ETSI TR 103 414 V1.1.1 (2016-09)

TECHNICAL REPORT

TETRA and Critical Communications Evolution (TCCE); Study into the provision of speech services over QAM channels

Reference
DTR/TCCE-04183

Keywords
radio, TETRA, V+D

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Important notice

The present document can be downloaded from:
http://www.etsi.org/standards-search

The present document may be made available in electronic versions and/or in print. The content of any electronic and/or print versions of the present document shall not be modified without the prior written authorization of ETSI. In case of any existing or perceived difference in contents between such versions and/or in print, the only prevailing document is the print of the Portable Document Format (PDF) version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at
https://portal.etsi.org/TB/ETSIDeliverableStatus.aspx

If you find errors in the present document, please send your comment to one of the following services:
https://portal.etsi.org/People/CommiteeSupportStaff.aspx

Contents

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (https://ipr.etsi.org/).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee TETRA and Critical Communications Evolution (TCCE).

Modal verbs terminology

In the present document "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

"**must**" and "**must not**" are **NOT** allowed in ETSI deliverables except when used in direct citation.

Introduction

The present document contains the results a study by ETSI TC TCCE WG4 into the extension of QAM channels in TETRA to carry speech in a native manner.

The reason for such a study was to investigate a further evolution of TETRA to improve its capabilities and take advantage of more flexible use of capacity and range/sensitivity of the TEDS QAM channels. Such a solution would allow closer integration of voice and data to improve site channel efficiency.

The advantages that were foreseen in undertaking the study are as follows:

a) To provide more flexibility for system designers, allowing the use of lower rate modulation schemes to obtain improved ranges, or higher rate modulation schemes to provide more capacity from a base station.

b) To allow the use of increased bit rate per spectrum bandwidth of higher modulation schemes to allow more calls within a given bandwidth from a single site, provided that the altered carrier to interference radio is managed.

c) To improve efficiency by allowing speech and data calls to be carried on the same carrier, with each timeslot being allocated to speech or data payload on a slot by slot basis on uplink and downlink.

NOTE: This does not imply that channel allocation for voice calls needs to be sent or changed on a slot by slot basis.

d) To allow simultaneous speech and packet data by sending both together in the same timeslot, or by more flexible management of timeslots.

e) To improve support for end to end encrypted speech, and also to enable the use of alternative codecs, by taking advantage of increased bandwidths for speech calls.

This study is the result of deliberations in TCCE WG4, and represents the concepts and ideas discussed up to the time of publication which could be the basis of further work, which in turn could lead to change requests to [i.1] to introduce this functionality. However further work would be required to determine solutions to standardizing this functionality, and the standardization process could diverge from the concepts presented in the present document.

1 Scope

The present document contains the results of a study into the extension of QAM channels in TETRA to carry speech in a native manner. The areas where change can be foreseen to the TETRA standard [i.1] to date in order to add this functionality are described at a high level.

NOTE: The present document is not sufficient to enable change requests to be generated against [i.1], and further standardization work would be needed to arrive at a solution. A standardized solution could differ from the concepts presented in the present document.

The main body of the present document describes the requirements that are satisfied and the mechanisms envisaged within this study that could fulfil this extension of TETRA. Annex A presents some possible modifications which could be needed for PDUs and information elements specified in [i.1] and Annex B refers to [i.1] on a clause by clause basis to indicate which clauses could require changes in order to implement these mechanisms.

2 References

2.1 Normative references

Normative references are not applicable in the present document.

2.2 Informative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

The following referenced document is not necessary for the application of the present document but it assists the user with regard to a particular subject area.

[i.1] ETSI EN 300 392-2/TS 100 392-2 (latest version of either applies): "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 2: Air Interface (AI)".

[i.2] ETSI EN 300 392-7: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 7: Security".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in [i.1] and the following apply:

U-plane: plane for user traffic signalling

TM-SDU: SDU from the layer above MAC (i.e. LLC)

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACELP	Algebraic Code-Excited Linear Prediction
BS	Base Station

CCH	Control CHannel
CLCH	Common Linearization CHannel
CMCE	Circuit Mode Control Entity
codec	Coder-decoder
D8PSK	Differential 8 Phase Shift Keying
DL	DownLink
DQPSK	Differential Quadrature Phase Shift Keying
DTX	Discontinuous Transmission
IP	Internet Protocol
LLC	Logical Link Control
MAC	Medium Access Control
MCCH	Main Control CHannel
MCCH-Q	Main Control CHannel-QAM
MLE	Mobile Link Entity
MS	Mobile Station
PDU	Protocol Data Unit
QAM	Quadrature Amplitude Modulation
SCCH	Secondary Control CHannel
SDU	Service Data Unit
SNDCP	SubNetwork Dependent Convergence Protocol
SwMI	Switching and Management Infrastructure
TDMA	Time Division Multiple Access
UL	UpLink

4 Services for speech in QAM channels

4.1 Purpose

The purpose in extending QAM channels to prove a native speech mode service are as follows:

a) Lower rate modulation schemes such as 4-QAM offer better receiver sensitivity, thus leading to a higher range compared with phase modulation channels, if the average transmitted power remains the same across phase modulation and 25 kHz QAM channels.

b) Higher rate modulation schemes allow more capacity on a single carrier, allowing one base station carrier to carry more calls. Thus, when taken together with the previous point, the system designer can have more flexibility in trading off the capacity and range of base stations within the overall system design.

c) The increased bit rate per spectrum bandwidth of higher modulation schemes will allow more calls within a given bandwidth from a single site, provided that the altered carrier to interference ratio is managed.

d) Voice and data calls can be carried on the same carrier, and channel allocation can allow optimized channel efficiency by allocating each timeslot to speech or data payload on a slot by slot basis on uplink and downlink. This will give increased flexibility compared with the channel allocation schemes currently possible on phase modulation channels.

NOTE: This does not imply that channel allocation for voice calls needs to be sent or changed on a slot by slot basis.

e) Simultaneous speech and packet data can be achieved by sending both in the same timeslot, or by more flexible management of timeslots.

f) Improved support for end to end encrypted speech can be provided where more channel bandwidth is available to the end to end encrypted call, which will enable encryption synchronization to be sent without stealing from speech frames.

g) Alternative codecs could be supported, as different channel bandwidths can be provided.

4.2 Services

A scheme providing speech on QAM channels will support the following existing TETRA speech services:

a) Individual call - duplex and semi duplex

b) Telephone call

c) Group call

d) Variants of these, such as broadcast calls.

The scheme will also support the following TETRA supplementary services:

a) Late entry

b) Talking party identity

c) Calling party identity

d) Further supplementary services as required.

The scheme will also maintain support for security services such as authentication, air interface encryption and end to end encryption.

Although the principles described in the present document were primarily intended to support speech calls on QAM channels, they are equally applicable to other forms of media or data where a regular allocation of uplink and/or downlink data is required. These include possibilities such as video and telemetry.

5 Principles of the studied solution

The principles of the solution proposed as a result of this study are described in this clause.

Speech will be carried in existing physical layer uplink and downlink QAM slots, and will make use of the QAM rates, bandwidths and error correction mechanisms already defined for signalling and data.

Speech frames will be packed into timeslots using a new form of MAC PDU and sent directly in the channel timeslots, and will not be encapsulated in IP when carried over the air interface. This avoids the overheads associated with IP headers. It will be possible to carry signalling and user data within the same timeslots, where there is capacity available.

The channel allocation for assigning a speech call to timeslots on a QAM channel will use a modified version of the existing MAC channel allocation.

The slot grant will allow different timeslot repetition rates to be used, to take advantage of packing more (or less) than two audio frames into a timeslot - e.g. to allocate one timeslot in 8 to carry a call if 4 audio frames are packed into a timeslot on one of the higher rate channels.

The slot grant will specify a start timeslot and a repetition rate for timeslots for the call. It will be finite, so that the MS will lose the allocation if it does not receive 'top-up' grants which allocate further timeslots to the call. There will also be a mechanism to cancel granted slots, e.g. at the end of a speech item.

A similar process will allocate sets of timeslots on the DL for reception of the call, by defining a start slot and repetition rate. The MS will be able to save power ('snooze') between slots, and there will be the possibility of staying awake beyond a slot that is designated for speech reception in order to receive data. The process will be based on Napping, but changed from the current Napping procedures.

The uplink channel allocation may be independent from the downlink allocation, either within the associated carrier, or by making an allocation on a different carrier. The uplink allocation may specify a subslot and a bandwidth which is different from the bandwidth of the downlink.

The BS will not to permit link adaptation signalling by an MS receiving in a group call, as any MS could reduce the throughput for all MSs; and it is assumed that network design will enable an MS to find a better cell for the call if the link for a received group call is marginal. Link adaptation could be possible under command of the BS in an uplink transmission, and in both directions in an individual call.

The solution will permit end to end encryption of speech, and will also support alternative codecs. The extra available bandwidth of some channels may permit encryption synchronization to be sent without needing to steal bandwidth from the accompanying encrypted speech.

6 Mechanisms

6.1 General

This clause describes additional mechanisms to those specified in [i.1], or modifications to the mechanisms specified in [i.1] that might be needed in order to carry speech or other forms of media efficiently on QAM channels that are assigned to carry data.

NOTE: PDUs referred to in this clause are defined in [i.1].

6.2 Channel allocation

6.2.1 Existing TETRA mechanisms

In current packet data mechanisms, the MS may receive a change of allocated channel to enable it to receive a data transmission. This may be an augmented channel allocation, indicating the relevant parameters of a QAM channel. The MS would expect to receive signalling on the channel after it arrives. It does not expect to receive advance notice of a DL data transmission. It may receive a separate slot grant when on the new channel if it needs to send UL signalling.

If the MS is already residing on the channel where a call is to be received, no channel allocation need be sent in the MAC RESOURCE PDU, just indication of which slots in which to receive the call, although the use of channel allocation to put the MS into assigned mode may still be needed. The same is true when granting slots for transmission.

6.2.2 Potential mechanisms

For QAM speech, if a different channel is to be used to carry the call, it is expected to be possible to send a channel allocation and slot grant (for transmission) or repetitive reception information (for reception) together in the interests of efficiency. As slot and frame numbering has to be the same on all carriers of the base station according to [i.1], there would not be a synchronization issue for the MS in counting timeslots until the first intended reception slot, or first granted slot for transmission.

NOTE 1: Where a channel is shared between cells, the slot assignment needs to be synchronized between cells to avoid any potentially conflicting channel allocation. An implementation may prefer to avoid doing this, and to send slot grants or repetitive reception information on the shared channel itself.

The channel allocation and MAC RESOURCE PDU would be group addressed in the case of a group call, and could move all MSs in the call to a different channel from that on which they currently reside. Each MS would need to decide whether to complete any ongoing data exchange, or abandon it in favour of this new channel allocation, or continue it on the new channel.

NOTE 2: The actual mechanisms are for further study.

At the end of a call, the MSs could remain on the same channel, for example if the channel is being used as both common CCH and an assigned channel. Alternatively a fresh MAC RESOURCE PDU could be sent with a channel allocation to a different channel, or to go to an appropriate common control channel, which could be combined with CMCE signalling terminating the call.

NOTE 3: It is for further study what will happen if the MS was also involved in a non-speech data transmission during a speech call, and the MS is sent to a different channel at the end of the speech call before that data transaction has completed.

Link adaptation is possible in an individual call, but not in a group call. In an individual call, the MS could perform random access and send the L2-LINK-FEEDBACK-INFO PDU. The BS could respond and change the channel allocation (and slot grant); the MS can be informed that the new allocation replaces the previous one. Alternatively the BS could use the L2-LINK-FEEDBACK-CONTROL PDU to terminate (and ignore) the feedback, or even use a new value of 'Link feedback control type' to temporarily deny the request and prevent the MS from sending more requests for some period of time. It could be useful for the BS to indicate whether feedback would be possible or not during the call when it sent call set up signalling.

In a group call, any MS could affect the rate and throughput for all MSs. It is better therefore that the mechanism is not allowed to change the rate of the downlink in a group call.

6.3 Independent uplink

In addition to the normal uplink which is currently described in [i.1], where the frequency of the uplink of a channel is indicated in system related broadcast information as a duplex offset from the downlink frequency, an alternative uplink channel can be assigned that is different from this normal uplink. This will allow several low channel bandwidths, lower data rate uplinks to be used in conjunction with a higher bandwidth downlink channel allocation. This will allow the MS to maximize its available transmit power to give the best possible energy in each transmitted symbol. By allowing the BS to provide greater frequency separation between different MS uplink transmitters, it could also reduce the probability of a weak uplink signal being degraded by adjacent channel effects from another strong uplink signal.

The mechanism could be used for QAM data transmissions as well as speech.

6.4 Slot granting

The requirements for allocating slots on an uplink channel for voice calls are:

a) To provide a delay until the first slot allocated to the call, indicated as a delay from the current timeslot

b) An allocation of a half slot or one or more full slots on the UL to carry the call

c) A regular repetition (which is likely to be different to the initial delay)

d) A number of slots for which the allocation is valid

e) A means of 'topping up' the slot grant to extend its period

f) The replacement of earlier grants with later ones such that grants can overlap, and the end of the granted slots is defined as the furthest end point from any of the grants

g) Late entry to the granted slot pattern

h) Indication whether the slot counts for initial delay and repetition include the 18th frame

Whereas it might be possible to create a repeating pattern by using the current multiple slot granting mechanisms in [i.1] by sending several separate basic slot grants in a multiple slot granting element, possibly changing the rules for granting slots in Frame 18 so that the slot counts can be made or signalled to jump over Frame 18, the mechanism would be limiting and would lead to a large element with large numbers of basic slot grants if a reasonable grant period (several seconds) was desired.

Therefore use of the currently reserved element in multiple slot granting is preferred, to create a different type of slot grant more suitable for the purpose of transmitting streaming media. A possible redefinition of the element, with future extension, is shown in Annex A.

An MS may receive a conventional slot grant as well as a repeating slot grant, to allow transmission of data during a speech transmission to provide a simultaneous speech and data service.

6.5 Repetitive reception

To allow receiving parties to make efficient use of battery power during call reception, the BS can indicate to the MS during which timeslots it will need to be awake to receive call information, and the MS is allowed to immediately return to a power saving mode following reception of each timeslot. However to help with BS downlink scheduling, the BS could be allowed to slip a DL speech frame into a following slot, hence if the MS expected to receive something in a particular timeslot but nothing was sent, the MS remains awake until something was received in the call (subject to end of call timeouts, etc.). The MS will return to power saving once the information has been received, waking in the originally granted slot pattern.

The BS may also signal to the MS within a designated reception timeslot that it is expected to remain awake beyond that designated slot if there is further information (e.g. data) to send to that MS.

The slip of information transmission into a following slot by the BS, or transmission of further data in subsequent slots, will not cause any slip of the subsequent receiving pattern, and if the BS does need to change the slots in which information is received, it will need to cancel an existing pattern and send a new one to the MS.

The same mechanism will also be applied to the transmitting MS, to inform the MS when to wake in case of downlink signalling.

A new mechanism, referred to as 'snoozing', is defined in the present document as the napping mechanisms defined in [i.1] do not readily adapt to the repetitive reception process. The element is included in the revised channel allocation element shown in Annex A.

6.6 Access assignment

The Access Assignment Channel is used in [i.1] to indicate whether the (normal) uplink of a timeslot is reserved or available for random access. No changes are foreseen as a result of this study, despite the change in purpose of the uplink on a control/data channel to allow streamed media to be carried. This study does not propose dividing the uplink of the channel into sub-channels in the frequency domain, hence the Access-assign PDU will remain valid for the entire timeslot.

6.7 MAC traffic format

The existing MAC, LLC and MLE headers used for data transfer would be inefficient to carry speech or other media due to the number of header bits required in addition to the speech or media. Therefore the traffic frames will be placed in new MAC PDU types directly, with internal headers relating to the traffic contents.

A MAC DATA PDU containing user data may also be contained in the same timeslot following speech; this also allows fragmentation into following timeslots.

7 Protocol revisions

7.1 General

In this clause, the modifications considered to the protocols specified in [i.1] that would implement the mechanisms described in clause 6 within the study are described. Where PDU and information element modifications have been considered, the modifications considered within the study are presented in Annex A.