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Universal Mobile Telecommunications System (UMTS);**

LTE;

**Interworking between the IP Multimedia (IM) Core Network (CN)
subsystem and Circuit Switched (CS) networks
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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
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