3GPP S2 **Tdoc S2-000505**
Tokyo, Japan

TITLE:	**A Comparison of H.323v4 and SIP**
DATE:	**5th January 2000**
SOURCE:	**Nortel Networks**
ABSTRACT:	This contribution compares and contrasts SIP to H.323v4 to help aid operators and vendors in the selection of a single least common denominator control protocol for the ps "domain" or perhaps more appropriately "plane" of UMTS Release 2000. The format anticipates the concerns, in the form of questions, which may arise from 3GPP members.
RECOMMENDATION:	For facilitation of protocol selection.

Note: Every effort has been made to ensure the completeness and accuracy of this document. Any inaccuracies or discrepancies are unintentional and should be brought to the attention of Nortel Networks for clarification/correction.

1 Background

Two standards have emerged for signaling and control of VoIP telephony: ITU-T H.323 and the IETF Session Initiated Protocol (SIP). These protocols, although resulting in the same end-user service (telephony), differ in the approach to providing signaling functions. H.323 is based more on a monolithic bloc derived from H.320 for traditional of the traditional circuit-switched ISDN multimedia, and SIP favors a more lightweight approach based on HTTP. In order to provide telephony services, the UMTS Release 2000 network requires a call control protocol to initiate and manage connection establishment,

1.1 Key Motivators in Network Protocol Reconsideration:

- A single call signaling protocol is desired with distributed functionality across the UMTS Release 2000 network elements.
- A single call control signaling protocol enhances the ability to provide a virtual Home Environment for telephony and multimedia services.

1.2 Protocol Selection Options:

Option A: Deploy a UMTS Release 2000 architecture using only the ITU-T H.323-based Call Control Protocol.

Option B: Deploy a UMTS Release 2000 architecture using only the Session Initiation Protocol (SIP) based Call Control Protocol.

1.3 Scope of the Requested Effort:

The scope of the analysis is for the vendor community (and Carriers) to come up with a broad comparative analysis of call control protocol options. The analysis should aim to answer the several questions that are listed in the following section of this document. The criteria for the analysis are:

- **Time to market**
- **Estimated quantification of the work effort required**
- **Identification and Qualification of the Impact on Network Elements**

The analysis should conform broadly to the assumptions outlined in this ad-hoc document.

1.4 Key Assumptions for the Analysis:

- Standardization completed by 3GPP Release 2000 (EOY 2000).
- Supporting requirements and assumptions set forth in UMTS release 2000 (It is recognized that this work is still in progress).

2 H.323/SIP Comparison: Important Factors to Consider When Choosing a Protocol

For responding, consider H.323 protocol version 4. Backwards compatibility to older versions need not be considered.

Does either call control protocol (H.323 or SIP) provide a significant advantage over the other in terms to capability, time-to-market, complexity, operations, administration, management, intersystem operation, compatibility with other technologies, or evolution? Please quantify, and consider the following:

2.1 Complexity

2.1.1 Message set comparison

We consider air-link optimized call flows for setting up a mobile to PSTN call using SIP and H.323v4. The number of messages exchanged and typical message sizes are compared. A functional comparison is made of the various messages in the SIP and H.323 protocol suites.

We have assumed that a mass deployed default audio codec is known to both sides and thus a graceful media negotiation mechanism is not necessary.

The transport layer for both SIP and H.323 is assumed to be UDP (again for fair comparison). However, since the last message for call set-up (200 OK for SIP and CONNECT for H.323) is related to a billable event, the receipt of these messages must be acknowledged.

In the SIP case, the 200 OK message is always acknowledged. For H.323, we have chosen to utilize the ANNEX E mixed mode feature of H.323 to provide an ACK message for the H.225 CONNECT message.

The flows assume that pre-answer announcements can occur.

2.1.1.1 Optimized Call Setup Flows

Figures s1 and s2 shows the call flows for SIP and H.323, respectively.

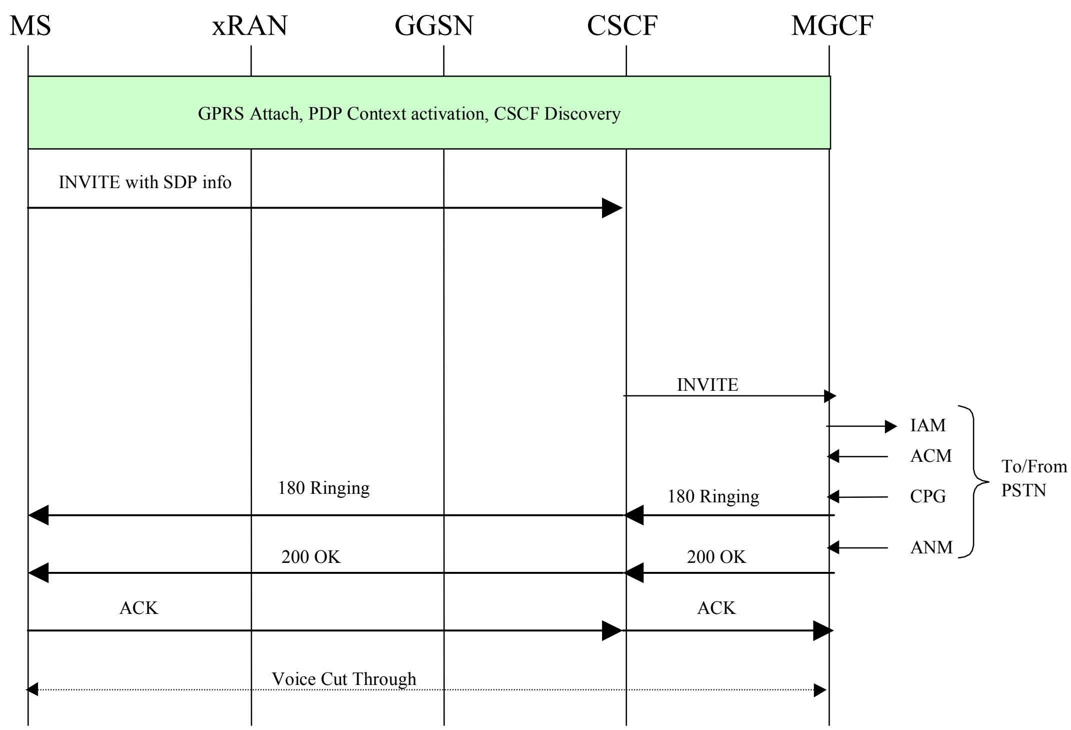


Figure s1. Partial call flow for a mobile-PSTN call using SIP

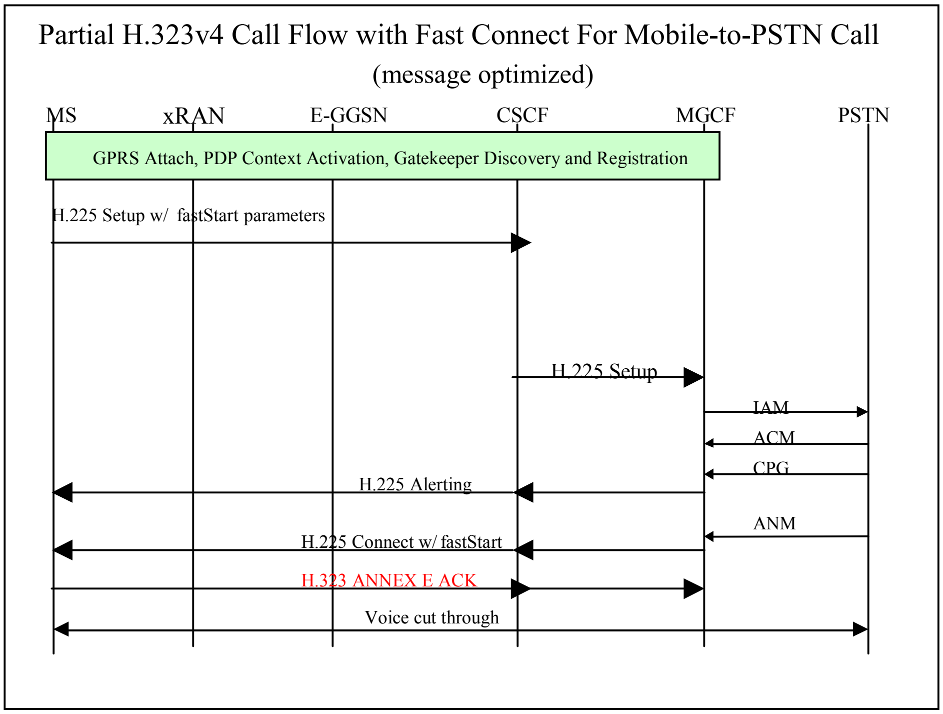


Figure s2. Partial call flow for a mobile-PSTN call using H.323

Figure s2 employs the most optimal method for call setup that H.323 could provide that matches SIP reliability. It employs the use of the H.323 Annex E mixed mode transport protocol; however. It should be noted that the use of Annex E as a transport layer is not well established and issues such as as gatekeeper to gatekeeper acking are still in question.

2.1.1.2 Message Size Comparison

Table st1 compares the number of message exchanged and typical sizes of the call control messages exchanged. For H.323, the messages are already ASN.1 PER byte-aligned encoded and hence, compressed. For the SIP messages, we have applied both a simple tokenization technique and a text compression algorithm similar to LZ77. We recommend that only the tokenizing method is used for last hop wireless specific signaling.

	SIP	**H.323v4**
# Of Messages Exchanged	4	4
Typical Message Size (bytes)	INVITE : 231 OK : 271 Ringing: 165 ACK : 171	Setup : 390 Alerting : 97 Connect : 280 ANNEX E ACK: 32
Total bytes	838	799

Table st1 : Message set comparison based on call flows

The following table compares the results of the various compression schemes applied to the original SIP messages of our example trace. All sizes are in bytes.

SIP messages	original size	size after simple tokenization	size after tokenization & LZ77 compression
INVITE 180 Ringing 200 OK ACK	346 217 440 237	297 180 396 218	231 165 271 171
Total	1240	1091	838

Table st2 : Comparison of the compression schemes on SIP messages

2.1.1.3 Functional Message Set Comparision

Functions	SIP	H.323
Registration	REGISTER, ACK	RAS: RRQ, RCF, RRJ, GRQ, GCF, GRJ
Admission	INVITE	RAS: ARQ, ACF, ARJ
Call Initiation	INVITE	H.225 Setup
Capability Exchange	SDP, OPTIONS, NOT SUPPORTED	H.245 Open Logical Channel (Can be embedded in H.225 for Fast Start). H.245 Terminal Capability Set and Ack can be used for slow-start.
Resource Allocation	No specific signaling	Some consider ARQ, GRQ a crude form of resource reservation at admission. For per-session allocation there is No specific signaling.
Status	Any response/acknowledgement messages (1xx-5xx) ex. O.K., Ringing, Progress, ACK	Alterting, Progress, Call Proceeding, Connect
Teardown	BYE, Ack	Release Complete
Reliability	Has timers and session methods/messages to achieve reliability. SIP is reliable over UDP. SIP can also bundle requests and responses.	Relies on TCP, SCTP or Annex E. H.323 over UDP alone is not reliable. H.323 has timers as well.

Conclusions:
In terms of message set comparison for complexity SIP and H.323 are similar. SIP implements an ACK to the OK (answer) message; whereas, H.323 was designed to rely on either TCP or Annex E for this function. There may still be a few issues with the proxy of the Annex E ack between gatekeepers. Annex E is not widely deployed and there are many variants. We do not believe it’s interoperability has been significantly tested in open interoperability events.

In terms of pure message size SIP and H.323 using TCP are roughly the same. SIP has 4 basic messages in the session setup. H.323 fast-connect has only 3 but must rely on the transport layer for reliability, increasing the transport level messaging and bytes transmitted. If Annex E mixed mode with Gatekeeper proxying of transport ACKs is used in conjunction with H.323 for reliability then H.323 fast connect has less total bytes transmitted than SIP even after applying compression.

H.323 over pure UDP is not currently reliable. H.225 could be altered in H323v4 timeframe to have a session level ACK message to the Connect in order to provide reliability over UDP.

2.1.2 Encoding, parsing and generation

H.323 signaling protocols such as H.225 and H.245 use ASN.1 byte-aligned Packed Encoding Rules (PER). This involves the use of encoding and decoding code generated by an ASN.1 compiler from a text based ASN.1 syntax definition file.

It should be pointed out that for very simple messages that employ ASN.1 PER, direct generation of the encoding can be used. This greatly increases development time and usually limits parameter optionality. In other words, it defeats a primary reason for using ASN.1 in the first place.

This does eliminate the need for a separate encoding process but can only be used under very restricted circumstances.

This analysis assumes that encoding is performed as a separate process from generation.

SIP is a pure text based protocol patterned after HTML. SIP generation involves no separate encoding overhead. Tokenized compression does not employ any extra processing. Total compression mechanisms would involve extra CPU processing. On the receiving end SIP employs a text parser similar to the parsers used in web browsers to determine the values of fields identified by tags.

2.1.2.1 Encoding and Generation:

Encoding and generation are separate processes in H.323; whereas in SIP, there is no separate encoding process; therefore, it is safe to say that there may be close to 2 times the number of operations performed per field in H.323 as there are in SIP. This is due to each field having to be looked at once for generation and once for encoding (i.e. twice the number of reads and twice the number of writes into the correct stream format).

We should point out again; however, that direct combined generation and encoding can occur on the H.323 signaling. In this case SIP and H.323 are roughly equivalent. But again, this has a dramatic affect on development time and this method may not be able to be used at all depending on the optionality of the fields contained within the message.

SIP message generation employs string writes of headers and string writes of values corresponding to these headers. It is fairly straightforward and easy to program making it less complex. Many processors sold today are optimized for string operations using a single clock cycle.

If tokenizing compresssion is employed on the SIP messages which involves simple substitution of the header fields and other known reserved words with tokens, no additional overhead is incurred. It is a part of generation. In addition, this form of compression does not destroy the extensibility mechanisms of SIP. If the header or field does not have a matching token, the header or field is not tokenized. They are left as is.

We also need to point out that PER aligned encoding employs the use of a preamble bit map to describe characteristics of the records and primitive types contained thereafter. The encoding of the preamble for each field that employs optionality and extensibility is deterministic upon the values of the fields at run time. Having to deal with these bit operations during the encoding process incurs a significant amount of additional per-field processing overhead on H.323 that does not exist with SIP.

This processing translates to up to 5 extra write operations per field.

Conclusion: SIP compression/generation overhead is less than H.323 ASN.1 PER encoding/generation. This will have an effect on the mobile stations' power budget as well as the overall complexity of the CSCF.

2.1.2.2 Decoding and Parsing:

H.323 signaling protocols employ ASN.1 PER compiler generated decoding procedures to decode incoming messages.

SIP employs a text parser, which identifies fields via tags and values thereafter separated by delimiters. Comparisons with each TAG parsed must be string compared with a list of all of the available headers. Hashing functions are means of optimizing this process. SIP has been designed with TAG values that facilitate this hashing optimization. The number of operations to perform this hash is expected to be on average 3. SDP is parsed in a similar manner.

Both methods are order N operations. SIP parsing involves byte and string comparisons; whereas, H.323 involves per-field bit map operations as well as string comparisons. Many of the same points above apply to decoding as well for H.323.

H.323 decoder processing translates to 5 read and comparison operations per field.

If simple tokenizing is employed to reduce SIP tags to 1 byte tokens, the affect on processing overhead is actually reduced as tokenizing takes only one hash operation.

Conclusion: SIP parsing overhead is about the same as H.323 ASN.1 PER decoding. It's basically as wash. SIP parsing overhead used in conjunction with tokenized compression is less than H.323 ANS.1 PER decoding. This will have an effect on the mobile stations' power budget as well as the overall cost of the CSCF.

2.1.2.3 Overall Conclusion and Ramifications:

SIP is less complex than H.323 for parsing, decoding and compression, encoding and generation.

CPU overhead is implementation specific and could vary accordingly.

Care should be taken in the selection of the processor of mobiles, MGCFs and CSCFs if SIP is chosen as the least common denominator. Processors should be optimized for string operations.

Another point that should be made here is ease of programming and vendor selection. As H.323 is complex with regard to encoding/decoding, the number of competing vendors and operators for the source and services which employ the protocol may be reduced.

2.1.3 Debugging

SIP debugging is quite simple as the signaling is based on text and tags similar to HTML. This means that SIP debugging is self-adapting as standards and new services are defined and deployed. No work or extra time is needed to update the debugging tools.

H.323 debugging requires specialized tools that must adapt to the standards as they change.

Conclusion:
The advantages detailed above favor SIP and the affect is reduced TTM intervals as well as reduced complexity for development.

2.1.4 Reuse of existing code and procedures

H.323's signaling methods and flows were patterned upon the ISDN multimedia standards. However, as the standard progressed, most of the fields that H.225 and H.245 required (ex. RTP port) were added and have little similarity to the ISDN world. The H.323 standard did base their signaling off of Q.931 but for a quite different purpose. H.225.0 is used to set up calls completely independently of media connections, contrary to ISDN. Furthermore, H.225.0 only includes a subset of the Q.931 protocol, and very often, the meaning for each common element is completely different. The original idea was to facilitate interworking, but in retrospect, it seems to have created more confusion in the end.

Most vendors have implemented H.323 using generic H.323 stacks (there are a very limited number of interoperable and maintained stacks. These stacks have little in common with existing ISDN stacks.

SIP comes out of the web world. One could possibly re-use or pattern a SIP parser off an MGCP or HTML parser. As SIP has been designed in a modular fashion, it was designed to be a defacto push technology for the web. So one could say that it was designed to re-use and interoperate with all of the web technologies and even some H.323 technologies such as: HTTP, ACAP, LDAP, FTP, RTP, RTSP, T.120, etc.. providing the session layer and user location functions seamlessly for appropriate application session establishment without any need for changes to these other protocols and applications.

Conclusion:
Some of the issues related to PSTN interworking have been thought-out for a longer period of time in H.323 than with SIP (ex addressing received earlier attention in H.323, tones and announcements, etc.). However, the underlying principles are ”portable” (and were indeed ported over from the ISDN). The porting of these principles to SIP is under way right now largely due to the interest of operators and vendors that recognize the long-term benefits.

If you see more future generated revenues based upon the fast integration of web technologies and see the TTM value and flexibility that SIP provides operating in control of both circuit-based endpoints as well as IP endpoints, then SIP wins.

We do need to mention the HTTP URL parameter that will be part of H.323v4. This parameter significantly evens the playing field for web-based applications.

2.1.5 Service and feature implementation and protocol interactions

Please refer to the previous and following sections.

2.1.6 Options and methods for implementing services that are available in the protocol

Both protocols can be made to support the IN model, the CGI model, 3rd Party redirection, feature code signaling, etc..

H.323 has in the past been geared towards a particular set of services: those defined in H.450 (itself derived from QSIG), T.120 data applications and basic voice/video setup. We now see that H.323 can be used to push an http URL in version 4.

SIP was designed to be the defacto push and presence application. It was designed to use SDP as it's media description language and which allows for many of the new features already such as pushing a URL. As these protocols were built to interwork with web technologies from the start, it basically has an advantage over H.323 in this respect. Many of the web-related functions are already built in; whereas, H.323 will likely have to add new fields such as an HTTP URL parameter. SIP and SDP can be used to facilitate T.120 applications as well.

Conclusion:

H.323 seems to have to add specific fields or new values for parameters in the signaling in order to allow newer web related services. This means that operators will be waiting on their vendors intervention to provide these new related web services more often than with SIP. A prime example of this is the addition of an http URL parameter in H.323v4.

There is no need to extend SIP or even SDP to achieve the same functionality. Due to it's modular design, it does and always has done this function transparently.

2.1.7 Interworking with the PSTN

This is a problem that has to be ironed out period. The signaling protocol has little to do with it. Most of the issues being debated now deal with inband streaming and QOS interactions. The signaling protocol employed whether it be foo or bar is irrelevant.

One could try to make the case made in section 2.1.4 **Erro! Argumento de opção desconhecido.** but that's the end of it. SIP used to have a bit of leg up by providing for ISUP encapsulation. H.323 has now included a similar mechanism in H.323v4.

2.1.8 Implementation in UMTS Release 2000 network elements and user devices

SIP stack size is less than H.323. This has the effect of reducing the cost (memory) of the devices.

We believe SIP has a significant TTM advantage due to it's text-based debugging and modular design.

2.2 Extensibility

2.2.1 Compatibility among versions (built into the protocol?)

Both suites of protocols have version identifiers that can be used to control extensibility mechanisms based on version.

SIP: SIP does not have explicit requirements for compatibility among versions. Unknown/unsupported headers are ignored by default. This reduces code size and protocol complexity. Also, this provides flexibility in terms of developing/evolving features and makes encoding/decoding clean and concise.

An adverse effect of this may be that features supported by the older versions may not be supported by newer version. Some may see this as an advantage, however; baggage gets dropped over time and code size is kept in check. Keep in mind that this perceived adverse affect is optional. You can keep supporting the past as long as you wish or drop it at any time you deem fit.

The REQUIRE header allows for an end system implementation to require a specific header. The PROXY REQUIRE and a proposed SUPPORTED header in addition to the REQUIRE header allows for intermediary proxies, registrars and redirect servers to require a specific header.

H.323: H.323 requires full backward compatibility. This ensures continuous support of existing features. Note that, although standards explicitly specify backward compatibility, vendors may chose to support only last 2 or 3 versions. This may reduce size of messages and protocol/ implementation complexity. Due to enormity of H.323's baggage already, we recommend that support of old versions prior to H.323v4 not be supported.

2.2.2 Feature evolution

SIP: Using SIP, feature may be evolved by extending or defining new SIP header information. Current SIP RFC defines default headers and some extensions. New extensions can be added as a part of separate RFCs. Refer to "Ability of carriers to define own features and services" section of "Extensibility" of this document for details.

H.323: H.323 defines NonStandardParameter structure to extend vendor specific (proprietary) features. For extension/modifications to NonStandardsParameter, refer to "Ability of carriers to define own features and services" section of "Extensibility" of this document. If changes are made to existing capabilities or control message parameters other than NonStandardParameter, new version of corresponding specifications may need to be issued. Additionally, new features could be implemented using the new H.450.1 generic functional protocol.

2.2.3 Ability for carriers to define own features and services

SIP: In general, SIP defines methods, default request/response headers and status codes. Service definition is included in the header itself. The extensions to existing headers may require changes in RFCs where original header is defined. But the new headers can be easily added by having definition in separate RFC or other standards process.

It provides a hierarchical namespace of status and also for features/services. In this way, new features/services can be added easily without changing message contents. For example, new (perhaps proprietary) features/services can be introduced in two ways 1). By registering new features/services with Internet Assigned Numbers Authority (IANA). 2). By deriving hierarchically from the feature owner's Internet domain name, giving hints to where further information might be found.

Also, SIP client can inquire about SIP server abilities first or proceed under the assumption that the server supports the extension and then back off if the assumption was wrong.

Currently, SIP does not employ a traditional graceful mechanism for two-way service negotiation. It does have two ungraceful methods, however; "not supported" and the more pessimistic "options" method. The thinking here was that negotiation will likely not have to be done on most session transactions. To tax the state machine and increase messaging because of this seemed sub-optimal. The MMUSIC WG in the IETF has an action to resolve some general issues with the simultaneous asymmetric multiple media negotiation. We expect them to make minor enhancements to SDP to resolve the issue as opposed to SIP, though.

Organization headers can also be used in conjunction with the contact and record-route headers to implement operator specific enhancements. If the proxy require is not used and given that operators have control over the source for their application proxies as well as the source in the UA's, new services can be deployed over an existing proxy based wireless capable IP environment without any intervention of vendors.

The built-in extensibility mechanisms, ASCII nature of SIP and it's modularity will significantly impact the ability of operators to define and deploy their own services without intervention from vendors in short market windows.

H.323: In H.323, non-standard capabilities and control messages may be issued using the NonStandardParameter structure defined in ASN.1. This NonStandardParameter structure consists of vendor codes (NonStandardParameter identifiers) and data associated with the particular code. Note that the type of data is Octet String and may be unlimited. OBJECT IDENTIFIER can be used to identify vendors, service providers or organization and are also hierarchical (but not maintained by IANA).

Since vendor codes are defined as a part of ASN.1 definition, additional vendors need to request change in specifications. Definition of services and some other parameters (other than NonStandardParameter) in ASN.1 may be extensible and hence modifiable. If additional parameters are required, they can be added to existing version of H.323 as optional parameters in the end of message structure.

Due to the complexity of defining new parameters in H.323. Deployment of new services will likely result in the operator waiting for their vendors to add the new parameter.

2.2.4 Modularity of the protocol to allow for the easy evolution to new services and features

SIP: SIP encompasses mainly user location, registration and basic session signaling. For advanced services/features, other functions like capability exchange, service discovery, QoS, directory access, conference control are essential. All these functions are resided in separate protocols and can be used with SIP without making any changes to the SIP protocol. In addition, SIP's extensibility allows for new headers to be added without changes to pass-through proxies or even client user agents.

H.323: H.323 is an umbrella feature containing vertically integrated sub-protocol suite of H.225, H.245, H.450, RAS, Q.931 etc. Hence, from feature/service perspective, there is no clean separation of these sub-protocols. This results in to more interactions between its sub-protocols. Also, most of the services are in-built and intertwined between more than one sub-protocol.

2.2.5 Ability to work with existing and new multimedia codecs

SIP: SIP uses Session Description Protocol (SDP) to convey the codecs supported by an endpoint in a session. Codecs are identified by string names that can be registered by any person or group with IANA and then used. This means that SIP can work with any codec and other implementation can determine the name of codec and contact information for it, from IANA.

H.323: The GenericCapability type in H.323 allows new codecs to be specified in such a way that a new version of H.245 syntax does not need to be issued.

2.2.6 Third party call control mechanisms

This mechanism can be defined as the ability for a party to set up a call between other parties without necessarily participating in the call e.g. a secretary dials for a manager, operator service etc.

These mechanisms allow a third party to instruct another entity to create and destroy calls to other entities upon requested by server. Here, a call control protocol like SIP and H.323 can be used between the server and third party. As the third party executes the instructions, status messages are passed back to the server. This allows the server to take further actions based on some local program execution.

SIP: SIP can support this type of mechanisms by using "also" header in requests and responses. In addition the contact and record-route headers allow for a 3rd parties or home elements to always be in the loop.

H.323: There is no standard comprehensive way to do third party call control in H.323. The FACILITY redirection feature can be used to “deflect” a call at setup time to another location.

2.3 Scalability

2.3.1 Support for large numbers of domains (wide area addressing, user location, etc.)

H.323: The initial intent of the protocol was for the support of LANs, so it was not inherently designed for wide area addressing. The concept of a zone was added to accommodate wide area addressing. Procedures are defined for "user location" across zones for email names. Annex G defines communication between administrative domains, describing methods to allow for address resolution, access authorization and usage reporting between administrative domains. In multi-domain searches, there is no easy way to perform loop detection. Performing the loop detection can be done (using the PathValue field), but introduces other issues related to scalability (e.g. how to define the field value and how the value should change when the network configuration changes).

SIP: SIP inherently supports wide area addressing. When multiple servers are involved in setting up a call, SIP uses a loop detection algorithm similar to the one used in BGP, which can be done in a stateless manner, thus avoiding scalability issues. The SIP Registrar and redirect servers were designed to support user location.

In both SIP and H.323, the burden of scalability is in the SIP server or Gatekeeper, the underlying transport layer, and in the way they can communicate with their peers. They can both accommodate different topologies (hierarchical, flat, etc.). They can both make use of DNS, directories, internal translation databases or other location and translation mechanisms which can facilitate global deployment.

2.3.2 Ability to handle large numbers of calls

This issue is primarily implementation/deployment specific. Theoretically, a stateless implementation allows a gateway/server to support a larger number of calls. Both protocols can or can be made to support n to n load balancing. The complexity and statefullness of the distribution nodes and endpoints is dependent upon the transport protocols used as well as the translation mechanisms employed.

H.323: Although not initially supported, H.323 call control can be implemented in a stateless manner. A gateway can use messages defined in H.225 to assist the gatekeeper in performing load balancing across gateways.

SIP: Call control can be implemented in a call stateless manner. SIP supports n to n scaling between U/As and servers. SIP takes less CPU cycles to generate signaling messages; therefore a server could theoretically handle more transactions. SIP has specified a method of load balancing based upon the DNS SRV record translation mechanisms. This method may not be appropriate for a carrier grade environment. For instance it does not handle local translations. There is a current proposal to alter DNS for this task, however (local DNS).

2.3.3 Maintaining of call states (stateless or stateful) effect on scalability

This issue is also implementation/deployment specific. However, typically, scalability is reduced when a server must maintain call states as a given call must use the same server while the call is active or the call states must be transferred if a server goes down. Stateless processing allows a given transaction to be processed by any server providing the required functionality, since theoretically no transient information need be maintained by the server.

H.323: Supports both stateful and stateless processing. Scalability is reduced in the stateful mode as the same server (or shared memory) must be used while the call is active. Most existing carrier grade implementations of H.323 gatekeepers are designed to be call stateful and employ hot-sparring technologies in order to meet carrier grade requirements. This increases the complexity of these devices. There is nothing in the H.323 specifications that prevent gatekeepers from being implemented using call-stateless N+1 redundancy and load sharing technologies, however.

SIP: Transactions in servers and gateways can be stateless or stateful. The stateful mode decreases scalability as the same server (or shared memory) must be used while the transaction is active. Most current implementations of SIP proxies are designed to be call stateless and rely on N+1 redundancy and load-sharing technologies to meet carrier grade requirements. This reduces the complexity of these devices. There is nothing in the SIP specification that prevents the use of hot-sparing technologies from being used, however.

2.3.4 Elements that must maintain states

This issue is primarily implementation dependent.

H.323: In a call-stateful implementation, the terminal, the Gatekeeper and any Gateways must maintain states. In a call-stateless implementation, the terminal must maintain states. Most current H.323 gatekeeper implementations are designed to be call stateful.

SIP: In a call-stateless implementation, only the terminal (i.e. endpoints) must maintain states. In a call-stateful implementation, the terminal, the proxy server and perhaps the redirect and registrar servers (if implemented separately) would maintain call states. Most SIP proxy implementations are designed to be call-stateless.

2.3.5 Signaling message processing

This issue is implementation dependent; however, 2 factors greatly influencing the scalability are whether call states are maintained and the use of UDP versus TCP to transport the signaling messages.

No connection states are required in UDP, so scalability is improved. Stateless calls using are more scaleable than stateful calls.

SIP can be based on UDP and stateless call processing without any loss of reliability. H.323 requires either TCP or the transports defined in H.323 Annex E to have the same level of reliability.

The encoding/decoding/parsing mechanisms also have an impact on scalability due to consumption of CPU processing.

H.323: Messages are ASN.1 encoded, using aligned PER.

SIP: Messages are text based. Same message set is used between services and call control entities.

2.3.6 Conference sizes, conference control (centralized vs. distributed)

The implementation of a central control point for conferences reduces the size and number of supportable multiparty conferences. Distributed conference processing scales more easily to larger numbers and sizes of multiparty conferences.

H.323: H.323 conferencing was initially based on a centralized conferencing control mechanism requiring an MC. To support larger conferences, H.323 allows an application layer multicast conference concept. H.245 provides feedback during conferencing allowing receivers to control encodings, transmission rates and error recovery. This mechanism does not easily scale for multipoint conferences. However, H.332 extends H.323 for ”loosely-coupled” conferences.

SIP: SIP conferencing is originally based on distributed conference control, thus larger conferences can be easily supported. SIP uses RTCP for conference feedback. Since RTCP is also distributed this feedback mechanism easily scales to support larger conferences. SIP can also work in conjunction with centralized conferences. While SIP can be used as the signaling protocol to implement an MC function (SIP robot or the UA itself), SIP does not provide any specific method of floor control, etc..

2.4 Resource Utilization and Management

2.4.1 Resource required during the call

The resources required during the call is addressed in two parts - (i) resource required during call set-up in terms of air-link bandwidth, and (ii) resource required in terms of CPU power and memory in the client and the server.

Air-link bandwidth required during call set-up:

The air-link bandwidth required during call set-up is related to the number and size of messages exchanged. It is expected that there will not be any spectacular difference in the number of messages and the total number of bytes exchanged over the air link for a typical call set-up using SIP or H.323. Comparison for a typical mobile-PSTN call set-up scenario is given in Tables st1 and st2.

Resource required in terms of CPU power and memory in client and server:

H323V4 : H.323v4 messages can be storage efficient if packed encoding rules (PER) are used. H.323v4 is a fairly complex set of protocols and includes H.225 for call signaling, H.245 for call control, H.332 for large conferences, H.450.x for supplemental services, H.235 for security and encryption and H.246 for inter-operability with circuit-switched services. Many services require complex interaction between these sub-protocols. The protocol state interactions and management in the client is directly related to these interactions. The H.323v4 server can be either stateful or stateless as TCP or UDP can be used for the transport (this is different from call state). Usage of UDP or mixed Annex E transport significantly reduces the memory requirements. Call state information needs to be kept in the Gatekeeper when a "Gatekeeper Routed" call model is used. If a "Direct Routing" call model is used, then the Gatekeeper may be used only for Registration, admission control and address translation.

SIP : Textual formats used in SIP are less space efficient than ASN.1 PER. On the other hand, the SIP parser is fairly simple and can be written in less than 500 lines of code, which is much less than a general ASN.1 decoder.

SIP/SDP are also less complicated than H.323 suite. This implies that maintenance of protocol state information at the client and server are expected to be less burdensome than H.323v4. In SIP, the transaction through proxy-servers may be either stateful or stateless. In case of stateful servers, the requirements for maintaining state information are expected to be roughly equivalent for both SIP and H.323v4.

2.4.2 Resource minimization

In the wireline case, there are three proposals dealing with this issue all using SIP: (i) DOCSIS proposal (PacketCable Dynamic QoS Specification), which distinguishes between the *authorized, admitted* and *committed* resources (each upper bounded by the previous one), (ii) Usage of SIP to provide QoS guaranteed path (draft-gibson-sip-qos-resv-00.txt), which attempts to minimize the call set-up delay by tightly coupling the transport resource allocation with the session set-up, and (iii) Interdomain IP Communications with QoS … (draft-sinreich-interdomain-sip-qos-osp-00.txt), which proposes end-to-end usage of RSVP (for signaling only across multiple inter-operator domains – not int-serv) to set-up resource reservation. The current proposal for H.323 is similar to (ii) referenced above, however, it is expected that similar solutions could be worked out for H.323 as well and hence, the wireline resource reservation / minimization issue is by-and-large protocol independent.

However, one of the primary concerns for the cellular wireless access is minimization of the amount of information exchanged over the air-link. This calls for minimizing the number of messages (as well as the message sizes) exchanged for call set-up/resource reservation over the wireless link. We have proposed ways for minimizing the over-the-air messaging for both SIP and H.323 for a typical mobile-to-PSTN call as shown in Figures s1 and s2. Also, the ASN.1 PER encoding followed by H.323 will generally allow smaller message sizes than the basic textual SIP messages. Since the existence of suitable compression / encoding techniques for SIP messages is unknown, we have used simple text compression techniques to compress the SIP messages. The comparison is shown in Table st2.

2.4.3 If compression is applied to each signaling protocol (H.323 and SIP), what gains could be achieved percentage wise.

As H.323 signaling protocols employ ASN.1 PER aligned encoding rules, subsequent compression on these messages would be minimal.

We have chosen to apply both simple tag tokenizing techniques which we recommend to the SIP messages in order to achieve an optimal compression with no extra CPU overhead. In addition we have also applied a common text compression technique. We do not recommend the use of this method. The results are characterized below:

Method	SIP	H.323
Tokenization	13 – 19 %	Not Applicable
LZ77 after Tokenization	18 – 22 %	1-3 %
TOTAL	31 – 41 %	1-3 %

We expect that work to progress on tokenized compression methods for SIP and SDP will result in no need for applying a separate LZ77 compression mechanism.

2.5 Services

2.5.1 Services supported

H.323: Call Hold, Call Transfer (Blind, Alternative and Operator Assisted), Call Forwarding, Call Waiting, Conferencing (Multicast, Multi-unicast, Bridged, Consultative), Call Park, Call Pickup, Call Completion on Busy Subscriber, Calling Line ID, Message Waiting Indication**.** The services supported are standardized in the H.450 series of specifications, however, the actual support of these services is implementation specific and they are not currently widely deployed. There are some doubts in the industry if the H.450-series services will ever become widely deployed.

SIP: Call Hold, Call Transfer (Blind, Alternative and Operator Assisted), Call Forwarding, Call Waiting, Conferencing (Multicast, Multi-unicast, Bridged), Call Park, Directed Call Pickup, Calling Line ID, Call Return, Follow-me, Find me, Camp On, Call Queuing, Automatic Call Distribution, Do Not Disturb, Third Party Call Control. Note that the actual support of the services is implementation specific. The current SIP RFCs do not rigorously define the services. These are usually left to white-papers and perhaps informational RFCs.

2.5.2 Delay times to acquire services, both basic (such as dial tone or post-dial delay) and supplementary services

H.323: Example: With Fast Call Setup, there is a delay of 3-4 roundtrips (using TCP). Using UDP, call setup delay can be 1.5-2.5 round trips, depending upon whether or not a gatekeeper is involved. (Note, however, that with the fast call setup, capabilities are not exchanged, but can be exchanged later using H.245). Simultaneously sets up TCP connection as well to provide support in the case that the UDP setup fails.
SIP: Call setup delay is equivalent to H.323 Fast call setup. However, the establishment of the TCP connection (as a backup for the UDP setup failure) is sequential.

2.5.3 Billing and accounting

It is expected that distributed billing models can be applied to both protocols equally well.

H.323: Billing and accounting are not explicitly defined by the protocol, however, mechanisms existed and can be defined depending upon the requirements of the service provider. Gatekeepers can maintain logs and generate CDRs. A Gatekeeper can also instruct gateways to send copies of specific messages, for billing and accounting purposes. Version 4, which is still in draft form, may be adding procedures to provide the billing information (call duration, call termination cause, etc.) from the gateways to its gatekeeper to aid in the generation of CDRs. Annex G specifies that administrative domains may request other domains to provide them information about the usage of resources in specific calls. UsageIndication messages may be provided at any stage of the call. The ETSI TIPHON group has defined OSP for this purpose.

SIP: The functionality for billing for SIP depends on whether the service provider plans to charge for SIP services, for gateway services to the PSTN, or for carrying media data. For SIP services, the Authorization header can be used to indicate a customer identity that associates a SIP request with a billable entity. SIP server operations can be charged based on server logs or, for real-time billing, via AAA. For gateway services, the gateway can generate call detail records (CDRs). When QoS mechanisms are involved in a call, it would seem likely that these mechanisms would be responsible for the charging mechanism. Actual accounting records may be generated by AAA protocols or log files. Also, note that the DCS group drafts propose a billing extension to SIP for messaging between proxies. There is also a current proposal in the SIP working group to use OSP for Accounting purposes.

2.5.4 Comparison of services with existing wireless or wireline services (do the same set of services exist, not necessarily implemented the same).

H.323: Service set equivalent to existing wireless and a subset of existing wireline (specifically centrex). Additional location based and internet specific services are being proposed for version 4. Version 4 also plans to transport more of the wireline PBX features.

SIP: Service set equivalent to existing wireless and a subset of existing wireline (specifically centrex). Additional location based and internet specific services are supported by design intent.

2.5.5 Capabilities exchange services provided in the protocol

H.323: Uses H.245 protocol for capabilities exchange. The complete set of what a terminal can receive and decode is made known to the other terminal via its capability set. Precise information about each terminal's capabilities can be expressed in the CapabilityDescriptor structure. In the case of fastStart, an endpoint presents OpenLogicalChannel structures all of the capabilities it can support in both directions. The receiving end will choose from that list and return the OpenLogicalChannel structures it chose to use in a response message.

SIP: Uses SDP for media capabilities exchange. SIP itself also relies on the require and proxy-require headers with the Not Supported and Options methods for session signaling capability exchange. Typically, each endpoint tells the other what capabilities it can receive. The confirmation of which one is chosen is implicit in where the media streams will be sent. Callers can use an OPTION request to find out the capabilities of the callee. Another option for capabilities exchange is to use an INVITE message, which is replied by a 480 Unsupported message, which results in a second INVITE. One drawback is that SDP does not currently support asymmetric and simultaneous capabilities of audio and video encoding. The MMUSIC working group of the IETF has a work item to solve the asymmetic negotiation issue. They will likely enhance SDP "m=" tag's value formats. The value of providing this ability is somewhat in question, however.

2.5.6 Personal mobility services (delivery of services wherever the subscriber is located, network independent) and location based services

H.323: H.323 can redirect a caller to other addresses. Gatekeepers offer an inherent way for terminals to register/unregister at different locations. User preferences can be specified in user-user signaling. There are plans in version 4 to define these services (a substantial amount of work on mobility for H.323 has been started in ITU-T SG16). In the ETSI TIPHON WG 7, proposals have been made for the support of IP mobility, but nothing has yet been standardized.

SIP: SIP inherently supports the concept of personal mobility and location based services, when a call is being setup, using the SIP redirect server, server forking, registrar server concept and allowing users to proxy requests. Additional proposals have been made, but not accepted, for the support of IP mobility (characterized by frequent roaming and changing of location during a call) using the SIP redirect server. SIP was designed to use any protocol to access a location server – including SIP to a registrar server. We support the use of directories enabled with authentication and authorization mechanisms for home domain location servers.

2.5.7 Interworking and interoperability with legacy networks (wireline and wireless)

H.323: There are specific standards (H.246) specifying interoperability with legacy circuit switched networks. However, the H.246 recommendation and it's various annexes are not sufficiently complete in scope to be very valuable and should be viewed as a "guide" rather than a specification.

SIP: The standards approved to date do not specify interworking or interoperability. However, internet drafts have been written to define the interoperability.

2.5.8 Interworking and interoperability with other IP call control protocols (e.g., cable)

H.323: No explicit interworking and interoperability is defined with other IP call control protocols.

SIP: The DCS group drafts in the IETF SIP WG specify the interactions required in an IP cable environment.

ITU SG-16 has taken on the task of interoperability between SIP with H.323.

2.5.9 Security services provided, authentication of users and network elements, data privacy and encipherment

H.323: Authentication and security for H.323 is optional; however, if it is provided it must be in accordance with Recommendation H.235. RTP, which supports encryption, can be used to carry media. Between administrative domains, when authentication, data integrity and encryption is desired for messages, the IETF IPsec procedures are applicable (specifically, RFC 1825, 1826 and 1827). The ETSI TIPHON specifications define countermeasures to ensure a secure TIPHON compliant system. Security requirements are based on customer, service and network provider objectives for confidentiality, integrity, accountability, availability and non-repudiation. Lawful interception (a requirement in some countries) is also to be supported.

SIP: SIP can encrypt and authenticate signaling messages. RTP which carries the media supports encryption. IPsec procedures are also applicable between inter-domain network elements. The DSC group has made a proposal for support of lawful interception with SIP; however, there was great resistance in the IETF plenary to take on this type of work in the IETF. Perhaps this specification should be a work item for the ITU.

2.6 Wireless standards consideration

2.6.1 Where and how would any wireless specific changes to the protocol be implemented?

Ideally, there would be no wireless specific changes to these protocols. For location and personal mobility based services, a common wireline/wireless unified directory should provide user and terminal location as necessary for these protocols.

Tokenized compression of SIP and SDP could be standardized by the IETF, 3GPP or the WAP forum and issued to the IETF as an information RFC. It may be that the existing message sizes are acceptable and as such no wireless specific optimizations would be needed.

2.7 OA&M

2.7.1 What OA&M procedures are available?

The INOW initiative has developed MIBs for many H.323 components such as the Gatekeeper and client.

There are several initiatives in the IETF Pint WG and PacketCable initiative have defined MIBs for the SIP proxy servers and clients.

2.7.2 What level of fault detection and monitoring are available?

The MIBs are used to convey fault and monitoring information to management entities.

H.323 has a loop-back command defined. This function is accomplished in SIP using a SIP robot and existing methods, SDP media tags and values.

3 H.323/SIP Comparison Questions

Assume that the UMTS Release 2000 network architecture is to be developed using either:

- the H.323 and associated standards (H.245 control; H.225.0 connection establishment; H.332 conferences; H.450.1, H.450.2, H.450.3 supplementary services; H.235 security, and H.246 interoperability with circuit-switched services) or
- the SIP protocol.

3.1 Describe and compare the purpose and intent of the H.323 and SIP protocols. Consider why they were developed, what networks they were developed for, what services they were intended to support/provide, etc.

3.1.1 H.323

H.323 was initially based and patterned off of the H.320 family of standards. The intent was to define a multimedia conferencing system for LAN environments which did not provide any guaranteed quality of service. It's base control and bearer establishment signaling were purposefully patterned off the N-ISDN and B-ISDN circuit-based multi-media networks. Due to this adherence to circuit based multimedia standards, H.323 has recently evolved to work over and interwork with ATM transports as well.

Interworking H.323 with the H.320 family of ITU standards in slow start is somewhat straight forward. With fast-start, many new interoperability issues come up. We should also mention that many of the fields that were added to H.323 signaling base protocols such as: H.225 and H.245 were text based fields. By defining these fields this way the interoperability with the traditional all binary protocols of H.320 and the PSTN is diminished.

Another intent of the H.323 design was to allow multimedia applications at bandwidths on the order of 28Kbps. At the time Internet access was pretty much limited to this speed using dial-up networking. This is one of the reasons why ASN.1 PER aligned encoding rules were used on the signaling.

The initial goals of H.323 were to provide multimedia conferencing applications such as voice, video and T.120 data applications such as whiteboarding. The typical client interface was a separate standalone application with user interfaces specifically designed for these applications in mind. As H.323 was not designed for managed LAN environments such as integrated services or differentiated services, it included separate resource management mechanisms in the H.225 signaling standard. In version 2 modular RSVP interactions were specified as a step of the H.245 bearer establishment procedures.

Mobility (both terminal and personal) were not a consideration until 4Q1999 as a candidate addition to H.323v4.

Security was put in later after the intitial versions. It was added on as a new protocol, H.235.

Support for services were also an add on (H.450). H.450 services are derived from QSIG supplementary services, but they are sufficiently different to be difficult to interwork.

H.323 has also evolved itself over the past year into a more scalable wide area networking standard that can be interworked with managed large scale networks.

Although many claim that gatekeeper implementations can be call stateless, few if any implementations exists that are. It would certainly require much work (and thinking) to implement a call stateless version of a gatekeeper.

3.1.2 SIP

The earliest records we have on SIP and it's ancestors are from the early 1990s professional multimedia conferencing organization mail archives.

Actually there were predecessors of SIP; namely, IVSD (Turletti, INRIA) and MMMC (Schooler, ISI). After much wrangling over the name, concepts from each protocol were merged into SIPv1 (Schooler and Handley). SIPv1 only handled the session setup transaction. There was no BYE transaction.

During this process of standardization, Henning Shulzrinne was also working on a protocol called SCIP which was designed to work with RTSP. SCIP had the capability for session termination. SCIP also used TCP for the transport.

Again after much wrangling over the name and transports, etc.. The concepts were again merged into SIPv2 which people now simply call SIP. SIP took the best of both of these worlds.

The main goal of the MMUSIC WG was to create a generic session signaling protocol that could have many uses including multimedia but also multi-player games, bank transactions, etc.. SIP was designed with modularity, security, personal mobility, reliability, extensibility, scalability, the World Wide Web, browsers, java applets, SDP, SAP, SMIL, RTSP, DNS and the PSTN in mind.

It was also designed with personal mobility (location management) from the beginning.

In addition, SIP was designed to interwork in a modular form with unmanaged IP networks as well as hybrid, integrated and differentiated service networks.

3.1.3 Conclusion:

Based on the above statements, the following distinguishing points can be identified:

• The H.323 protocol is developed to make audio, video and T.120 multimedia applications suitable to work in the packet based network environment	• The SIP protocol is developed for any application suitable to work in the packet based network environment and is written in a way that is much more compatible with other "web" protocols
• This protocol remains tightly coupled with the telecommunication standards	• This protocol remains loosely coupled with any specific application standards

3.2 Describe and compare the status of the H.323-series protocol standardization and the SIP protocol, in terms of availability and maturity of standards for product development.

While both camps will say that one is more "standardized" than the other. They are essentially in the same boat when comparing H.323v4 and SIP. H.323 proponents will claim that every single feature supported is explicitly spelled out in rigorous form. Therefore they will claim that it is more standardized and complete.

Actually this is the nature of the design of the H.323 family of standards. It has been designed with the mindset that new fields have to be added or protocols created to add new functionality; therefore it must be standardized. Examples of this are the addition of H.450, support for T.140 services and security.

The SIP proponents are often affronted when someone says that SIP needs much more standardization for services. The feeling is that it was designed to manage sessions for any application and feature.

In response, they have gone through the exercise of writing white-papers and informational drafts describing how most of the legacy PSTN features can be implemented using SIP. But again people keep insisting that these interactions must be standardized further.

Bearing in mind the mindset of the SIP inventors, this is like asking them to standardize every web portal that could ever be built using HTML, CGI and Java Applets. It truly points out the issue of "what is and isn't a standard feature". However, a significant effort is underway in the IETF to produce and informational RFC that spells out a standard set of traditional services for telephony.

Strictly speaking, H.323 is more rigorously defined (spelled out) on a per-feature basis, and more interoperability testing has been attempted thus far. It is; however, a painful process because of the complexity of H.323. This makes achieving interoperability very difficult and limited to very basic functionality. Many advanced services were tested for interoperability successfully at the last SIP bake-off.

In the end, signaling protocols are signaling protocols and they both need standards work. Neither explicitly defines all of the services needed and neither defines how QOS should interwork with the PSTN in the most optimal way.

We do believe that SIP can be extended and developed more easily from a TTM perspective. We also actually believe that more features are referenced and defined in white-papers and informational RFCs for SIP than for H.323. The white papers describe the features interactions in a single paragraph as opposed to creating new fields and doing flows.

The standards of both protocols are sufficiently developed for product development. Many of the work items for both protocols are common. An example of this is QOS interworking. H.323 does have more standards documents and greater detail of explanation.

3.3 What standards changes are envisioned to H.323, and associated standards, to support the UMTS Release 2000 architecture? What standards changes are envisioned to SIP, and associated standards, to support the UMTS Release 2000 architecture? What is the estimated development time for and availability each of these standards? Compare the standards process for H.323 and SIP.

For H.323, we would expect to see:

- Support of fast-connect is mandatory in all version 4 implementations.
- Interactions with personal number mobility defined.
- A connect acking mechanism defined for fast-start when UDP is used as the transport.
- ITU standard on H.323 and lawful interception
- Annex H - User and Service Mobility (not terminal mobility) in H.323.
- Annex I – Packet based telephony over Error Prone channels

Progress on these last annexes has so far been very difficult because of very different points of view from the participants (most specifically over session layer specific terminal mobility). A full standard will take quite a bit of time to complete. Expect an October- December 2000 timeframe.

For SIP, we would expect to see:

- An more detailed informational RFC detailing a best common practice for standard features (A current SIP WG work item).

- A best commons practices informational RFC for dealing with the PSTN (A current SIP WG work item)..
- ITU definition of lawful interception.
- An informational RFC defining conferencing procedures.
- An informational RFC on interactions with QOS signaling and the PSTN.

Both:

- Standard specifying SIP/H.323 interworking. (IETF and ITU should work together on this).

Required functionality and development duration go hand in hand. Product delivery dates are best conveyed between specific operators and vendors. We will say that the use of SIP will likely result in less development time than H.323 for the same functionality. Wireless inappropriate H.323 clients have been bundled with a certain operating system, however.

H.323 is being defined in the ITU, an extremely professional and seasoned organization with many years of telecommunication standardization under their belt. The group has been quite pro-active in bringing H.323 up to it's current level and has greatly diminished the "thinking" that it takes forever for ITU to do anything. They bring a level of professionalism to standards and understand exactly what needs to be written down in order for companies and individuals from all over the world to implement it. They also bring the maturity of the business and legislative influences to the problem domain and are definitely more suited to handle issues such as lawful surveillance. SG – 16 is relatively democratic, open minded and allows for participation from many companies. The work of the group is very open to input.

The SIP environment is quite loose. Most of the work and lobbying on contributions happens behind the scenes. It worked quite well when the IETF was a smaller set of individual researchers. As the IETF has grown, the interests of large companies has become quite pervasive. This has resulted in more disputes over solutions to issues and off-session (big company meetings and consortia) to apply ITU-like principles to IETF protocols. While this has been useful to work out many details, often times the solutions to problems, go against the simplicity, scalability and extensibility principles that the original authors had in mind. Most of the IETF area directors are extremely professional and have often served well as arbitrators for these disagreements.

We believe that work on SIP will be shared more between the ITU the IETF in the future. 3GPP may be the best candidate to work on SIP for wireless environments. The tokenized compression could be done either by 3GPP or perhaps the WAP forum.

Development Time: Perhaps 3GPP steering groups should coordinate these activities among members to facilitate these standards going forward in the required timeframe. If stateless call processing is a desired function, then perhaps the IETF would be the best place for most of this work. Many people in the ITU and ETSI are not in favor of such thinking. Given a correct set of functional requirements being defined and proper management, any standards body is suitable for the work. The IETF may not meet with the frequencies necessary; however, adhoc activities that meet with more frequency can be employed by working groups to achieve the desired timeframes. There is no reason why the adhoc activities can not occur as a part of an industry consortium, 3GPP, ETSI, or the ITU for that matter.

3.4 Describe and compare the overall complexity of implementing the H.323 and associated multimedia control protocols, and the SIP protocol.

Starting from scratch, H.323 is by far more complex due to it's ASN.1 PER encoding and multiplicity of protocols. H.225, H.245, H.450, H.235, etc.. In additional, the parameter extensibility mechanism are archaic and complex. H.323 has a ton of baggage which becomes apparent in its parameter definitions. The usefulness of this baggage has little value and only serves to confound the design staff and blow up the protocols. This baggage should be removed in H.323v4 and backward compatibility should not be required if H.323v4 is chosen.

Working SIP implementations can be created in relatively short development intervals which has been proven many times by many individuals within many companies and educational institutions. SIP is far less complex to implement.

3.5 Describe and compare the complexity of implementing H.323 and associated protocols, and the SIP protocol, in the UMTS Release 2000 architecture.

H.323 is more complex to implement than SIP and SDP. Please refer to complexity section above.

3.6 How widespread do you envision H.323 deployment into other access technologies, such as Internet Telephony and Cable? How widespread do you envision SIP deployment into other access technologies, such as Internet Telephony and Cable?

We see that SIP will become the leader in the market due to it's simplicity, stack size, modular design and interoperability with the WWW and browser environments and most importantly it's TTM differentiator. As a bonus it also does telephony and interworks and can facilitate control of services for the PSTN as well.

H.323v2 has a significant installed base due to a certain OS manufacturer bundling a well known H.323 client with the OS. This is not H.323v4 and does not support fast connect.

It should be noted that almost all OS manufacturers and browser vendors are taking significant interest in SIP while the focus seems to be diminishing on H.323. The industry is split right now. Each company has people in both camps. Each company is working on the interworking of the two protocols. We see SIP as the long term winner due to TTM, complexity considerations, web synergy and flexibility which translates into product cost. We believe it has the best hope of giving service providers more control of their services with less dependence on their vendors' schedules.

3.7 Describe implementation benefits or challenges faced with H.323's binary representation of messages, based on ASN.1 and packed encoding rules. Include considerations for interoperability, reuse, etc. Describe implementation benefits or challenges faced with SIP's text encoding of messages, similar to HTTP and RTSP. Include considerations for interoperability, reuse, etc. Compare the benefits and challenges of the encoding of the H.323 protocol and the SIP protocol.

Please refer to section **Erro! Argumento de opção desconhecido.**, **Erro! Argumento de opção desconhecido.** for an elaborate discussion of these matters.

3.8 Describe and compare H.323's and SIP's support for distributed call signaling and resource management.

Both could be made to work with a distributed call model. It would be easier to use SIP and SDP.

3.9 Discuss and compare the interoperability of an H.323-based UMTS Release 2000 network and a SIP-based UMTS Release 2000 network with the existing networks (wireless and wireline). Identify and compare complexities with the interworking that might be present using H.323 and SIP (i.e., gateway complexity).

H.323: There is nothing inherent in H.323 that facilitates interworking with the PSTN. Despite some superficial similarities in H.225.0 messaging and Q.931, there is not much in common with the two approaches. Practically speaking, H.323 and the PSTN are dealt with separately and the protocols are completely terminated on both sides of a gateway. There is very little in common between STANDARD H.323 protocol stacks and ISDN protocol stacks. There is no one-to-one mapping between ISUP or DSS1 and H.323. Legacy PSTN signaling is typically binary formatted but typically does not employ ASN.1 PER. Complexity is exacerbated by the existence of both fast-start and slow-start methods.

SIP: There is no one-to-one mapping between ISUP or DSS1 and SIP. Legacy PSTN signaling is typically binary. SIP is ASCII formatted. MGCP is text based and one H.248 encoding is also text based.

SIP and H.323 fast-connect are basically just as difficult to inter-work with other networks.

Elements would have to either support both protocols or a gateway device would have to be used as an intermediary between the networks. Most deployable PSTN gateways use text-based MGCP which employs SDP. It would be easier to map SIP to these MGCP-derived protocols. The Megaco protocol has 3 encoding variants. The mass deployment and support of each encoding variant would be a dubious undertaking. One encoding variant of H.248 is very similar to MGCP.

3.10 Describe and compare the basic set of services that would be supplied in a UMTS Release 2000 network with H.323 and SIP implementation. How would additional services be added for: carrier defined, third party defined, and standards defined services? How are trigger points defined? Where are the services supported, visited or home network?

H.323: The following services can be supplied in UMTS Release 2000 by using H.323:

> Call Hold, Call Transfer (Blind, Alternative and Operator Assisted), Call Forwarding, Call Waiting, Conferencing (Multicast, Multi-unicast, Bridged, Consultative), Call Park, Call Pickup, Call Completion on Busy Subscriber, Calling Line ID, Message Waiting Indication**.** The services supported are standardized in the H.450 series of specifications, however, the actual support of these services is implementation specific and they are not currently widely deployed. Implementing new H.450 features in H.323 requires new standardization which is extremely slow, or proprietary new features. In both cases it requires new "software loads" in the client and server side. H.323v4 may add new stimulus signaling invocation mechanisms (H.323 Annex L) to invoke features. That would be in "competition" with the native H.450 features. There is also a push for tunneling "legacy" signaling between gateways or gatekeepers in H.323 to allow for interworking with PSTN. Basically, the whole supplementary aspects of H.323 is being re-thought. The H.323 protocol suite can be standardized to support IN triggers, carrier defined and 3rd party defined and controlled services.

SIP: The following service can be supplied in UMTS Release 2000 by using SIP:

Call Hold, Call Transfer (Blind, Alternative and Operator Assisted), Call Forwarding, Call Waiting, Conferencing (Multicast, Multi-unicast, Bridged), Call Park, Directed Call Pickup, Calling Line ID, Call Return, Follow-me, Find me, Camp On, Call

Queuing, Automatic Call Distribution, Do Not Disturb, Third Party Call Control. Note that the actual support of the services is implementation specific. The current SIP RFCs do not rigorously define these services. SIP can support IN triggers, carrier defined and 3rd party defined and controlled services without further standardization.

3.11 Describe and compare the OAM&P services that would be supported in a UMTS Release 2000 with H.323 and SIP implementation.

Management of the SIP proxies and H.323 gatekeepers is a necessity. This is largely implementation specific and should probably not specifically depend on the session protocol. The INOW initiative has been effective in extending some OA&M funtions into the protocol itself. Both the IETF IPTEL WG and the packetcable consortium have developed SIP based MIBs.

3.12 Discuss and compare how accounting and billing would be performed in the UMTS Release 2000 network with H.323 and SIP.

This issue is protocol agnostic. The exact details (protocols and formats) need to be worked out first by a set of functional requirements stemming from the operators of the UMTS Release 2000 environment. Synergies with other billing and accounting systems chosen by other access technologies should be a consideration.

The INOW initiative has embedded accounting metrics into the H.323 signaling. SIP leaves this function to other protocols but provides enough information itself for distributed inter-operator billing.

The proxies and gatekeepers must be involved in CDR generation. The inter-operator clearinghouse concept needs to be decided as well as agreement on the distributed per-transaction vs per-call billing. We expect that CDR records will be generated and transferred to billing servers in an extremely extensible message format to provide for the flexibility needed to allow for the rapid introduction of new services.

3.13 Assuming firm standards text exists for an H.323 implementation in the UMTS Release 2000 architecture by 4Q00, is it possible to have General Availability of an UMTS Release 2000 network with H.323 multimedia call control solution deployed by the end of 2Q 2001 with this architecture?

Discussion of specific dates for deployment are best handled between specific operators and vendors.

3.14 Assuming firm standards text exists for an SIP implementation in the UMTS Release 2000 architecture by 4Q00, is it possible to have General Availability of an UMTS Release 2000 network with SIP multimedia call control solution deployed by the end of 2Q 2001 with this architecture?

Discussion of specific dates for deployment are best handled between specific operators and vendors.

3.15 Please identify which, if any, of the UMTS Release 2000 defined interfaces you believe will be proprietary available in 2Q2001.

It is hoped that none will be necessary but many of the interfaces and interactions have yet to be defined. At this point we consider it to be in our best interest to discuss possible candidates for proprietary interfaces between specific vendors and operators.

3.16 Describe any alternate means of staging deployment of function to facilitate the introduction for this architecture.

Staging deployment of function is contingent on all interfaces and interactions of network elements being defined.

3.17 Describe and compare how a subscriber obtains a consistent and full set of services, no matter what network they may be located in, using the UMTS Release 2000 architecture with H.323 and SIP signaling. Specifically address which network elements in both the visited and home networks would be involved and require signaling interaction, what element controls bearer resources, etc.

A "gatekeeper using gatekeeper-routed signaling" in H.323 parlance is roughly equivalent to a "proxy SIP server" in SIP parlance. This model can be used in both cases to provide services to end-points. These models allow the network to keep a tight control on supplementary services being offered to the user.

A "gatekeeper using direct-routed signaling" is roughly equivalent to a "SIP redirection server" in SIP parlance. This model is much less intensive on the gatekeeper or SIP server, but it means that services can be provided end-to-end without network participation.

There are ways in both SIP and H.323 to redirect signaling to a "proxy server" to provide specific or consistent services.

There is no reason why a network should be constrained to a particular model. Service providers should have the liberty to decide which is applicable for any specific service.

Both SIP and H.323 allow for much flexibility, provided that they are implemented carefully.

These issues are protocol agnostic.

3.18 For the purpose of highlighting and comparing the signaling complexities, provide example call flows using both H.323 and SIP signaling and the UMTS Release 2000 architecture to support one or more of the following services:

3.18.1 Mobile originated call to a landline phone

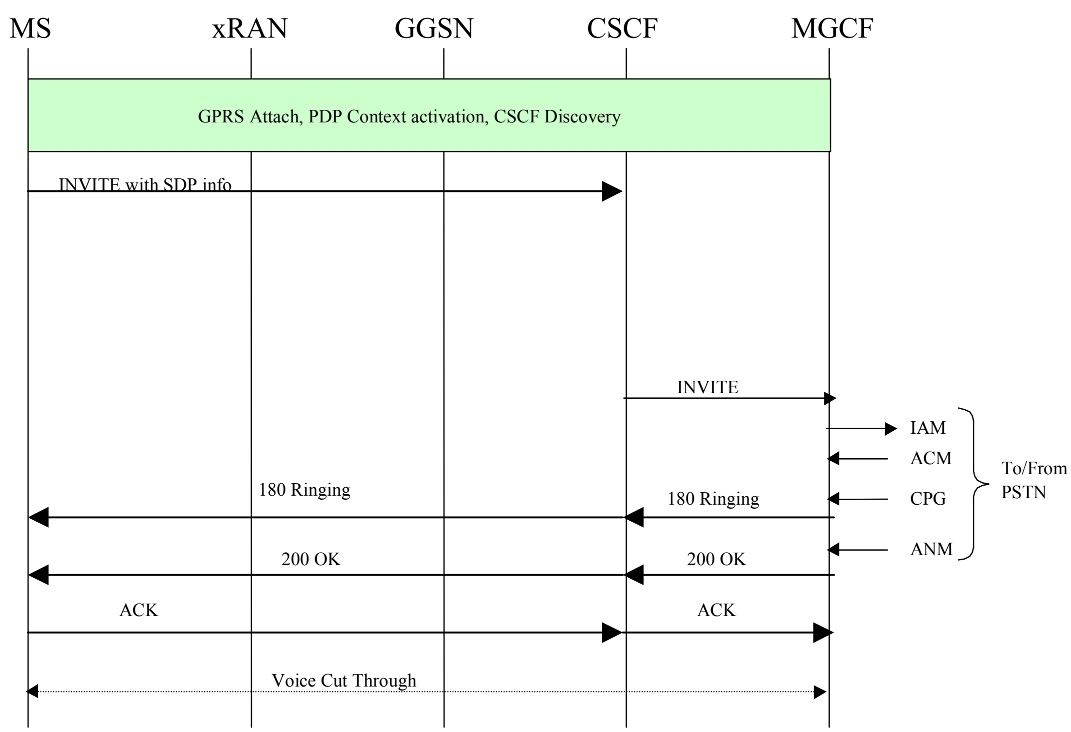


Figure 19a1. Using SIP

For details of the call flow, see the text beneath Figure 18a. The radio access resource reservation part, being protocol independent, has been omitted from Figure 19a1.

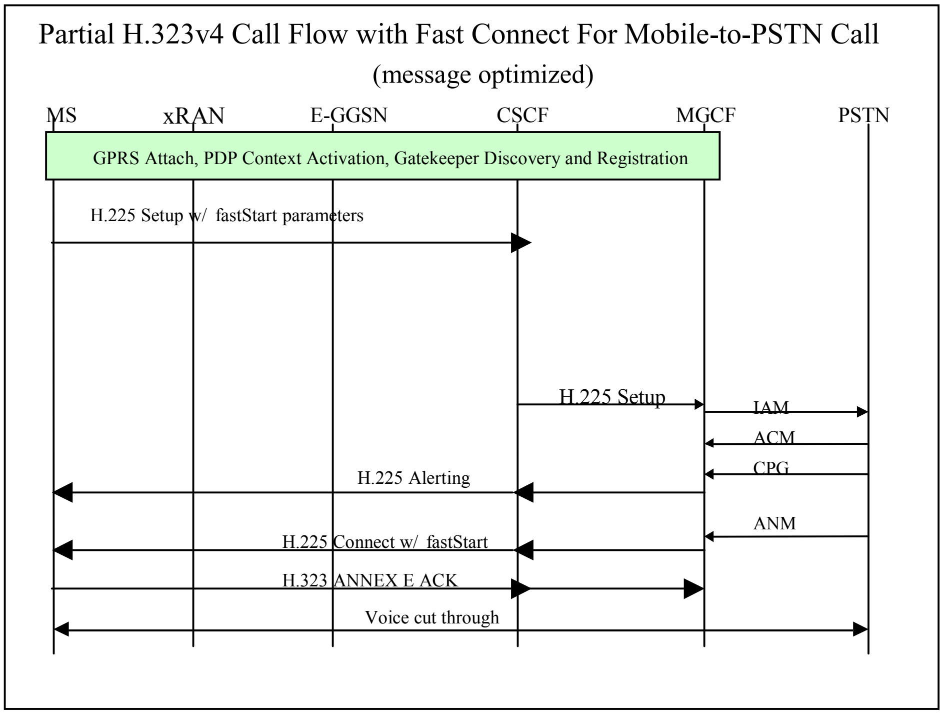


Figure 19a2. Using H.323V4

For details of the call flow, see the text beneath Figure 18b. The radio access resource reservation part, being protocol independent, has been omitted from Figure 19a2.

3.18.2 Mobile originated call to a mobile phone

See case d)

3.18.3 Mobile terminated call from a landline phone

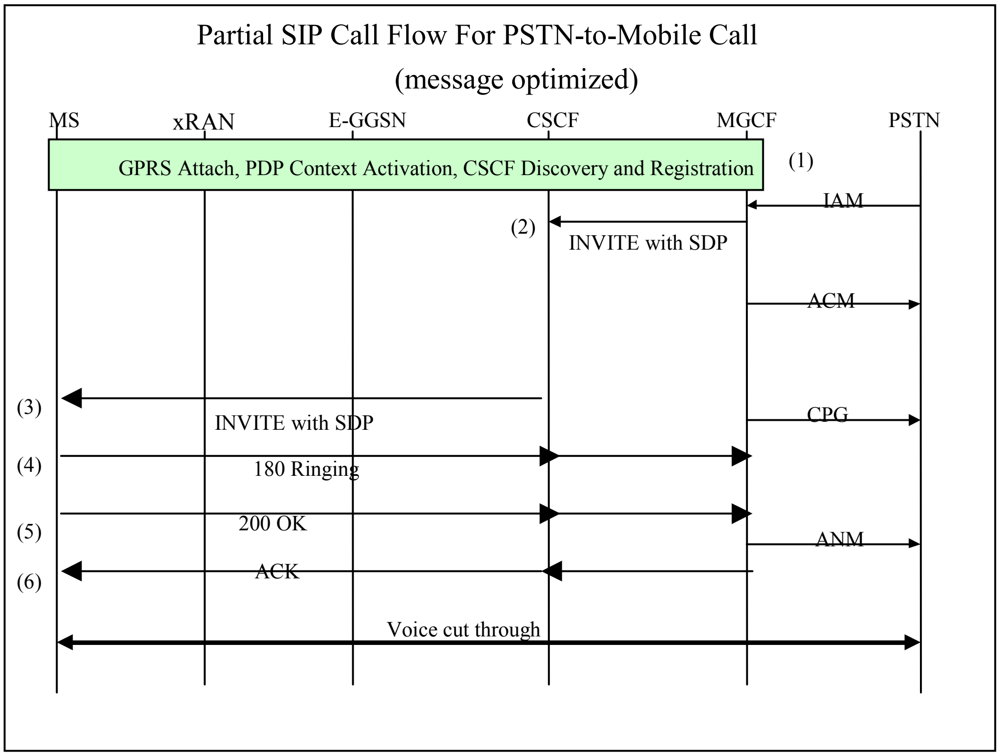


Figure 19c1. Using SIP

CALL FLOW DESCRIPTION :

(1) The MS performs a GPRS attach, primary PDP context activation and CSCF discovery.
(2) MGCF receives an IAM message from the PSTN. It sends a SIP INVITE message (with SDP element) to the CSCF. It also sends the ACM and CPG messages back to the PSTN.
(3) The CSCF forwards the SIP INVITE message to the MS.
(4) On receiving the INVITE message, the MS generates the ringing and sends a 180 Ringing message to the MGCF indicating that the called user's phone is ringing
(5) When the called party answers the phone, the MS generates a 200 OK message and sends it to the CSCF, which proxies it to the MGCF (Note: the 200 OK message can trigger billing here). The MGCF generates an ANM message and sends it to the PSTN.
(6) The MGCF sends an ACK message to the MS. After this, the voice is cut-through.

NOTE: The radio access resource reservation part, being protocol independent, has been omitted from the flow.

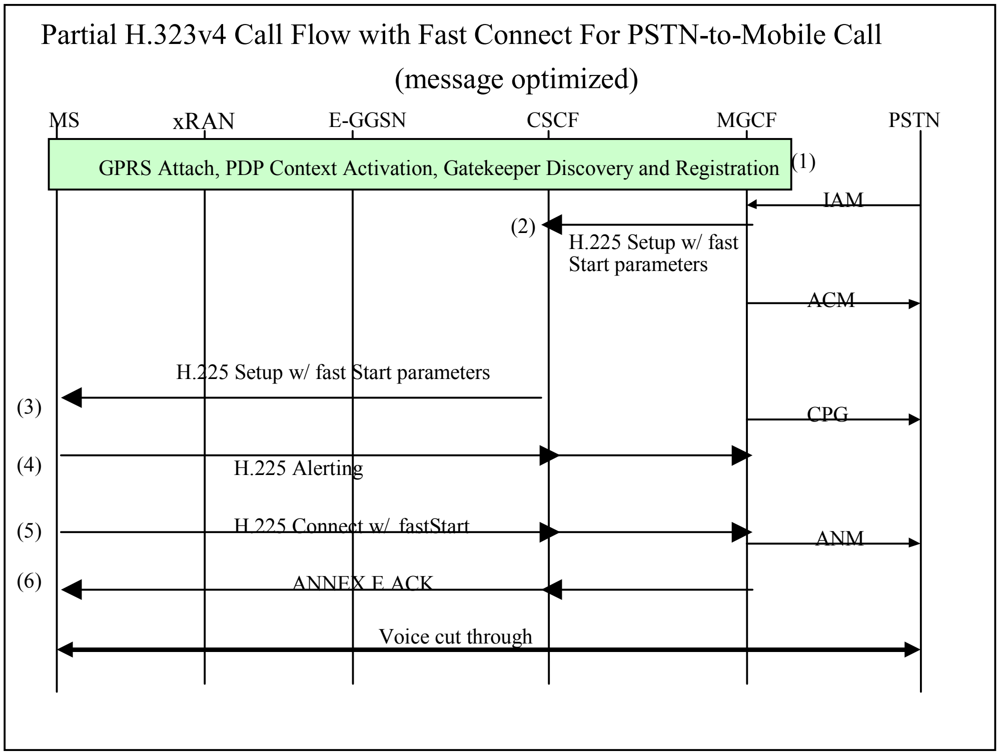


Figure 19c2. Using H.323

CALL FLOW DESCRIPTION :

(1) The MS performs a GPRS attach, primary PDP context activation and CSCF discovery.
(2) MGCF receives an IAM message from the PSTN. It sends a H.225 SETUP message (with FastStart element) to the CSCF. It also sends the ACM and CPG messages back to the PSTN.
(3) The CSCF forwards the H.225 SETUP message (with FastStart parameters) to the MS.
(4) On receiving SETUP message, the MS generates the ringing (or may also receive in-band ringing tone from PSTN at this point) and sends a H.225 Alerting message to the MGCF indicating that the called user's phone is ringing.
(5) When the called party answers the phone, the MS generates a H.225 CONNECT message with FastStart parameters and ANNEX E header with the AckRequested bit set, and sends it to the CSCF, which proxies it to the MGCF. The MGCF generates an ANM message and sends it to the PSTN.
(6) The MGCF responds with an ANNEX E ACK message. After this, the voice is cut-through.

NOTE: The radio access resource reservation part, being protocol independent, has been omitted from the flow.

3.18.4 Mobile terminated call from a mobile phone

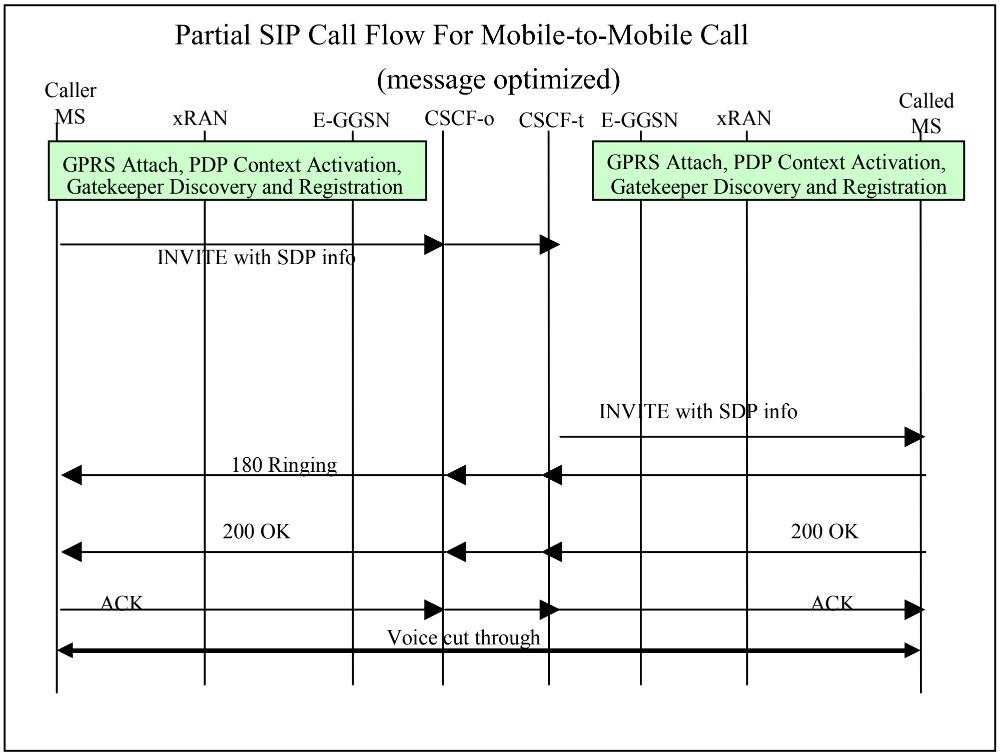


Figure 19d1. Using SIP

For details of the call flow, see the text beneath Figure 18c. The radio access resource reservation part, being protocol independent, has been omitted from Figure 19d1.

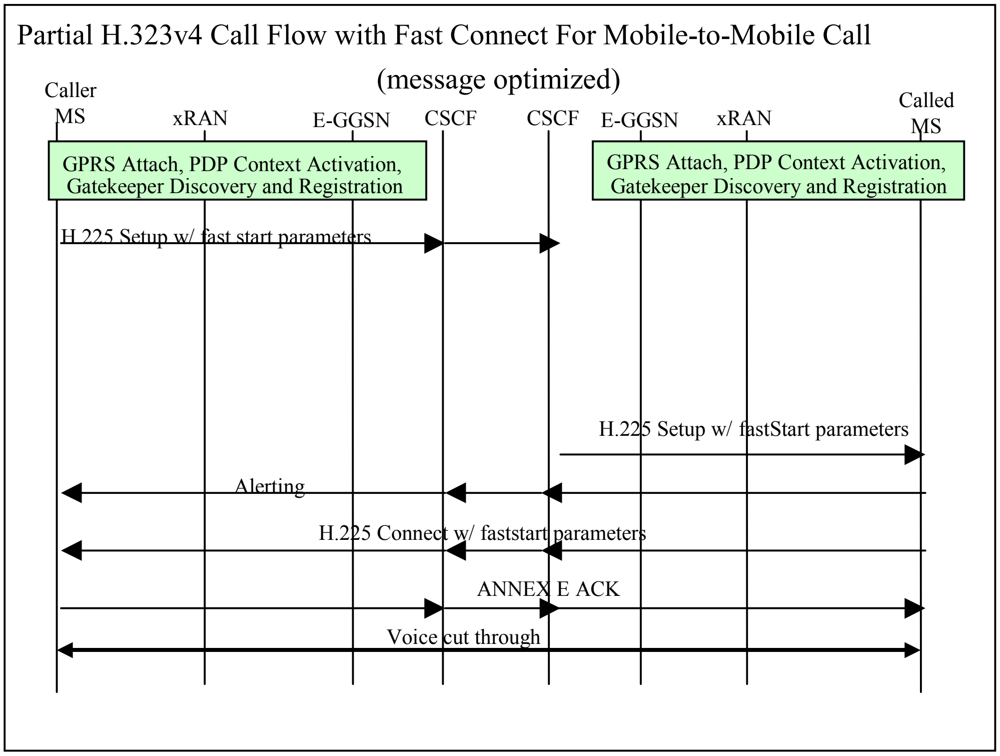
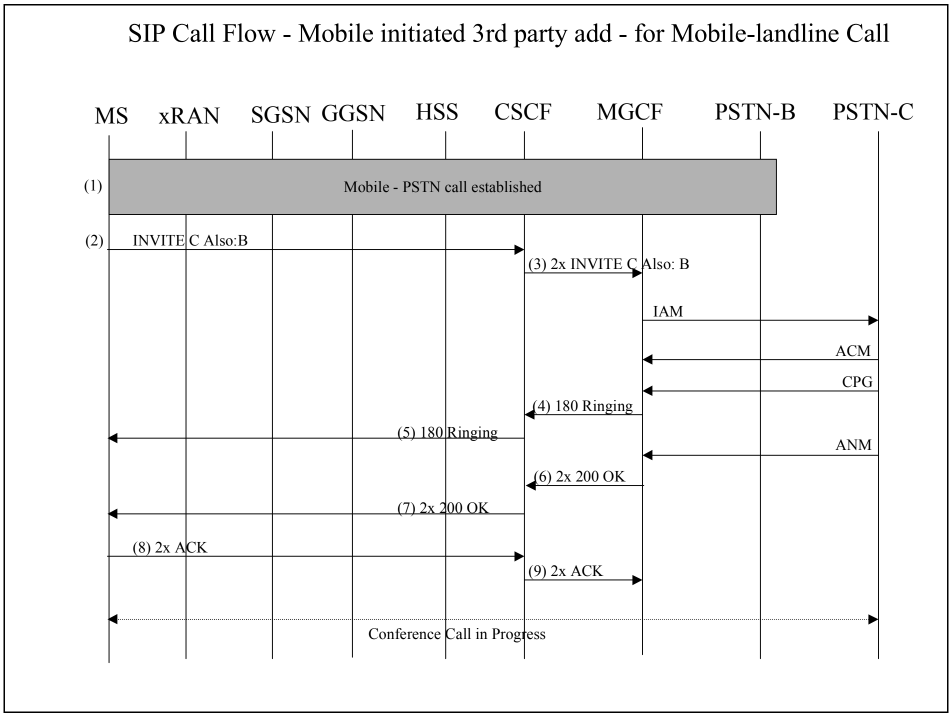


Figure 19d2. Using H.323

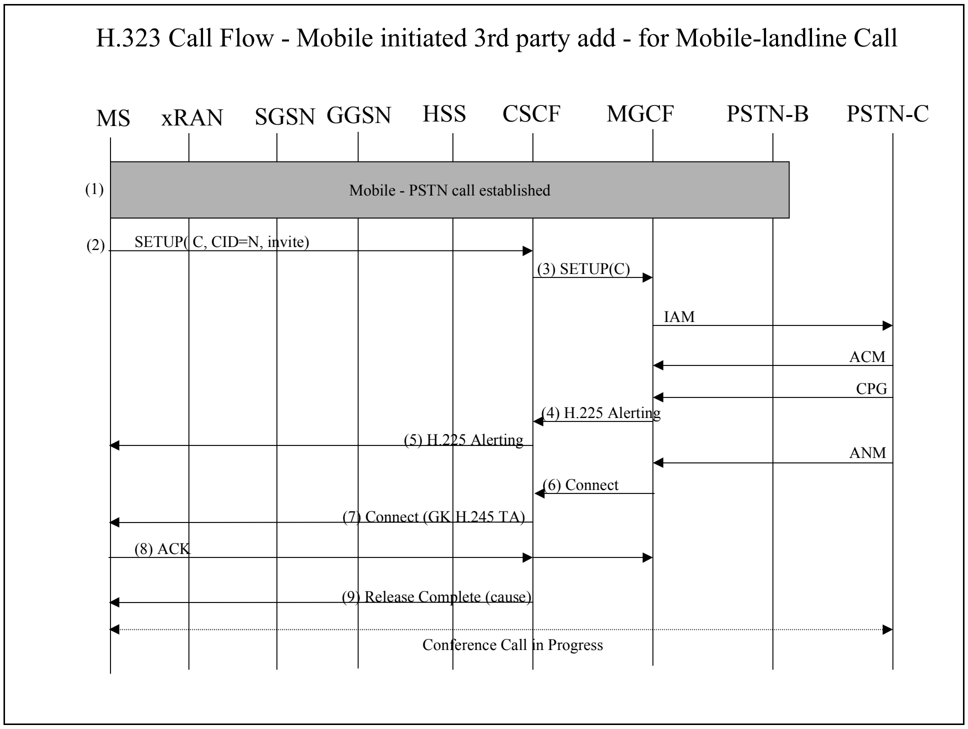
For details of the call flow, see the text beneath Figure 18d. The radio access resource reservation part, being protocol independent, has been omitted from Figure 19d2.

3.18.5 Mobile party adding in a third party (landline) in an existing mobile to landline connected call

The use of multicast is assumed, for now, as the interface to the MRF is not currently defined. The mobile is essentially acting as the MC in this case. The CSCF accounts for the billing of the functionality. SIP employs the INVITE transaction exchange to facilitate notification of media changes to PSTN-B. H.323 employs an adhoc conference by issuing a FACILITY message exchange with PSTN-B.

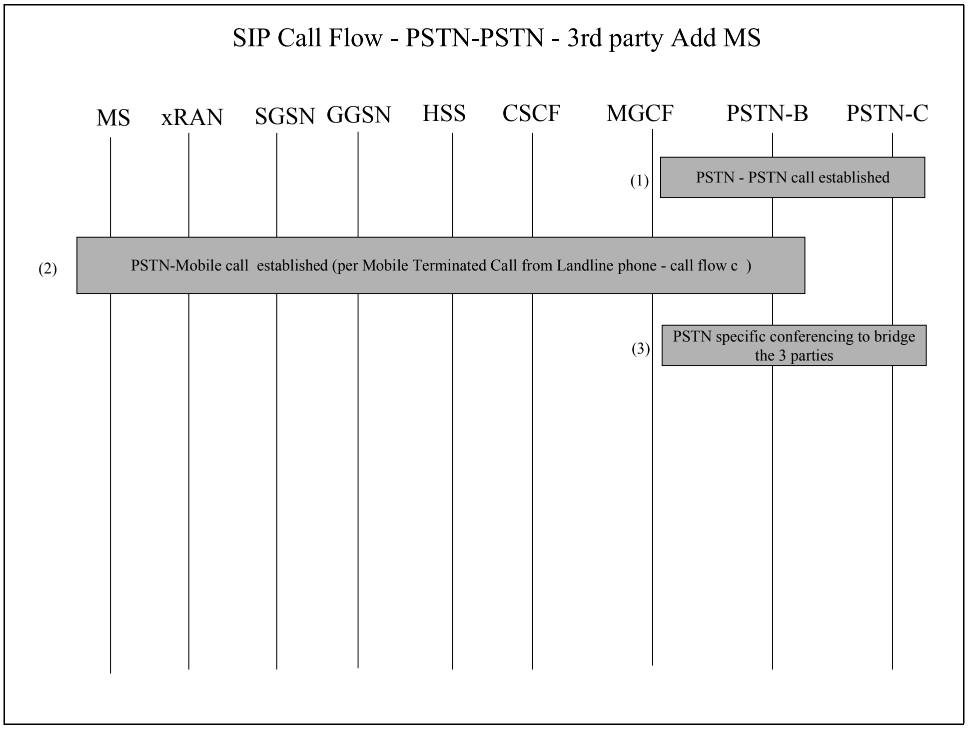


(1) A mobile originated call with party B in the PSTN has been established per call flow (a).
(2) The MS allocates a multicast address, joins the address and sends two SIP INVITE message to the CSCF. The CSCF does a "translation" on both and determines that interfacing through the MGCFs is necessary to establish a call with party C. One INVITE is proxied to the MGCF for part B and the other to the MGCF for C. The also indicates a conference. The multicast address is included in the SDP to transmit and receive from.
(3) Upon receipt of the INVITE message, The MGCF starts exchanging a series of ISUP messages with the PSTN to set-up the call to party C.
(4) On receiving the CPG message from the PSTN, the MGCF sends an 180 Ringing message to the CSCF.
(5) The CSCF forwards the 180 Ringing message to the MS, which triggers ringback in the MS.
(6) On receiving an ANM message from the PSTN, the MGCF sends a 200 OK message to the CSCF. The party B MGCF replies back directly after confirmation of it's join as well.
(7) Upon receipt of each 200 OK, the CSCF forwards each 200 OK to the MS.
(8) The MS responds with an ACK message for each 200 OK. The CSCF begins accounting for the service.
(9) The CSCF forwards the ACKs to the MGCFs.

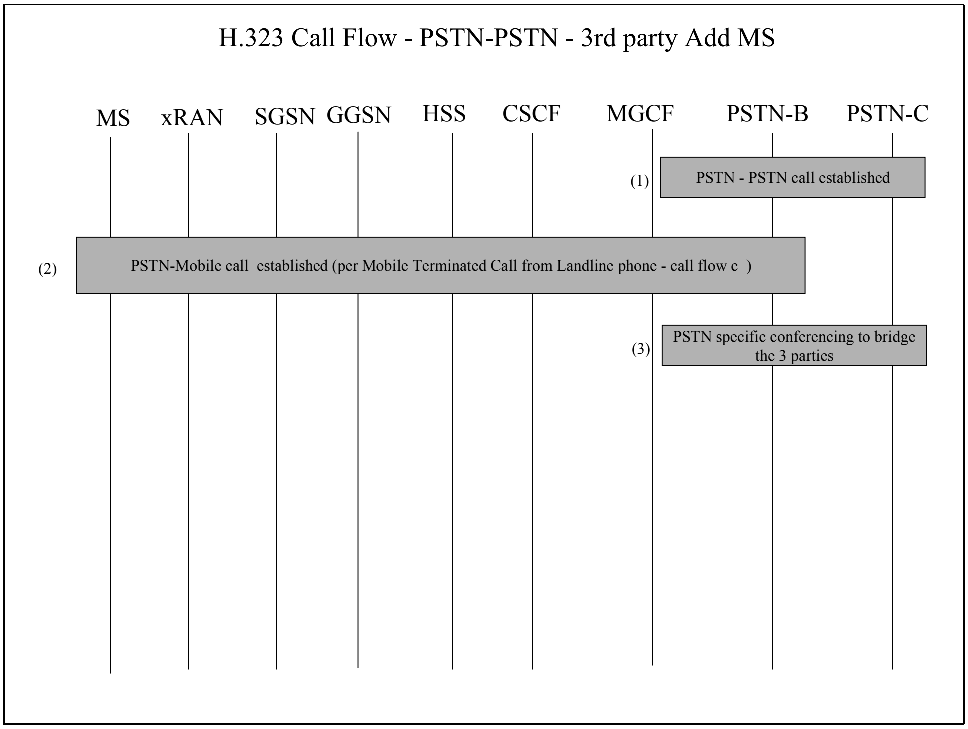


(1) A mobile originated call with party B in the PSTN has been established per call flow (a). In addition a facility message exchange is sent to PSTN-B to create an adhoc conference detailing a multicast address to use and appropriate multicast and media information.
(2) The SETUP message is sent by the MS to the CSCF. The CSCF does a “translation” and determines that interfacing through the MGCF is necessary to establish a call with party C. The CSCF sends SETUP to the MGCF to establish a call with Party C.
(3) Upon receipt of the SETUP message, The MGCF starts exchanging a series of ISUP messages with the PSTN to set-up the call to party C.
(4) On receiving the CPG message from the PSTN, the MGCF generates a H.225 Alerting message to indicate the called user’s phone is ringing. The MGCF forwards the message to the CSCF.
(5) The CSCF forwards the H.225 Alerting message to the MS. The MS starts ringback at this point.
(6) On receiving an ANM message from the PSTN, the MGCF sends a Connect message to the CSCF. The CONNECT can use the ANNEX E header with the AckRequest field set.
(7) Upon receipt of the Connect message, the CSCF forwards the Connect message to the MS.
(8) The MS responds with an ANNEX E ACK message which may be used to trigger billing. The voice is cut-through.
(9) The MS responds with an ACK message. The CSCF sends a Release Complete (cause) to the MS to Release the initial “call” to Party B since the call is now connected through the MRF.

3.18.6 Landline party adding in a third party (mobile on UMTS Release 2000 network) in an existing landline to landline connected call

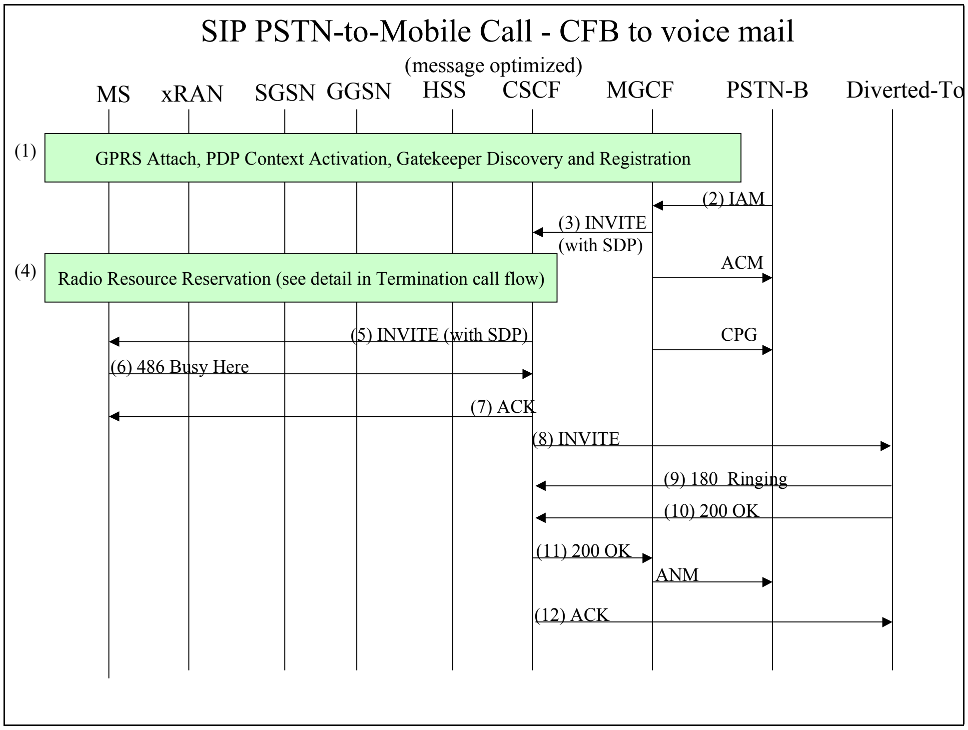


(1) A PSTN-PSTN call is established.
(2) Party B in the PSTN terminates a call to the MS per call flow c.
(3) Upon successful establishment of the mobile terminated call, Party B in the PSTN initiates PSTN specific signaling to bridge the mobile terminated call with the call established with Party C.

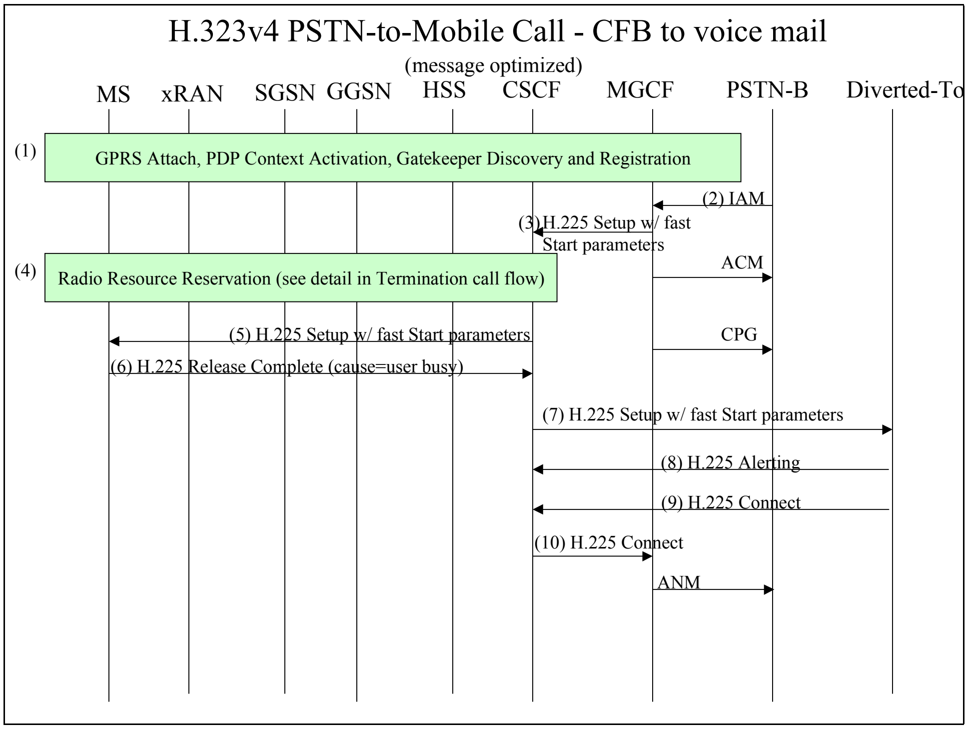


(1) A PSTN-PSTN call is established.
(2) Party B in the PSTN terminates a call to the MS per call flow c.
(3) Upon successful establishment of the mobile terminated call, Party B in the PSTN initiates PSTN specific signaling to bridge the mobile terminated call with the call established with Party C.

3.18.7 Mobile terminated call from a landline phone, mobile busy, forwarded to voice mail

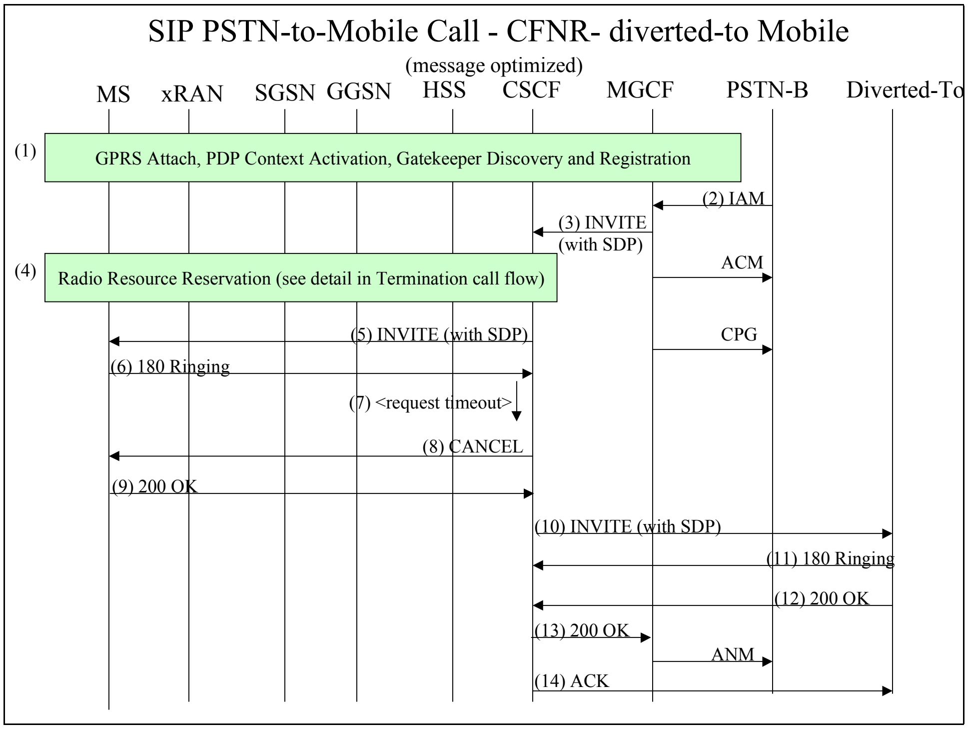


(1) The MS performs a GPRS attach, primary PDP context activation and CSCF discovery.
(2) MGCF receives an IAM message from the PSTN.
(3) MGCF sends a SIP INVITE message (with SDP element) to the CSCF. It also sends the ACM and CPG messages back to the PSTN.
(4) The radio access resource reservation is accomplished per the Mobile Termination call flow.
(5) The CSCF forwards the INVITE message (with SDP element) to the MS.
(6) On receiving the INVITE message, the MS (who is busy on a call) sends a 486 Busy Here message to the CSCF.
(7) Upon receipt of the 486 Busy Here message, the CSCF sends an ACK message to the MS in response to 486 Busy Here.
(8) The CSCF determines the diverted-to party by accessing the UMS entry for the MS (or using subscriber info cached in the serving system during the CSCF registration process). The CSCF sends an INVITE message to the requested Diverted-To party (voice mail in this case, which is assumed to be a SIP device for this scenario).
(9) The Diverted-to party responds with a 180 Ringing message.
(10) Once the voice mailbox of the mobile is connected to the original calling party in the PSTN, a 200 OK message is sent to the CSCF (Note: the 200 OK message can trigger billing here).
(11) Upon receipt of the 200 OK message, the CSCF forwards the 200 OK to the MGCF. Upon receipt of the 200 OK, the MGCF sends an ANM to the PSTN Calling Party (indicating that the call has terminated to another party - i.e. voice mail).
(12) The CSCF responds to the 200 OK message with an ACK to the Diverted-To party.

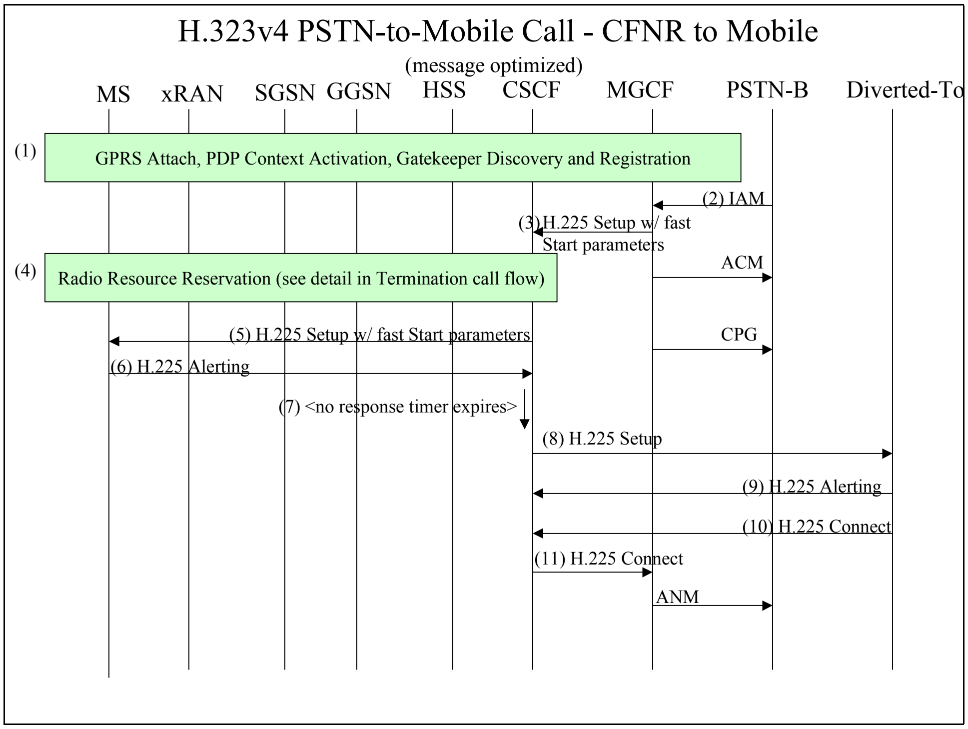


(1) The MS performs a GPRS attach, primary PDP context activation and CSCF discovery.
(2) MGCF receives an IAM message from the PSTN.
(3) MGCF sends a H.225 SETUP message (with FastStart element) to the CSCF. It also sends the ACM and CPG messages back to the PSTN.
(4) The radio access resource reservation is accomplished per the Mobile Termination call flow.
(5) The CSCF forwards the H.225 SETUP message (with FastStart parameters) to the MS.
(6) On receiving SETUP message, the MS (who is busy on a call) sends an H.225 RELEASE COMPLETE message with cause=user busy.
(7) Upon receipt of the RELEASE COMPLETE message with cause=user busy, the CSCF invokes the CFB feature. The CSCF determines the diverted-to party by accessing the UMS entry for the MS (or using subscriber info cached in the serving system during the CSCF registration process). The CSCF then sends a SETUP message to the diverted-to party (voice mail in this case, which is assumed to be an H.323 device for this scenario).
(8) The diverted-to party responds with an ALERTING message.
(9) Once the voice mailbox of the mobile is connected to the original calling party in the PSTN, a CONNECT message is sent.
(10) Upon receipt of the CONNECT message, the CSCF sends a CONNECT message to the MGCF. The MGCF sends an ANM to the original calling party (indicating that the call has terminated to another party - i.e. voice mail)

3.18.8 Mobile terminated call from a landline phone, no answer, forwarded to subscriber defined mobile phone



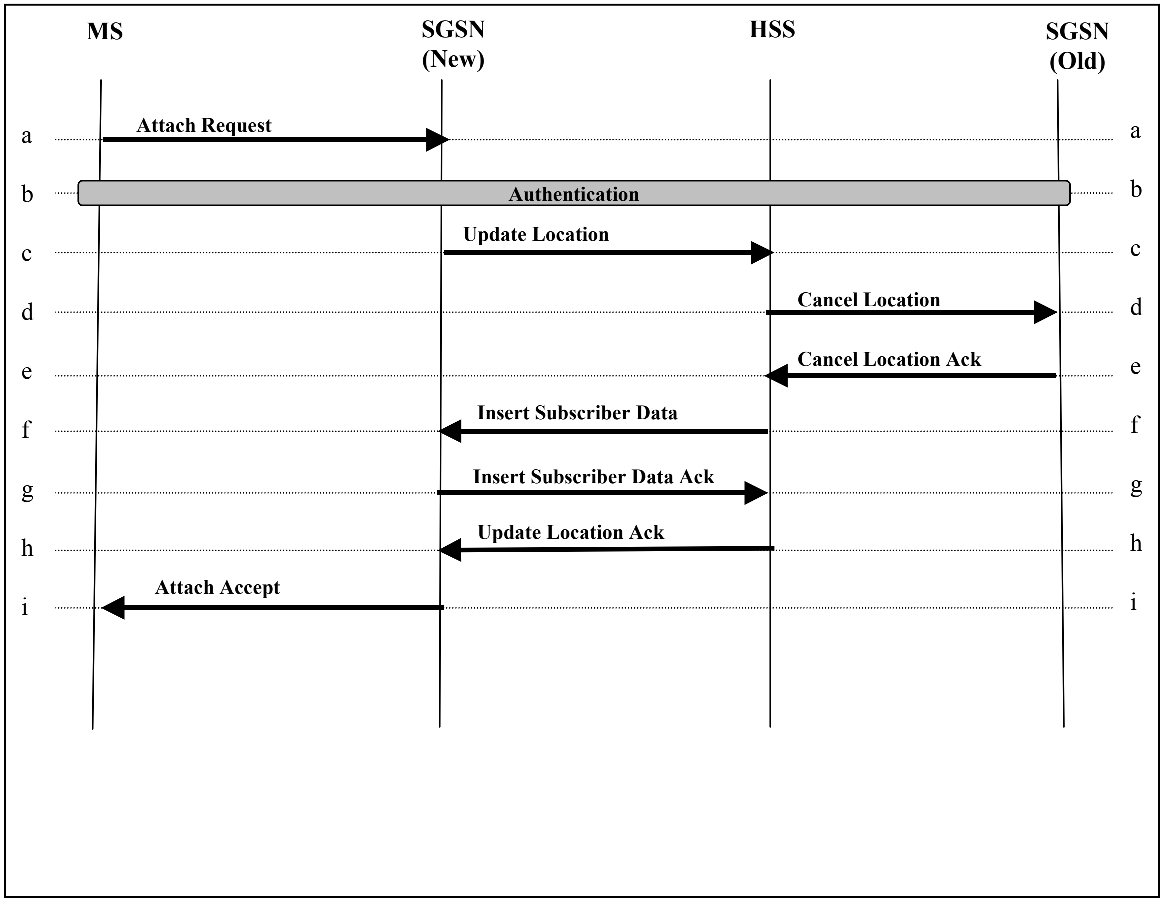
(1) The MS performs a GPRS attach, primary PDP context activation and CSCF discovery.
(2) MGCF receives an IAM message from the PSTN.
(3) MGCF sends a SIP INVITE message (with SDP element) to the CSCF. It also sends the ACM and CPG messages back to the PSTN.
(4) The radio access resource reservation is accomplished per the Mobile Termination call flow.
(5) The CSCF forwards the INVITE message (with SDP element) to the MS.
(6) On receiving INVITE message, On receiving the INVITE message, the MS generates the ringing and sends a 180 Ringing message to the CSCF indicating that the called user's phone is ringing .
(7) The subscriber does not answer the ringing mobile and the request timeout expires.
(8) The CSCF invokes the CFNR feature and sends a CANCEL message to the MS.
(9) The MS acknowledges the CANCEL with a 200 OK.
(10) Upon receipt of the 200 OK message, the CSCF sends an INVITE to the Diverted-To mobile (Note: this sending of the INVITE is equivalent to step (7) of the Mobile Termination SIP call) .
(11) On receiving the INVITE message, the Diverted-To party generates the ringing and sends a 180 Ringing message to the CSCF indicating that the called user's phone is ringing .
(12) When the called party answers the phone, the Diverted-To party generates a 200 OK message and sends it to the CSCF (Note: the 200 OK message can trigger billing here).
(13) The CSCF forwards the 200 Okay message to the MGCF. The MGCF generates an ANM message and sends it to the PSTN.
(14) The CSCF sends an ACK message to the Diverted-To party. After this, the voice is cut-through.
(Note: that only a subset of the entities involved in this flow for steps 10-14 message are shown - they are equivalent to the mobile termination call flow since the Diverted-To party is a mobile.)



(1) The MS performs a GPRS attach, primary PDP context activation and CSCF discovery.
(2) MGCF receives an IAM message from the PSTN.
(3) MGCF sends a H.225 SETUP message (with FastStart element) to the CSCF. It also sends the ACM and CPG messages back to the PSTN.
(4) The radio access resource reservation is accomplished per the Mobile Termination call flow.
(5) The CSCF forwards the H.225 SETUP message (with FastStart parameters) to the MS.
(6) On receiving SETUP message, the MS generates the ringing (or may also receive in-band ringing tone from PSTN at this point) and sends a H.225 ALERTING message to the CSCF indicating that the called user's phone is ringing.
(7) The subscriber does not answer the ringing mobile and the no response timer expires.
(8) The CSCF invokes the CFNR feature and forwards the SETUP message to a subscriber defined mobile phone.
(9) Though all the entities involved are not shown , this step is equivalent to step (7) of the PSTN to mobile termination call flow. The Diverted-To MS sends an ALERTING message to the CSCF.
(10) Upon receipt of the CONNECT message from the Diverted-To MS, the CSCF forwards a CONNECT message to the MGCF. (Note: that the entities involved in this message are equivalent to step (8) of the PSTN to mobile termination call flow.)
(11) Upon receipt of the CONNECT message from the CSCF, the MGCF sends an ANM to the original calling party.

3.18.9 Mobile originated call while roaming in another UMTS Release 2000 network

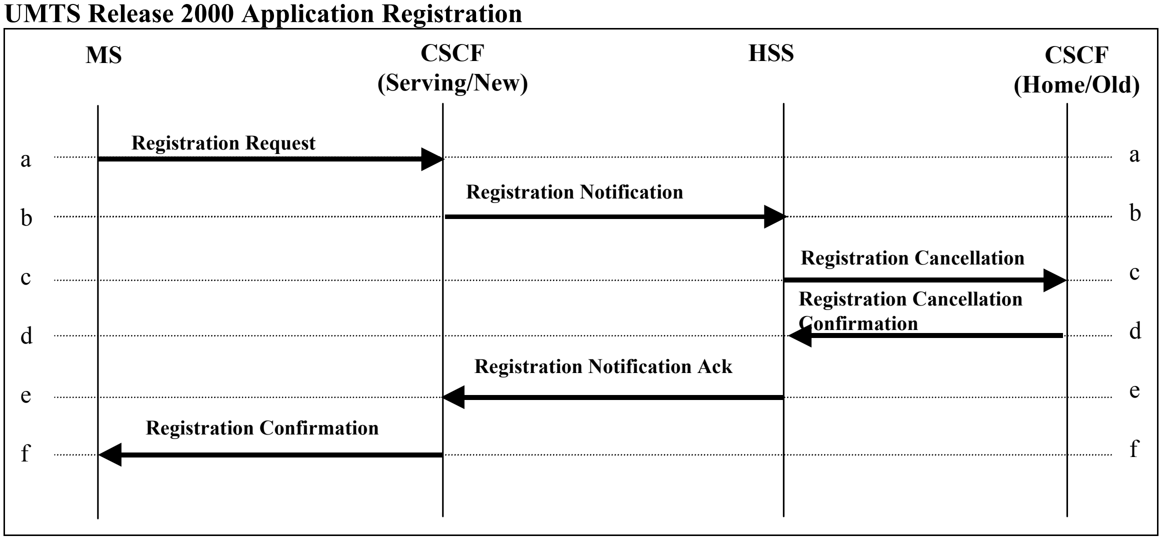
UMTS Release 2000 Attach/Registration



a) The MS initiates the attach procedure by the transmission of an Attach Request message to the SGSN. If message is received with P-TMSI, the SGSN sends an Identity Request message to the old SGSN to get corresponding IMSI. If received IMSI is unknown, the SGSN, the SGSN sends Identity Request message to MS. If Attach Request is received with IMSI, this procedures are not required.
b) Authentication is performed based on the security information stored in HSS.
c) The SGSN then sends an Update Location message to HSS.
d) The HSS sends an Cancel Location message to home/old SGSN with cancellation type set to Update Procedure.
e) The old SGSN recognizes request. If there are no ongoing transaction exists, it acknowledges with Cancel Location Ack.
f) The HSS sends Insert Subscriber Data to the new SGSN. If recent cache of the profile is available from the local directory in UMTS Release 2000 network, this message may not include subscriber information. In this case, it is assumed that SGSN will access that information directly from local cache.
g) The new SGSN validates the MS's presence in the new routing area. If all checks are successful then the SGSN constructs an MM context for the MS and returns Insert Subscriber Data Ack message.

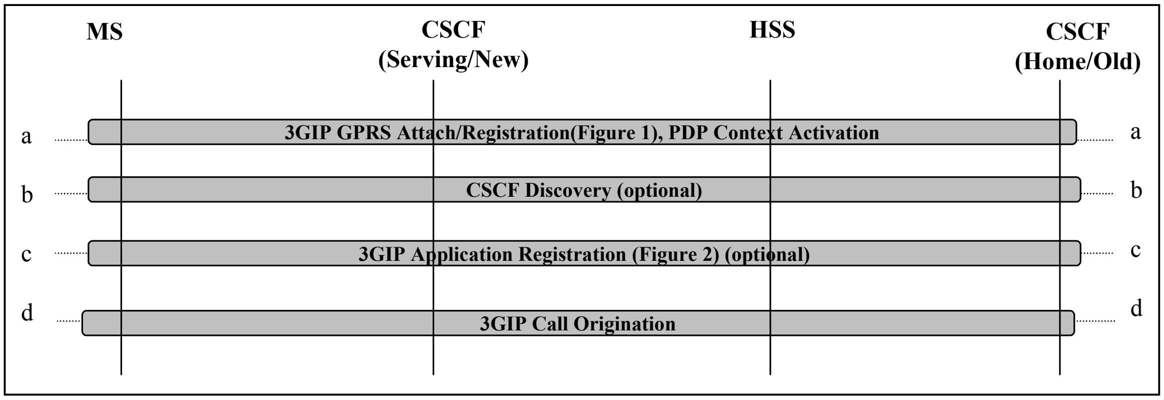
h) The HLR acknowledges the Update Location message by sending an Update Location Ack to the SGSN.
i) The SGSN confirms completion of attachment and registration by sending Attach Accept message to MS.

UMTS Release 2000 Application Registration



Description
Application Registration procedure discussed here is an optional method to register serving CSCF with HSS. Alternative method may include caching Serving CSCF information as a part of CSCF Discovery procedure mentioned in step (b) of Figures III and IV or some other means.
a) The MS initiates application registration by sending Registration Request message to CSCF with subscriber identity. Note that MS obtained identity of CSCF from "CSCF Discovery" procedure.
b) Serving CSCF requests subscriber profile from HSS by sending a Registration Notification message.
c) The HSS sends a Registration Cancellation message to old CSCF.
d) The old CSCF erases current information of the subscriber and acknowledges the request by sending Registration Cancellation Ack.
e) The HSS provides subscriber information to serving SGSN in Registration Notification Ack message.
f) Application Registration is confirmed to MS by sending a Registration Confirmation message by serving SGSN.

UMTS Release 2000->UMTS Release 2000 Call Origination



Description

a) Mobile performs Attach/Registration procedure as discussed above. It also activates PDP context using "PDP Context Activation Procedure" discussed in GSM Recommendations 3.60.

b) As discussed in Call Control Baseline document, MS performs two types of Registration procedures

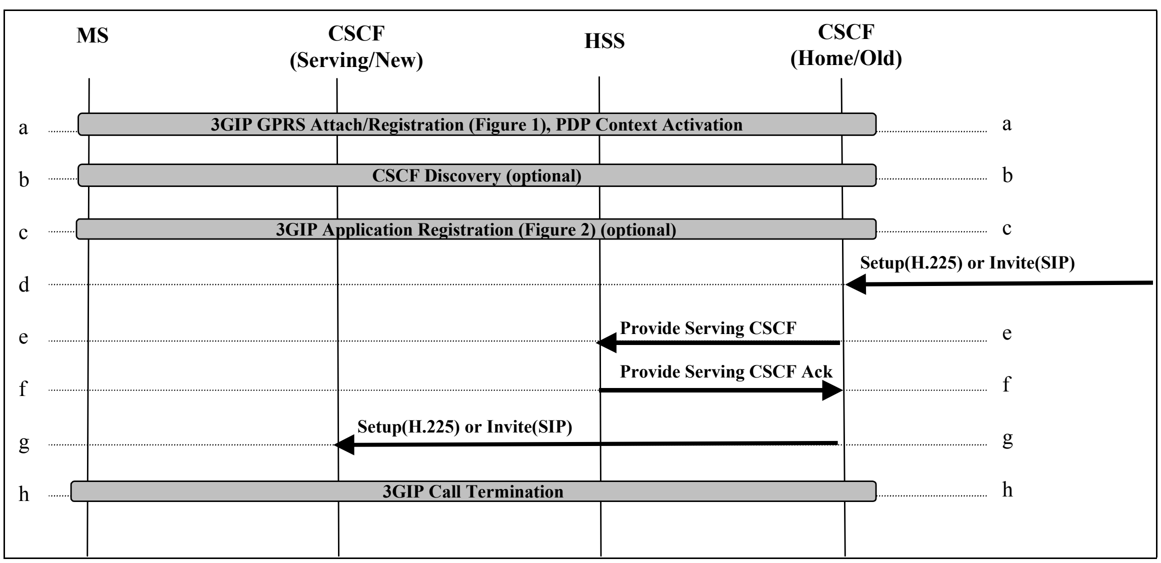
- GPRS Registration which includes steps shown in Figure 1
- Application Registration which includes steps shown in Figure 2

Application Registration requires identity of serving CSCF. Hence, "CSCF Discovery" procedure is performed to identify serving CSCF. It is assumed here that this procedure will be same as being worked on in UMTS Release 2000 Call Control group.

c) The MS initiates Application Registration as shown in "Figure 2 UMTS Release 2000 Application Registration" scenario.

d) Rest of the steps should be same as discussed for UMTS Release 2000 Call Origination scenario.

3.18.10 Mobile terminated call while roaming in another UMTS Release 2000 network



Description

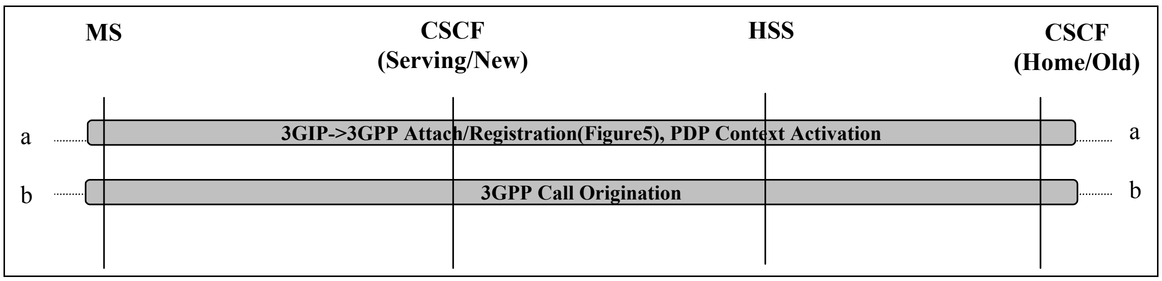
a) Mobile performs Attach/Registration procedure as discussed above. It also activates PDP context using "PDP Context Activation Procedure" discussed in GSM Recommendations 3.60.

b) As discussed in Call Control Baseline document, MS performs two types of Registration procedures

- GPRS Registration which includes steps shown in Figure 1
- Application Registration which includes steps shown in Figure 2

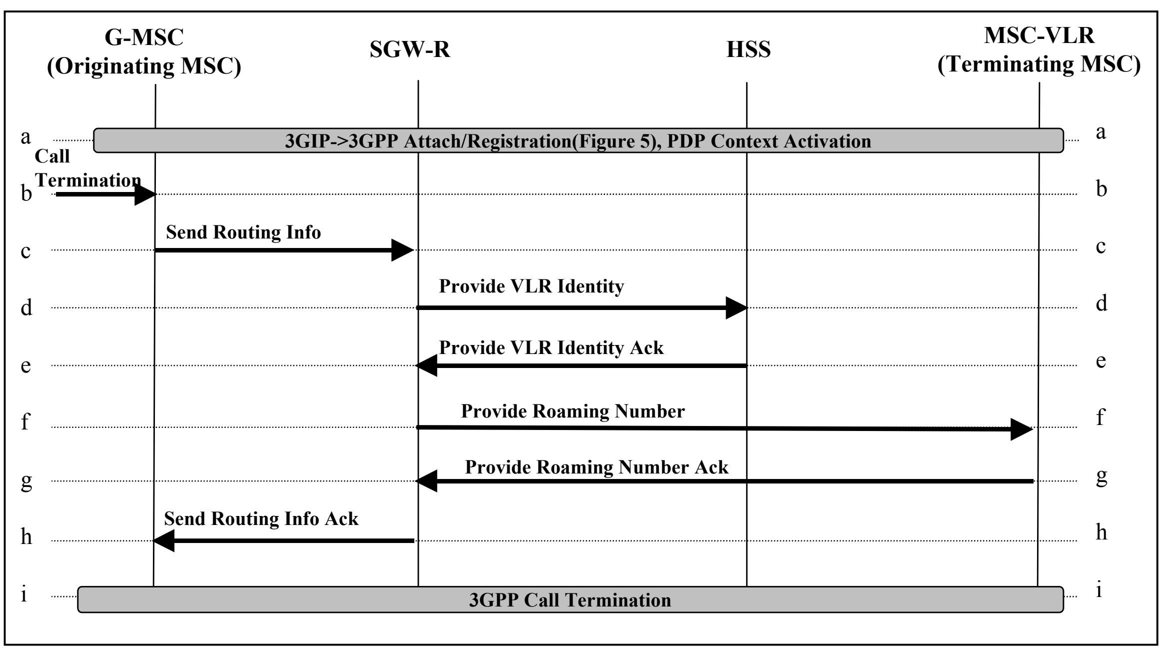
Application Registration requires identity of serving CSCF. Hence, "CSCF Discovery" procedure is performed to identify serving CSCF. It is assumed here that this procedure will be same as being worked on in UMTS Release 2000 Call Control group.
c) The MS initiates Application Registration as shown in "Figure 2 UMTS Release 2000 Application Registration" scenario.
d) Home CSCF receives call termination request in form of either H.225-Setup or SIP-Invite message.
e) Upon receipt of either H.225-Setup or SIP-Invite message, Home CSCF validates user and sends a Provide Serving CSCF message to HSS.
f) Based on the user identity, the HSS authorized user and check users' service capabilities. If successful, it sends serving CSCF address in Provide Service CSCF Ack message to Home CSCF.
g) Home CSCF forwards H.225-Setup or SIP-Invite message to serving CSCF.
h) Rest of the steps should be same as discussed for UMTS Release 2000 Call Termination scenario.

3.18.11 Mobile originated call while roaming in another 3GPP network



Description
a) Mobile performs Attach/Registration procedure as shown in "Figure 5 UMTS Release 2000->3GPP Attach/Registration" . It also activates PDP context using "PDP Context Activation Procedure" discussed in GSM Recommendations 3.60.
b) Rest of the steps should be same as discussed for 3GPP Call Origination scenario.

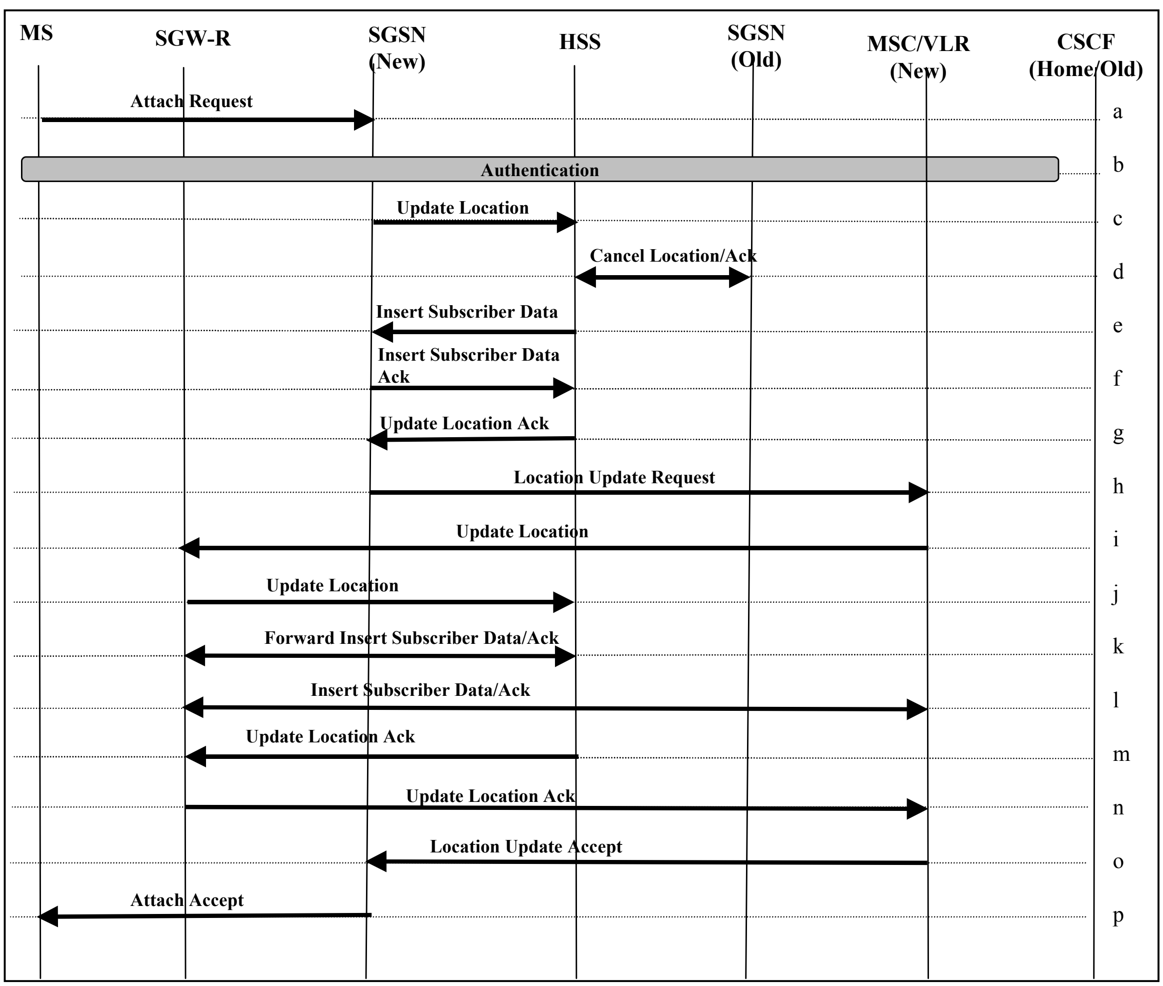
3.18.12 Mobile terminated call while roaming in another 3GPP network



Description

a) Mobile performs Attach/Registration procedure as shown in "Figure 5 UMTS Release 2000->3GPP Attach/RegistrationLocation Ack" . It also activates PDP context using "PDP Context Activation Procedure" discussed in GSM Recommendations 3.60.
b) Gateway MSC or (originating MSC in case of Optimal Routing Support) receives a call termination request.
c) G-MSC/originating MSC sends a Send Routing Information message to SGW-R to get Mobile Subscriber Roaming Number (MSRN). Here, SGW-R is acting as HSS/HLR for 3GPP network.
d) The SGW-R forwards this information by sending a Provide VLR Identity message to HSS.
e) HSS acknowledges this request by sending a Provide VLR Identity Ack message. This message contains MSC/VLR number. Note that this number is different from MSRN number.
f) Using the information received in the Provide VLR Identity Ack message, the SGW-R sends a Provide Roaming Number message to new MSC/VLR.
g) The MSC/VLR sends MSRN in a Provide Roaming Number Ack message to SGW-R. This message includes MSRN.
h) The SGW-R forwards MSRN to G-MSC/originating MSC in a Send Routing Information Ack message.
i) Using MSRN received in the Send Routing Information Ack message, G-MSC/ originating MSC routes incoming call to new MSC/VLR. Rest of the steps should be same as discussed for 3GPP Call Termination scenario.

3.18.13 SIM roaming into a UMTS Release 2000 network



Description

a) The MS initiates the Attach procedure by the transmission of an Attach Request message to the SGSN. If message is received with P-TMSI, the SGSN sends an Identity Request message to the old SGSN to get corresponding IMSI. If received IMSI is unknown, the SGSN, the SGSN sends Identity Request message to MS. If Attach Request is received with IMSI, these procedures are not required.
b) Authentication is performed based on the security information stored in HSS.
c) The SGSN then sends an Update Location message to HSS.
d) The HSS sends an Update Location message to home/old SGSN with cancellation type set to Update Procedure. The old SGSN recognizes request. If there are no ongoing transaction exists, it acknowledges with Cancel Location Ack.
e) The HSS sends Insert Subscriber Data to the new SGSN.
f) The new SGSN validates the MS's presence in the new routing area. If all checks are successful then the SGSN constructs an MM context for the MS and returns Insert Subscriber Data Ack message.
g) The HLR acknowledges the Update Location message by sending an Update Location Ack to the SGSN.
h) For updating location information, the SGSN sends a Location Update Request message to the VLR. Using information in this message, the VLR creates an association with the SGSN by storing SGSN Number.
i) Since MS is roaming from UMTS Release 2000 location area to 3GPP location area, the new VLR sends Update Location to the Roaming Signaling Gateway (SGW-R). Here, SGW-R is acting as HSS/HLR for 3GPP network.
j) SGW-R sends a Registration Notification message to HSS.
k) The HSS sends subscriber profile in a Forward Insert Subscriber Data message to SGW-R. The SGW-R immediately acknowledges this message by sending Forward Insert Subscriber Data Ack message.
l) The SGW-R also forwards subscriber profile information received from HSS to the new VLR in Insert Subscriber Data message to new VLR. The new VLR acknowledges this message by sending an Insert Subscriber Data Ack message to SGW-R.
m) This completes retrieval of subscriber profile from UMTS Release 2000 HSS. HSS confirm this by sending a Registration Notification Confirmation message to SGW-R.
n) The SGW-R forwards this confirmation to MSC/VLR by sending a Update Location Ack message.
o) New VLR acknowledges location update request by sending a Location Update Accept message to SGSN.
p) The SGSN confirms completion of attachment and registration by sending Attach Accept message to MS.

3.19 Describe and compare the security mechanisms in place with H.323 and SIP, and how these mechanisms meet operator requirements for fraud mitigation and end-user privacy.

SIP requests and responses can be authenticated and/or encrypted end to end using public key encryption. The To, From and Via headers are not encrypted as these are needed to properly traverse proxies and firewalls. Messaging can be encrypted hop by hop to protect against sniffers. Anonymizers can be used hop by hop on the From and To headers for additional privacy. VIA headers control the route of the message. They are removed on the return path so the is unable to obtain the topology information. If proxies are employed then the VIA headers can be stripped and cached on the forward path as well. This allows the caller’s topology information to be withheld as well. SIP was defined with security in mind and it’s affect on firewalls as well as wide area addressing. The packetcable consortium has spent the most time furthuring these exact encryption and authorization schemes to use in regards to SIP. SIP seems to have accounted for wide area addressing issues a bit better than H.235.

H.323 security was added as another specification H.235. H.235 provides for several types of encryption and authentication mechanisms. The INOW consortium has spent the most time picking the exact encryption and authorization schemes to use.

3.20 Explain how the UMTS Release 2000 architecture with the H.323 and SIP protocols support regulatory services, such as lawful surveillance and emergency services (including location).

Neither have explicitly defined these functions but they are somewhat protocol agnostic. The dcsgroup has published drafts and specifications with a proposed solution. Lawful interception is achieved by the addition of a new header using the dcsgroup proposal. H.323 could be altered using the same mechanism. Another mechanism would be to simply have the serving proxies perform authentication and authorization on the lawful entities wishing to intercept to data. The existing media description can be included in existing signaling that the interceptor is receive only or needs 2-way communication.

Emergency services have yet to be published by any SIP or H.323 proponent. The proxies when detecting an emergency number can query the end system to provide the granularity of location information that is necessary. An extension can be added to carry this information. The INFO method can be used for subsequent queries for location.

Similar methods can also be applied to H.323.

3.21 Recommendation

Which protocol does your company recommend for the UMTS Release 2000 architecture? Please list reasons why based on the above criteria and questions.

Criteria	H.323	SIP	Choice/Reason
Complexity	**Very complex**	**Simple**	**SIP – TTM / reduced complexity of development**
Message Set	Complex, many messages for similar functionality	Logically numbered responses for extension, smaller set of messages for same funcitonality	SIP – TTM and extensibility
Debugging	Have to alter tools on each extension.	Simple Tool developed once	SIP – TTM / reduced complexity of development
Re-use of code	H.323 and H.32x	SIP and Web	SIP – more modular
Service and Protocol Interactions	H.323 and H.32x	SIP and Web - more modular	SIP – more modular

Methods for implementing services	Can support all	Can support all	Equivalent.
Distributed Call Signaling	Can Support	Can Support	SIP – TTM / reduced complexity
Extensibility	**Extensible**	**More Extensibility**	**SIP – more options for extension**
Version Compatability	YES	YES – the Requires, Supported and Proxy-Require headers provide more flexibility than H.323	SIP – more flexibility to support for multiple variants co-existing.
Feature Evolution	Same as above	Same as above	Same as above
Operators in charge of own services	Less Ability – more complex ASN.1	Higher Ability – text formats and extension headers.	SIP – Operators will be less dependent on vendors to add new services.
Modularity	Umbrella Standard – designed for limited feature set.	Modular designed around other web technologies and can do GSTN services too.	SIP – built for web. H.323 originally derived from circuit world.
Codecs	Equivalent	Equivalent	Equivalent
3rd party CC	Facility redirect	Also header	Equivalent
Scalability	**Installed base designed for reliable transport**	**Designed for it**	**Equivalent**[1]
Wide Area Support	YES	YES	Equivalent
Large Number of Calls	YES	YES	Equivalent
Call States	Can do both	Can do both	Equivalent
Elements that must maintain states	Clients, MC, MGCF – CSCF optional	UA, MC, MGCF, CSCF optional	Equivalent
Msg processing	More processor overhead, smaller messages	Less processor overhead, larger messages	Comparable – bandwidth vs. component complexity decision
Conferencing	All modes – H.224 floor control	All modes – GCCP, SCCP or even H.224	Comparable – No RFC exists saying

[1] There are still issues with reliability for H.323 over UDP as there is no session layer acknowledgement to the connect defined. Installed based designed to be more stateful with reliable underlying transport.

		floor control	which to use for SIP.
DCS	Would have to be altered more than SIP	A closer original design fit	SIP – TTM
Resources	**No opinion**	**No opinion**	**Comparable**
Air-link bandwidth	Smaller Messages	Larger Messages	H.323 – smaller messages
CPU	More processing	Less processing	SIP – less processing
QOS/RRM Interactions	Same Issues	Same Issues	Equivalent
Services	High TTM	Low TTM	SIP – long term lower TTM and complexity
Supported Services	H.323 more explicitly defined	SIP defined in whitepapers/drafts	Equivalent but H.323 better standardization
Delay Times	Equivalent – still issues with use of UPD and reliability which are related	Equivalent	Equivalent
Billing	Needs work – consortia defined imbedded in protocol	Needs work – consortia defined – separate protocol	Comparable
GSTN services	YES	YES	SIP – TTM / less code
Capabilities Exchange	Better for media – worse for signaling extensibility	Worse for media – better for signaling extensibility	SIP – signaling is more of an issue.
Personal Mobility	Added nomadicity later v3 – location based services still ongoing	Designed for nomadicity – location based services still ongoing	Comparable
Legacy interoperability	H.246	Draft status	H.323
IP telephony interoperability	Monolithic / OS bundled client	DCSGROUP/ MGCP/ SDP	SIP
Security	H.235 added later. Worse for firewall traversal using UDP.	Designed for it originally. Better for firewall traversal.	Comparable
OA&M	Equivalent – consortia defined	Equivalent – consortia defined	Equivalent
Procedures available	Loop Back	Invite with SDP loopback media value and no alerting option	Comparable – both have MIBs defined by consortia.
Fault Detection	See above	See above	See above

3.21.1 CONCLUSION

The industry is rifted right now with this issue. We all owe a great deal to the authors of each set of protocols and the early implementers for getting us to the point we are now.

The investment that each company has already made and will make in this new global infrastructure warrants serious long term consideration.

H.323 does have more enterprise oriented and campus scale products deployed right now than SIP; however, the long term benefits related to and affecting time to market, extensibility, multi-party service flexibility, ease of interoperability, and complexity of development considerations lead us to recommend SIP.

Some business related questions that may warrant further consideration for the decision process are:

- Based on where the majority of investment capital is going right now, is it a higher priority to fit in with N-ISDN and B-ISDN circuit based multimedia telecommunications applications than it is to interwork and provide multimedia services designed for the web and it's technologies such as the browser?
- What is the extent of deployment for N-ISDN and B-ISDN multimedia equipment?
- If both protocols allow GSTN interoperability, which would be a better investment for both the short and long term?
- Compared to N-ISDN and B-ISDN multimedia, what is the extent of deployment for GSTN equipment.
- As operators, which protocol would allow you to deploy new services that require protocol changes with the least amount of intervention from vendors?
- As operators, which protocol would allow you to deploy the most services without any changes to the protocol at all?

We do feel that there is a need to support both protocols and provide interworking. Delivery times and resources applied to each are dependent on what operators, standards and consortia decide. A choice of a single protocol will have the effect of reduced complexity of development and time to market.

3.22 References

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