signalling and measurements on GSM-AMR mobile phones

Everyone is talking about high data rates and innovative data applications. But the development of classic voice transmission is also not at a standstill. The latest innovation in this field is adaptive multirate (AMR), and it is specified across systems for both UMTS and GSM. The R&S CMU200 provides high-quality measurement technology for AMR.

Another article about the R&S CMU200 can be found on page 32.

More information and data sheet at www.rohde-schwarz.com (search term: CMU200)

REFERENCES

[1] Universal Radio Communication Tester R&S CMU200 – Audio measurements on mobile phones. News from Rohde&Schwarz (2001) No. 172, pp 18–19

[2] Audio Analyzer R&S UPL – Measuring the acoustic characteristics of 3G mobile phones. News from Rohde&Schwarz (2002) No. 173, pp 15–17

Back to the future with AMR

When the connection in analog transmission gets worse, the voice quality slowly deteriorates. You begin to hear noise and crackling but can still communicate. The situation is different with digital voice transmission: The quality remains at virtually the same level for a long time, suddenly there are interruptions and shortly thereafter nothing but dead silence.

The idea behind adaptive multirate is to adapt the behaviour of digital voice transmission to that of analog transmission in order to maintain the connection for as long as possible. In other words, AMR increases the range of a cell, thereby reducing infrastructure costs.

The principle of digital voice transmission

With digital voice transmission in mobile radio networks, a coder converts the analog voice signal into a digital one. The coder assesses the signal and protects the information content to a greater or lesser degree by inserting additional redundancy bits. This explains why the high voice quality is maintained until shortly before the connection breaks off. Incorrectly transmitted bits are detected by the voice decoder on the basis of the redundancy bits and corrected. The better protection of the important voice information ensures that the good voice quality is maintained for a long time. When the decoder can no longer correct the errors, the voice information completely fails, which is perceived by the users as interruptions in voice transmission.

Reduced voice quality – longer connection

The voice failure can be delayed by increasing the protection against transmission errors. The more redundancy bits are used, the longer errors can be corrected while the connection deteriorates. Additional bits would require a greater transmission bandwidth, which, however, is limited. One possible solution is to reduce the number of information bits by the increase in redundancy bits; Reducing the payload bits, however, results in poorer voice quality, as not all the nuances of voice can be coded any longer. Under good propagation conditions, users do not want to give up good voice quality. Consequently, the voice quality has to be dynamically adapted to the connection quality – and that is exactly what AMR does.

Four voice codecs instead of one

AMR specifies eight full-rate and six half-rate voice codecs of different voice quality. The base station selects up to four different codecs and transmits them to the mobile phone together with instructions as to which voice codec is to be used for which signal quality. While the connection is on, the mobile phone cyclically measures the signal-to-noise ratio and the receive level, determines the RF connection quality on the basis of the results and selects a suitable codec for the downlink. Subsequently it requests the desired voice codec from the base station, which then decides if and under what conditions this codec is used.

The base station measures the connection quality and selects the codec for the opposite direction. The mobile phone must set the codec chosen by the base station and then inform the base station after having made the setting. In order for the base station and mobile phone to select the correct voice codec, data must be exchanged within the voice data packet (inband signalling).

Measurements on AMR voice coders

A mobile radio tester must basically check two things on an AMR mobile phone: the correct selection of the voice codec (inband signalling) and the voice quality. The necessary measurement functions are derived from this information. The mobile radio tester must have at its disposal all AMR voice codecs from which the user can make a selection and define instructions. The mobile radio tester checks the adherence to the instructions in the mobile phone by changing the RF level and evaluating the voice codec requested by the mobile phone. To assess the voice quality, it must be possible to select the actively used voice codec and to set it for the duration of the measurement independently of the mobile phone's request. The mobile radio tester determines the voice quality by means of audio measurements and an AMR-suitable BER measurement.

AMR tests with the R&S CMU 200

The option R&S CMU-K45 enhances the Universal Radio Communication Tester R&S CMU 200 into an AMR mobile radio tester that provides all required test functions in the accustomed convenient way. All AMR voice codecs can be selected and linked to instructions (FIG 1). In a clearly arranged menu, inband signalling during the call can be set and the mobile phone's response analyzed (FIG 2). The BER measurement automatically adapts to the active AMR voice codec. Together with the options R&S CMU-B52 and R&S CMU-B41, the R&S CMU 200 also performs all audio measurements [1, 2].

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FIG 1 The R&S CMU 200 provides all specified voice codecs. In the Rate Set Editor, up to four codecs are selected from the eight full-rate and six half-rate voice codecs and linked to user-definable instructions for the switching thresholds between the voice codecs. This rate set is transmitted to the mobile phone during connection setup.

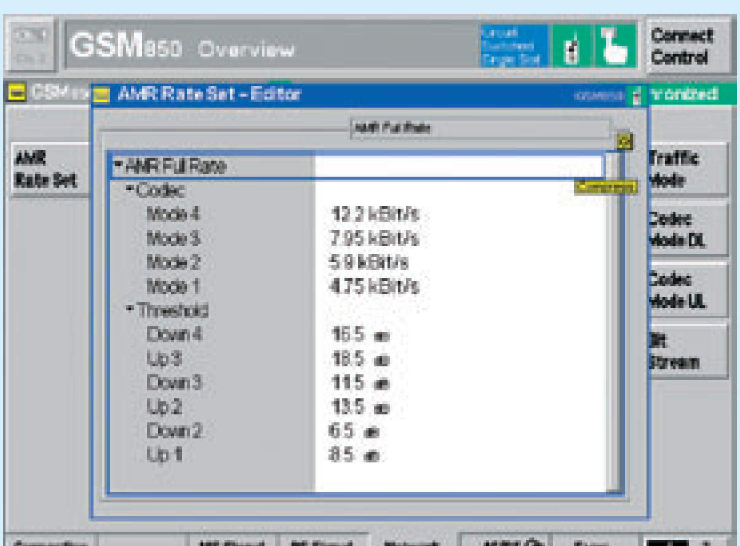

FIG 2 The user at all times has control over which voice codec is to be used in either connection direction. The active voice codec for the downlink is set in the "used by BTS" field, and the one for the uplink is set in the "commanded by BTS" field. The "requested by MS" field always displays the voice codec for the downlink currently requested by the mobile phone. The "used by MS" field shows the voice codec used by the mobile phone on the uplink.