white paper

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MMTel – a standard for multimedia services over IMS

With interoperability, performance and fixed mobile convergence, MMTel (multimedia telephony) is a standardized solution for offering VoIP-based telephony-grade and multimedia services. Accordingly, it gives operators a way of offering communications services that more clearly cater to consumers' needs.

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1 Executive summary

3GPP/NGN Multimedia Telephony (MMTel), a new global standard based on the IP Multimedia Subsystem (IMS), offers converged, fixed and mobile real-time multimedia services that allow users to communicate using voice, video, and chat. It also allows the user to share image files and video clips.

Users can easily change the service by adding and dropping media streams and calling parties during an ongoing session. They can also easily switch between sessions, devices and fixed or mobile connections, or start a new chat session, upgrade the session to a voice or video call, or add a new participant.

The MMTel standard contains clear operator-to-operator interconnect specifications. This means that users who belong to different operators can communicate with each other using all available multimedia services.

The standard – which is a joint project by the 3rd Generation Partnership Project (3GPP) and European Telecommunications Standards Institute/Telecoms and Internet Converged Services and Protocols for Advanced Networks (ETSI/TISPAN) standardization bodies – is an evolution of existing fixed and mobile telephony services. The intent is to eventually phase out circuit-switched technologies, replacing them with an all-IP solution.

A principal characteristic of the MMTel standard is that the mobile access is based on Internet Protocol (IP). This makes the standard future proof, and it works well with the current standardization activities ongoing within 3GPP where the 3GPP Long Term Evolution (LTE) and High-Speed Packet Access (HSPA) mobile access technologies are being defined. Therefore, for the first time, exactly the same services (that is, multimedia and supplementary services) can be delivered to fixed as well as mobile clients.

The MMTel standard will enable carriers and operators to market the mix of revenue-generating services that users have already begun to demand.

Although investments in IMS technology represent long-term commitment, by starting with MMTel operators and carriers can generate new revenue streams almost immediately – for instance, by adding new multimedia services to their offering.

2 Introduction to MMTel

2.1 New real-time multimedia services

Person-to-person communication solutions meet users' need to stay in touch with family members or colleagues across geographical distances. Since their inception, telephony solutions have catered to this need. They have also evolved. This evolution has mostly focused on the technical aspects of delivering services – for instance, the digitalization of the telephone network. User services, by contrast, have not seen a rapid evolution. And those changes that have occurred have primarily been introduced in mobile networks – for example, 3G video telephony and MMS.

Supporting both fixed and mobile telephony, the new MMTel standard is a fundamental evolutionary step that changes and enhances mass-market telephony services. With it, users will be able to enjoy several new real-time multimedia services, including voice, video, file sharing, instant messaging and conference calls. For examples of user services made possible by MMTel, see *Figure 1*.

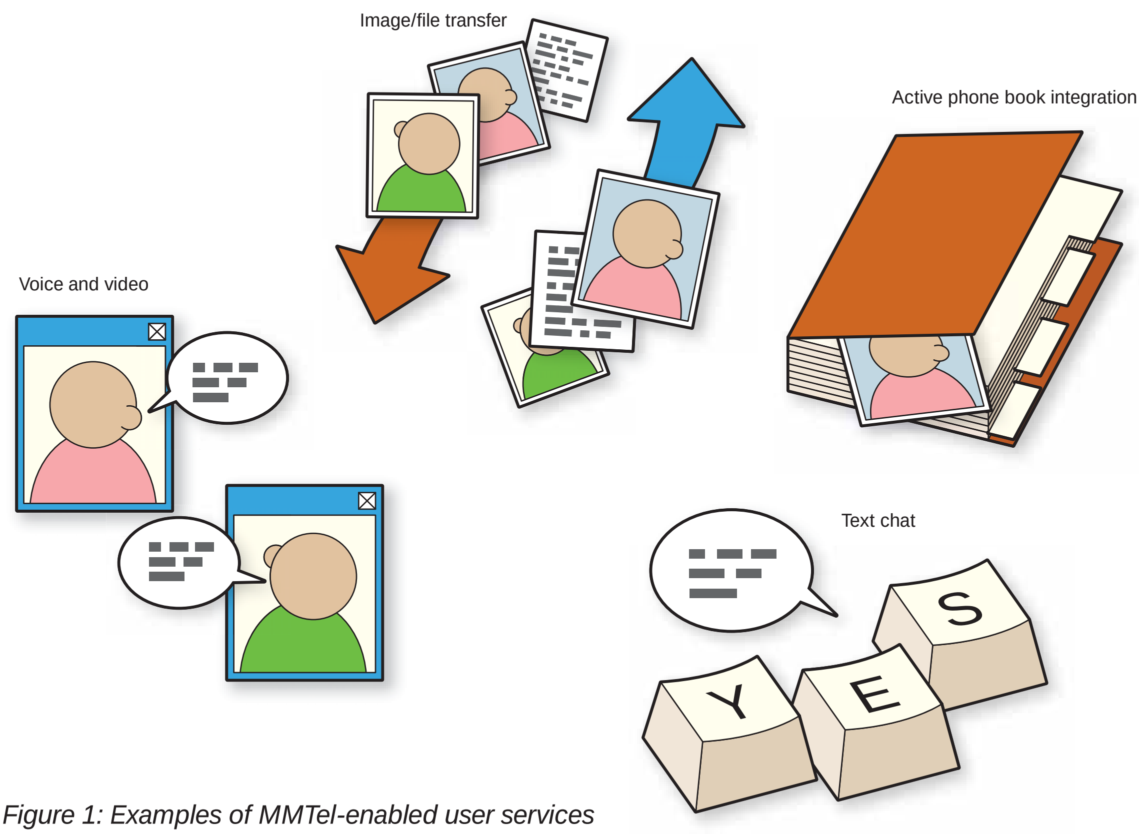


Figure 1: Examples of MMTel-enabled user services

2.2 Standardized services using IMS

MMTel is a service set that uses IMS architecture. IMS is a standardized architecture for controlling and delivering multimedia services that employ IP for transport and Session Initiation Protocol (SIP) for service signaling. It is important to note that in this context "standardized" refers only to the architecture (nodes, protocols, interfaces) and not to the services delivered on top of it.

In both WCDMA and GSM technology, the standards encompassed the architecture as well as the service set. This means that the service cannot be taken away from the architecture. IMS, however, is different. It is a standardized architecture but it does not include any standardized services. Consequently, IMS must be combined with something else e.g. MMTel services.

For a service to be deployed to the mass market, it must contain several standardized functions, including: a Network-to-Network Interface (NNI) that supports interconnection between operators; and a User-to-Network Interface (UNI) that enables users with standardized devices to benefit from the services.

One advantage of the IMS architecture is its versatility and flexibility. It can be used to deliver all kinds of services. Even so, this does not mean that all services over IMS have been standardized. In fact, only a few service sets have been standardized to date.

Besides MMTel, these are the standardized IMS service sets:

- Presence, standardized by the Open Mobile Alliance (OMA)
- Instant messaging, standardized by OMA
- Push-to-talk, standardized by OMA
- Video and image sharing, standardized by 3GPP and GSMA

2.3 The rationale behind the MMTel standard

MMTel is designed with the aim of replacing fixed and mobile circuit-switched telephony. Characterized by quality, interoperability and reliability, the MMTel standard is a telco-grade service. It therefore plays a major role in fulfilling operators' objectives to implement an all-IP architecture to replace current circuit-switched telephony networks. Because MMTel is based on IMS, operators can perform network consolidation which lowers capital and operational expenditure (capex and opex), and MMTel gives operators a host of new multimedia services – in other words, opportunities to tap into new revenue streams.

With features like network-centric presence-enabled address book, chat, image sharing, video sharing, and file sharing, IMS and MMTel can compete with rapidly emerging internet communities and services, such as MSN and Skype. It also allows operators to remain service-aware, which includes taking part in service delivery and charging.

Currently, most PC users access the internet over a fixed broadband connection. Looking ahead, more mobile broadband connections are expected to be used. In this context the operator community will be able to use MMTel as a vehicle for offering new multimedia services. Furthermore, in contrast to the best-effort services offered by internet players, MMTel is a telco-grade service characterized by quality, interoperability and reliability.

3 Opportunities and benefits

3.1 Business opportunities

The MMTel standard has the power to support true network transformation and deliver significant business opportunities. All the pieces of the puzzle are in place with both the technology and a standard to facilitate migration to an all-IP network environment. The move to an all-IP environment will occur gradually and operators will make the move in incremental steps. This approach caters to the needs of operators seeking additional revenue streams from new services and applications as they become available.

As part of a phased strategy, a carrier might migrate its service offering from local-exchange switches to an MMTel and packet-based core network and application server.

MMTel's global standard for fixed, mobile, and cable operators, makes interoperability easier, particularly as developers and vendors can use the same standard protocol.

The MMTel standard makes it easy for operators and service providers to design, develop and market revenue-generating multimedia services and solutions for their subscribers.

To some extent, the network transformation is already underway, because many service providers are already offering mobile PBX, business trunking and residential broadband VoIP services. Service offerings of this kind immediately create new revenue streams, which in turn, drives further service development.

There is no single roadmap for the network transformation. Each carrier must decide how it wants to proceed. The migration process comprises many phases and the legacy network structures and architectures vary considerably from carrier to carrier. Still, the road eventually leads to an all-IP environment with MMTel support, giving operators the opportunity to realize the revenue-generating potential of their broadband infrastructure.

In addition, service providers can give residential users a rich mix of services that can be tailored to fit individual service packages.

3.2 Operator benefits

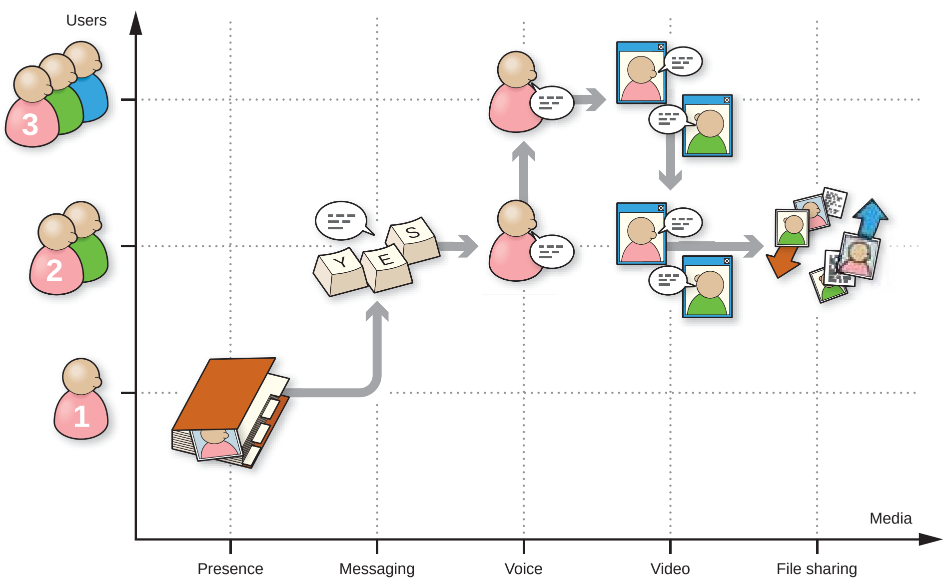
There are six main reasons why an operator should deploy MMTel:

- It gives users a complete and seamless services offering – whether their device is mobile or fixed.
- Users gain access to several multimedia features including video, chat and image sharing.
- The specifications of MMTel have been designed with the intent to replace all current fixed and mobile telephony solutions. This means that by choosing MMTel, operators can consolidate their networks and reduce capex and opex.
- Mobile-access technologies such as LTE and WiMAX do not support voice over circuit-switched technology [5].
- Global interconnect agreements for services between operators are key for the telecommunications industry. The standardized NNI enables operators to interconnect with one another, creating the potential for truly global, mass-market acceptance and profitability.
- Because MMTel is a standard made for the mass market it will support the development of inexpensive devices. Operators who choose the MMTel standard to maintain their position in the value chain can employ IMS and benefit from rapid and innovative application development. Being able to make a telephone call or send an SMS to anyone, and knowing that the communication will work across multiple operators and service providers, is a powerful concept that can be achieved only through global standards. The same holds true for all new services enabled by the MMTel standard.

3.3 User benefits

MMTel's basic communication capabilities allow a single SIP session to control media transfer. Two or more users can thus communicate in real time using different media components including [1]:

- Real-time voice; using legacy codecs such as adaptive multirate AMR or wide band codecs such as AMR-WB to further enhance the quality of the communication.
- Voice-synchronized real-time video transfer. The video stream can be simplex to realize a video sharing service or full duplex to realize a video telephony service. State-of-the-art video codecs such as H.264 can be configured to be used at high bit rates (>100Kbps) to allow for significantly higher video quality than the users are accustomed to when using the legacy circuit-switched video telephony services.
- Real-time text transfer (character per character) which can be used to realize a teletypewriter, a telecommunications device for the hearing-impaired.
- Non real-time text transfer (one message at a time) which is used to realize a session-based chat service similar to what MSN offers, for example.
- Image, video-clip and audio-clip sharing allowing users to share files that are displayed or replayed on the receiving terminal directly at reception using the specified file formats.
- File transfer to share a file of any type. The service is inherently flexible because it is based on IP transport and SIP/SDP session control. Users may add and remove different media types without having to stop or restart a session. The MMTel standard thus makes it possible for a single SIP session to control almost all services, see *Figure 2*. In the standard there are clearly defined operator-to-operator interconnect specifications. This means that users – belonging to different operators – can communicate with each other using all available multimedia services.



It is easy for the user to add or drop services and calling parties during an on-going communication session. Users can start a session with messaging, upgrade to a voice and/or video session, add other people to the session, send files, share pictures and more.

Figure 2: MMTel user scenario

The standard is backwards compatible, which means MMTel services can interwork with current fixed and mobile telephony standards. Among other things this means that an MMTel VoIP call can interwork with a Public Switched Telephone Network (PSTN) call, and an MMTel video call can interwork with a 3G circuit-switched video call.

Basic media transfer will not be sufficient for users of the MMTel standard. They will want to forward calls, identify a caller before answering a call and so forth. Therefore, attractive supplementary services will play an important role in creating a powerful and familiar telephony-type experience for users.

MMTel supplementary services apply to the entire communication session, regardless of active media components such as voice, video or text. One addition of note is the conferencing service, which enables users to add and remove call participants for the appropriate multi-party calls to be made.

4 MMTel standard: a background

MMTel is the new global service standard for real-time multimedia communication. Focusing on fixed access, ETSI and its working group TISPAN initiated work to standardize multimedia telephony in 2004.

TISPAN endorsed IMS as the core network, referring to 3GPP specifications for session control and basic communication. TISPAN also identified a need to extend the standards to include supplementary services enhanced with multimedia capabilities, which were not part of IMS. These services, referred to as "PSTN/ISDN simulation services", were completed at the end of 2005 [2].

A parallel activity, initiated in mid-2005, is being finalized within 3GPP to ensure that the same service can be used with 3GPP accesses to accomplish fixed-mobile convergence. 3GPP maintains its own service definition but refers to applicable TISPAN supplementary services [1]. In addition, to fulfill requirements for 3GPP accesses, the 3GPP standards specify media capabilities and media handling in greater detail [1,3]. MMTel was first introduced as a concept in 3GPP release 7 (R7), which was completed in December 2007.

In the following subsections, this paper describes the MMTel services in terms of the basic communication part (media capabilities) and supplementary services.

4.1 Interoperability for mass-market acceptance

Present-day communication (for instance, for telephony, SMS and MMS) relies heavily on global interconnect agreements. An equally important factor for MMTel is the use of a standardized network to network interface (NNI), which will enable the inter-operator use of MMTel services. For the successful mass-market acceptance of any new service, interoperability between operators, networks, and devices is key, and the NNI is a prerequisite for creating a mass market for MMTel (see *Figure 3*).

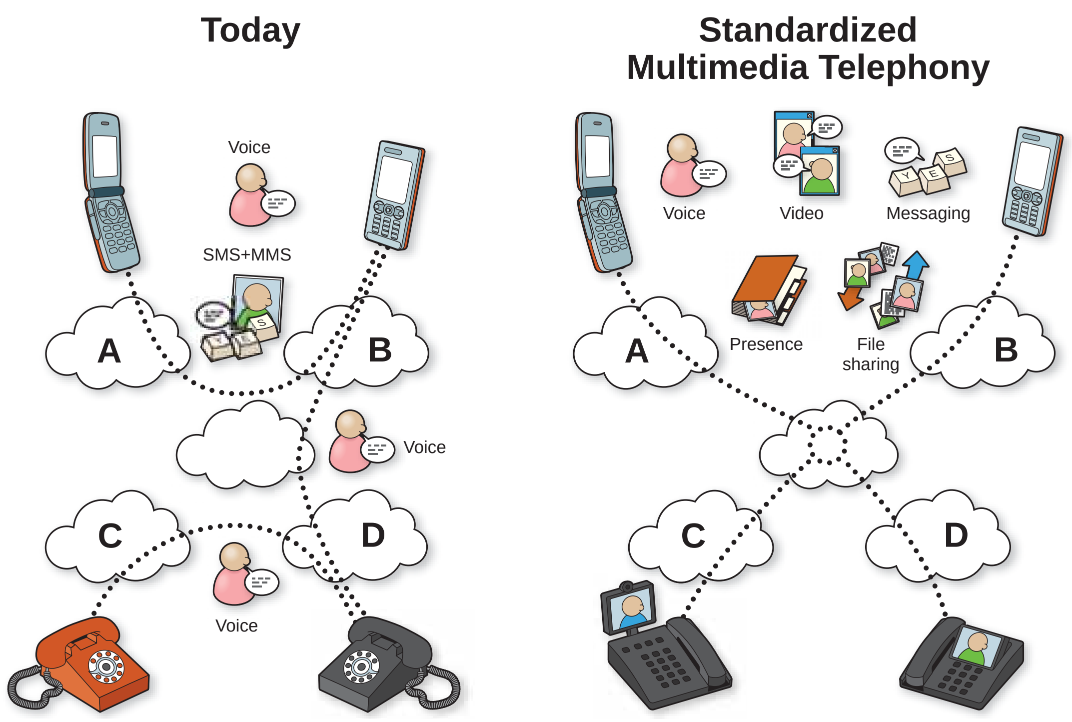


Figure 3: Looking forward: the MMTel standard enables operators to interconnect services

As can be seen in *Figure 3*, today interconnection is available for voice (fixed and mobile) and SMS (mobile). To some extent, but not in all operators' networks, interconnect for MMS is possible too. Because there is a standardized NNI interface in MMTel it is possible to interconnect all the multimedia features. This means it is possible for operators to become the world's largest multimedia community.

To date, MMTel has been positioned as a future mass-market service for real-time multimedia communication. However, it will also support the development of niche market services by being part of the 3GPP Communication Services (CoSe) concept [6]. This concept opens up Application Programming Interfaces (APIs) for MMTel and other standardized IMS service sets, including video sharing, push-to-talk and messaging. Third-party application developers may then reuse the APIs and certain other MMTel capabilities, such as NNI, in other applications, such as gaming and enterprise software (see "Communication Services Detailed Description" white paper).

4.2 MMTel, a single-session service

MMTel allows a single SIP session to control virtually all services. All available media components can easily be accessed or activated within the session. Compare this to a scenario in which a new session must be initiated for each new media component, such as voice or messaging. This requires control signaling from each session as well as active actions on the part of users.

Employing a single session for all media parts means that no additional sessions need to be set up to activate video, to add new users, or to start transferring a file. A typical scenario would begin with messaging and file transfer, and evolve into a voice and video session.

Although it is possible to manage single-session user scenarios with several conversations – for instance, using a circuit-switched voice service that is complemented with a packet-switched video session (refer to 3GPP Videoshare), a messaging service or both – there are some concrete benefits to MMTel's single-session approach.

A single SIP session in an all-IP environment benefits conferencing – in particular, lip synchronization, which is quite complex when the voice part is carried over a circuit-switched service and the video part is carried over a packet-switched service. In fixed-mobile convergence scenarios, the single-session approach enables all media parts of the multimedia communication solution to interoperate.

4.3 Standardized features of the media layer

The detailed specifications for legacy telephony services show how important media handling is for telecom-grade services. In these specifications, bit exactness and stringent requirements have been guiding principles for media coding and transport.

Legacy telecommunications standards and the traditions within IP transport are quite different, however. Although the same media codecs can be used in both cases, the very nature of IP transport introduces new challenges that, if ignored, degrade the performance of the service.

3GPP Release 6 (R6) included generic IP-based real-time communication but it did not address most of the IP-related media-layer issues. It specified codecs and transport protocols but did not describe a mechanism for securing the real-time performance and media quality of the communication session. The specification of MMTel in R7 gives a more thorough approach to media handling and processing. The 3GPP-standardized voice and video codecs for MMTel clients are:

- AMR and AMR wideband for voice; and
- selected profiles of the H.263, MPEG-4 Visual and H.264 video codecs.

The codecs use the Real-time Transport Protocol (RTP) formats defined by the Internet Engineering Task Force (IETF).

Apart from selecting suitable high-efficiency media codecs for real-time media, R7 addresses media layer adaptation, echo cancellation and noise suppression, coding and packetization guidelines for optimizing transport efficiency, PLMN/PSTN interworking, and media-related session negotiation.

One important addition addresses jitter management for voice. Traditionally, transport characteristics manifest themselves in the media flow through losses, either in the form of bit errors or the complete erasure of voice or video frames. In IP transport, degradation occurs in the form of full IP packet losses only, not single bit errors within a delivered packet. In addition, timing variations in data delivery occur (this is also known as jitter).

Jitter affects all real-time IP transport, but its impact on media varies. Voice, by its very nature, is not very forgiving of jitter or lost frames, because these are directly recognized as conversational degradation, and trigger decoder concealment operations. Video, on the other hand, will still show the most recent picture even if the updated data for the next frame is delayed. Therefore, the transition from circuit-switched transport (which uses dedicated resources) to packet-switched transport (which uses IP and therefore introduces jitter) is especially challenging to voice.

MMTel aims to maintain service quality by standardizing the minimum performance for voice. This level is met through requirements put on jitter buffer management.

4.4 Supplementary services

The MMTel services are wide-ranging and include supplementary services, regulatory services, and services that did not exist previously within PSTN/ISDN.

The addition of multimedia has brought many benefits to supplementary services including the communication diversion of multimedia services, ad hoc conferencing, and new services.

Communication diversion of multimedia services

Communication diversion is part of the PSTN/ISDN call-forwarding service. It supports Call Forwarding Unconditional, Call Forwarding Non Reachable and Call Forwarding Busy. It also supports the diversion of other multimedia according to relevant criteria such as presence, media, time, origin or anonymous. Users can create scripts that combine these criteria to express an individual diversion service.

Ad hoc conferencing

Another example to highlight is the ad hoc conference service. The service is invoked by subscriber A via signalling to conference service. The conference server distributes a message to the B-side including an accept or not accept option for the B-subscriber. Conference ID and PIN code are also transmitted but this information is hidden from the user in the SIP invite message in order to reduce complexity of the service.

If subscriber B accepts the conference call, an MMTel call (voice/video) is set up automatically via the conference server by using the hidden call information (the conference ID and the PIN code). Thus the call is completely handled from the MMTel client and the handling of PIN codes and conference IDs are automatic. The service becomes very easy and spontaneous to use compared to current existing conference services in PSTN/ISDN.

The Extensible Markup Language (XML) Configuration Access Protocol (XCAP) handles the user configuration of all MMTel supplementary services. Configurations are thus easy to manage using terminal menu systems.

Below is a list of the TISPAN R1 / 3GPP-supplementary services [4]. TISPAN and 3GPP have jointly optimized the set by adding services that are interesting for the future. These include user-configured multimedia identity, communication waiting, anonymous communication rejection, completion of communications to busy subscriber and advice of charge.

It is also possible to extend the supplementary service function – for example CDIV, by connecting it to internet services via interfaces that employ web services and the APIs of internet communities such as Facebook.

TISPAN R1 / 3GPP supplementary services

Supplementary service	**Basic functions**
Originating identification presentation (OIP)	Provides the terminating party with the asserted identity of the originating party.
Originating identification restriction (OIR)	Enables the originating party to withhold the information on its asserted identity from the terminating party.
Terminating identification presentation (TIP)	Provides the originating party with the asserted identity of the terminating party.
Terminating identification restriction (TIR)	Enables the terminating party to withhold the information on its asserted identity from the originating party.
Malicious communication identification (MCID) [Not used for 3GPP access]	A service for identifying malicious calls.
Communication diversion (CDIV)	Enables a user to have communications redirected by the network to another user. Variants of conditions include: unconditional, busy, no reply, not logged in, 'deflection' (=explicit forwarding during call setup phase).
Communication hold (HOLD)	Enables a user to suspend media within a session and resume later.
Communication barring (CB)	Allows a user to bar certain categories of outgoing and/or incoming communications.
Message waiting indication (MWI)	Enables the network, on the request of a controlling user, to indicate to the receiving user that there is at least one message waiting.
Conference (CONF)	Enables a user to participate in and control simultaneous communi cation with multiple users.
Explicit communication transfer (ECT)	Enables a transferring party A to transform two of that party's communications (for example an active communication to party B and a communication on hold to party C) into a new communication between party B and party C.

4.5 Regulatory services

TISPAN standardized regulatory services

Supplementary service	Basic function
Number portability	Users can keep their telephone numbers when they change operators.
Legal intercept	Government authorities may intercept traffic (both signaling and the media plane) without user awareness.
Emergency call	The IMS system applies location-based routing of emergency calls.
Carrier select	Users may select a carrier by adding a prefix before the phone number.
Carrier pre-select	Calls are routed via a specific carrier according to information in the subscriber profile.
Malicious communication identification (MCID)	A service for identifying malicious calls.

New services

With MMTel, one can create completely new services not previously available in PSTN/ ISDN. Examples of such services include:

Supplementary service	Basic function
Family service	A mini virtual private network (VPN) for members of a small group such as a family.
Contact service	Based on media type, time of day, presence information and so on, users can be contacted via the most suitable devices.

4.6 MMTel and circuit-switched interworking

At launch time, the initial MMTel service community will be small. Therefore, it will have to interwork with existing mass-market services. Simply put, if a user of MMTel cannot make a call to users of other, similar services, a mass market for this service will not develop.

Especially critical is interworking between the MMTel voice component and PSTN/ ISDN/PLMN circuit-switched voice. Standardization was initiated in 2005, and at present the interworking specifications from 3GPP and TISPAN provide a stable framework. Circuit-switched video telephony is a fairly widely deployed 3G mobile service.

3GPP tries to standardize interworking between MMTel voice/video and circuit-switched video telephony. One challenge is to translate between the single media stream of circuit-switched video to the two RTP-based media streams that transport voice and video.

Figure 4 shows the three standardized interworking scenarios between MMTel and circuit-switched services. There might also be a need for MMTel to interwork with local proprietary multimedia services. However, given the proprietary nature of such services, it is difficult to standardize these interworking functions.

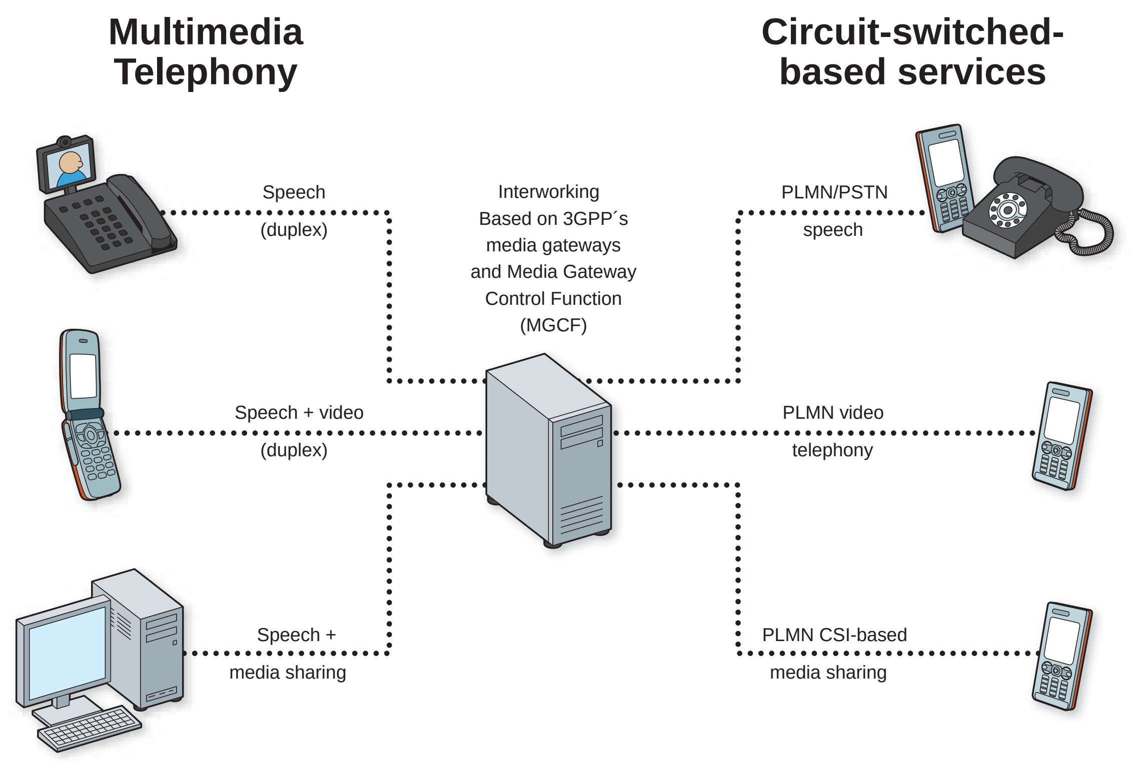


Figure 4: Important interworking scenarios between MMTel and standard circuit-switched services

5 MMTel devices

MMTel will support a variety of user equipment including IP phones, SIP-compliant PC clients, mobile phones, integrated access devices (IAD) and gateways. This equipment might fully support a variety of media including voice, video, image/file sharing and text, or it might support voice only.

The MMTel standard specifies a UNI, a set of codecs and media-processing functions in the devices (jitter buffer handling). This ensures the development of interoperable devices with predictable high quality. Because MMTel is a standard made for the mass market it will support the development of inexpensive devices.

5.1 Fixed access

For fixed access, it is assumed that IADs will dominate the market for single-media (voice) services, followed by PC clients and IP phones for multimedia use.

The common IP telephony box or ntegrated access device (IAD) that IP telephony customers use to connect a PSTN phone to a broadband access connection is one example of customer-premises equipment (CPE). Ordinarily, an IAD is composed of three parts: an Ethernet card, a SIP user agent and an interface to the PSTN phone. Occasionally, it also includes an xDSL modem.

5.2 Mobile access

Mobile devices for MMTel will be available in many different shapes and forms. It is anticipated that the vast majority of these will be multimedia-enabled.

The application on the devices that implements MMTel is called a client. Multimedia clients can be divided into two groups: downloadable and native.

Downloadable clients will be deployed when MMTel is introduced as a stand-alone complement to the circuit-switched mobile telephony service. In this case the user interface of the MMTel client will be separate from that of the native circuit-switched mobile telephony client.

Native clients, on the other hand, will be on phones when they are delivered from the factory. These clients are tightly integrated into the phone user interface for circuit-switched mobile telephony; in other words, users perceive only one telephony application. Based on current availability and user or operator preferences, the devices will select the best access.

Eventually, it is anticipated that the majority of clients for mobile devices will be native. The reasons for this are two-fold:

- MMTel is being introduced with fixedmobile convergence in mind. The goal is to phase out circuit-switched telephony. Therefore, to make the transition as smooth as possible, MMTel must be tightly integrated with the circuit-switched telephony application.
- Because the handling of media streams for real-time services, such as telephony, is so computationally demanding, it makes sense to implement this function at a low level and to employ native codecs, such as AMR, which are already part of, and have been optimized for, all mobile platforms.

6 Conclusion

The new MMTel standard will play a key role in the IMS multi-service ecosystem. For reasons of interoperability and industry acceptance, only standardized solutions can power a market with a multitude of interconnected operators and service providers.

In this context, MMTel is not just another IMS service but rather a global standard for the next evolutionary step of telephony service – that is, real-time multimedia communication, which will eventually replace fixed and mobile circuit-switched telephony services.

MMTel combines quality, interoperability, reliability, efficiency, regulatory services and supplementary services with the rich media and dynamics of internet community-based communication.

It can be launched over any access network that uses the IMS service engine. The service is designed to address any limitations caused by mobile access. As a result, it will run over mobile as well as fixed access.

As the network transformation from circuit switching to an all-IP environment gains momentum, the MMTel standard will give carriers and operators a rich, powerful and integrated service offering that can create almost immediate returns, in particular as consumers are already asking for these services.

7 Glossary

3GPP: 3rd Generation Partnership Project

AMR: Adaptive multirate codec

API: Application Programming Interface

CoSe: Communication Service

CPE: customer premises equipment

CS: circuit-switched

ETSI: European Telecommunications Standards Institute

HSPA: High-speed Packet Access

IETF: Internet Engineering Task Force

IMS: IP Multimedia Subsystem

LTE: 3GPP Long Term Evolution

NGN: next-generation networking

NNI: Network-to-network interface

OMA: Open Mobile Alliance

PS: packet-switched

SIP: Session Initiation Protocol

SDP: Session Description Protocol

UNI: User-to-Network interface

VoIP: Voice-over-IP technology (which enables users to transmit voice calls via the internet using packet-linked routes. VoIP is also referred to as IP telephony)

WiMAX: Worldwide Interoperability for Microwave Access

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