ATIS STANDARD

ATIS-1000027

OPERATOR SERVICES IN A NEXT GENERATION NETWORK (NGN) ENVIRONMENT

TECHNICAL REPORT

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ATIS-1000027, *Operator Services in a Next Generation Network (NGN) Environment*

Is an ATIS Standard developed by the **Signalling, Architecture, and Control (SAC) Subcommittee** under the **ATIS Packet Technologies and Systems Committee (PTSC)**.

Published by

Alliance for Telecommunications Industry Solutions
1200 G Street, NW, Suite 500
Washington, DC 20005

ATIS-1000027

Technical Report on

Technical Report on Operator Services in a Next Generation Network (NGN) Environment

Alliance for Telecommunications Industry Solutions

Approved February 2008

Abstract

This Technical Report (TR) describes the traditional set of operator services and considers how corresponding services could be supported the IP environment of a Next Generation Network (NGN).

Foreword

The Alliance for Telecommunication Industry Solutions (ATIS) serves the public through improved understanding between providers, customers, and manufacturers. The Packet Technologies and Systems Committee (PTSC) develops and recommends standards and technical reports related to services, architectures, and signaling, in addition to related subjects under consideration in other North American and international standards bodies. PTSC coordinates and develops standards and technical reports relevant to telecommunications networks in the U.S., reviews and prepares contributions on such matters for submission to U.S. ITU-T and U.S. ITU-R Study Groups or other standards organizations, and reviews for acceptability or per contra the positions of other countries in related standards development and takes or recommends appropriate actions.

The mandatory requirements are designated by the word *shall* and recommendations by the word *should*. Where both a mandatory requirement and a recommendation are specified for the same criterion, the recommendation represents a goal currently identifiable as having distinct compatibility or performance advantages. The word *may* denotes a optional capability that could augment the standard. The standard is fully functional without the incorporation of this optional capability.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, PTSC, 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, PTSC, which was responsible for its development, had the following roster:

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The Signalling, Architecture, and Control (SAC) Subcommittee was responsible for the development of this document.

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1 Introduction

This Technical Report (TR) describes the traditional set of operator services and considers how corresponding services could be supported the IP environment of a Next Generation Network (NGN). The services are seen as being composed of three distinct sets of functions:

1. Delivery of a call to the Next Generation Network (NGN) Operator Services System (OSS)
2. Call completion from an NGN OSS
3. Information delivery by an NGN OSS.

Operator service has historically begun with a voice call from the calling party to the OSS (either a live operator or an automated system) to request a particular service. The introduction of the NGN network architecture holds the potential to change how and if traditional operator services are provided. Current standards activities related to the evolution of these services to an IP-based NGN environment are reviewed and gaps are identified.

2 Definition of Information and Operator Services

Information and operator services are grouped into the following categories:

1. Information Requested (e.g., 411 dialed). This category includes:
    a. Directory Assistance (DA)
    b. DA Call Completion
    c. Non-Published Messaging
2. Operator Requested (e.g., 0- dialed). This category includes:
    a. Assistance
    b. Busy Line Verification (BLV)
    c. Operator Interrupt (OI)
    d. Transfer to a Emergency Services
3. Alternate Billing (i.e., 0+ dialed)
4. Network Assistance (i.e., the network provides assistance when the customer does not dial an operator-services-specific number). This category includes:
    a. Intercept
    b. Coin Services
    c. Handling of Restricted Lines.

The specific services listed above are defined in this section.

2.1 Information Requested Services

2.1.1 Directory Assistance/ Directory Assistance Call Completion

The telephone number for a given name and address is provided by an operator and/or automated system.

Call completion to the telephone number is an option available to the customer.

When requested explicitly or implicitly by the customer, an operator can complete a call for a customer to the directory listing number that is found.

Operator recall returns the customer to an operator when signaled by the customer (e.g., by depressing the flash-hook button.)

2.1.2 Non-Published Messaging

At a customer’s request, an operator can transmit a callback request to a party with a non-published number.

2.2 Operator Requested Services

2.2.1 Assistance

An operator can provide dialing instructions and rate information.

2.2.2 Busy Line Verification

An operator can verify whether a line is busy. This service is usually followed by Interrupt service if conversation is detected on the busy line.

2.2.3 Interrupt

An operator can, if necessary, interrupt the line, i.e., may join a conversation that has been detected using the Busy Line Verification (BLV) service.

There are two options by which the interruption may be invoked:

- If BLV is performed by attaching the operator to the target line with a scrambler to allow detection of conversation, then Interrupt service can be invoked by removing the scrambler.
- The service may be invoked as a separate call to the target line after Busy Line Verification.

2.2.4 Transfer to Emergency Services

An operator can transfer a customer to the nearest Public Safety Answering Point (PSAP) if a customer dials “0-“ rather than “9-1-1.”

2.2.5 Transfer to Inter-exchange Carrier

An operator can transfer a customer to various numbers.

2.3 Call Completion

An operator can complete a call for a customer. Operator recall returns the customer to an operator after the customer depresses the flash-hook.

2.4 Alternate Billing

Calling card / commercial credit card calls, collect calls (also known as Reverse Charging), or calls billed to a third party.

This service includes verification of the calling card / commercial credit card number or verifying that the called party or third party will accept the charges for the call.

2.5 Network Assistance

2.5.1 Coin Services

An operator can provide real-time rating of calls from coin telephones.

2.5.2 Connection Hold and Ring-Back

The connection hold and ring-back features allow the operator to remain in control of a coin station even after the calling party disconnects. These features may also be applied to non-coin stations for emergency services.

2.5.3 Intercept

Calls to non-working telephone numbers are intercepted; the calling party is informed that the number is no longer in service and the reason why it is not in service. The calling party may also be informed of a new working telephone number and the call may be completed to the new number.

2.5.4 Handling of Restricted Lines

Calls from certain restricted lines are forwarded to an operator for special handling.

3 NGN/IP Architecture

In this Technical Report (TR), the network architecture that provides the context for Operator Services is taken from the ATIS Technical Report on NGN Architecture [ATIS-1000018] with the assumption that the Operator Services System will be a part of a Voice over IP (VoIP) network.

3.1 Terminology

The following terms are used in this document.

Operator Services – The set of services currently accessed by dialing 411, NPA-555-1212, 0+10 digits, or 0- (i.e., the zero digit with no further dialing). Note that some or all of the services in the set may also be accessed via other dialing patterns.

Operator Services Provider (OSP) - The OSP is the provider of operator services to end users. The OSP provides retail services directly to end users, and provides wholesale services to other Services Providers, such as a Home Services Provider or Aggregation Services Provider.

Specifically, the OSP provides services to the end users that are served by Home Services Providers. The OSP needs to identify the Home Services Provider in order to provide such services as customized announcements, and may also need to know the identity of other intermediate entities for billing purposes.

Home Services Provider (HSP) - The HSP is the subscribed provider of voice services to the calling end user. In the case of a roaming mobile end user, the HSP may not be the network that originates the call. However, recognizing that current call setup procedures route such calls through the HSP, this TR assumes that all calls appear to the Operator Service as having originated in the HSP’s network. The HSP may be directly connected to the OSP or may communicate with the OSP via an Aggregation Services Provider. The HSP may also be an Aggregation Services Provider or OSP.

Aggregation Services Provider - The Aggregation Services Provider is the provider that routes DA calls from multiple HSPs to a DA provider. The Aggregation Services Provider may be an HSP and an HSP may send traffic to a single DA provider using multiple Aggregation Services Providers.

4 Application of NGN Architecture to the Operator Services System

Figure 1 illustrates the application of the NGN/IP architecture to operator services, based on the ATIS NGN Architecture [ATIS-1000018]. Most naturally and flexibly, one can think of an Operator Services System (OSS) within the general VoIP architecture context of application servers and media servers. Incoming calls may be first signaled to the operator services Application Server (AS), which contains OSS application logic and distributes calls to operator services Media Servers (MS) that interact with the caller and work with the AS to handle the calls. Calls may, for example, be completed to called parties, or may be transferred from one type of MS to another.

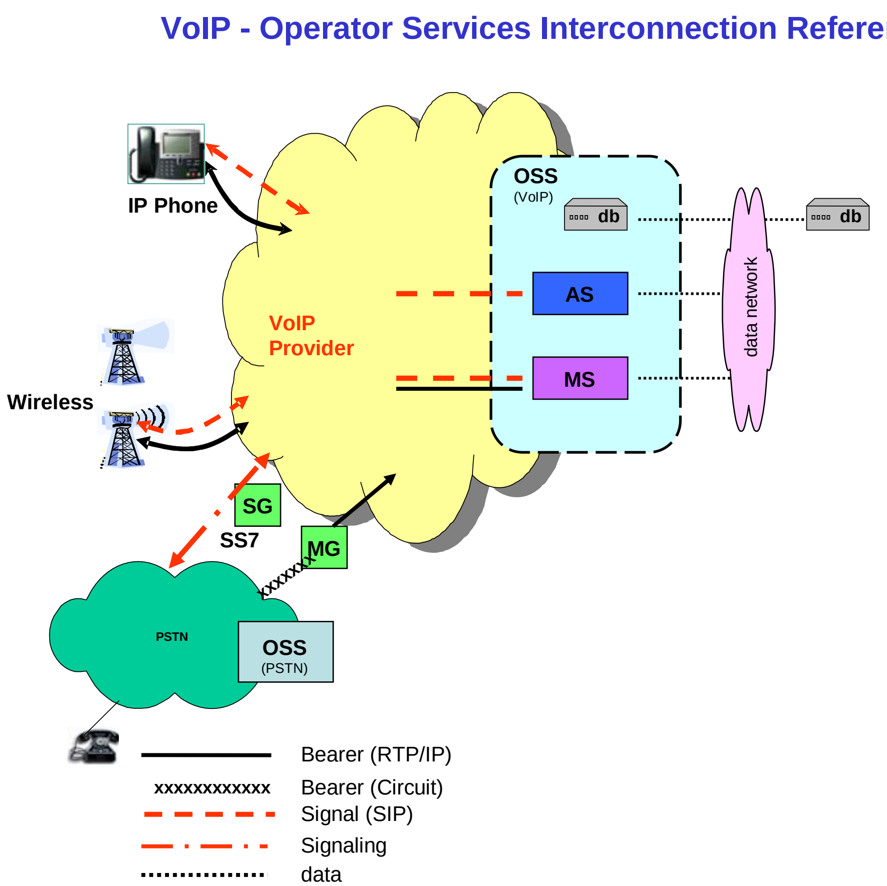

Figure 1: VoIP Services Interconnection Reference Diagram

In Figure 1:

AS = Operator services Application Server, in this case containing operator services applications
MS = Operator services Media Server, in this case supporting media interactions for operator services
OSS = Operator Services System
db = One of the variety of databases supporting Operator Services, some may be considered
to be part of the OSS, some may be considered to be external to it
MG = Media (trunking) Gateway
SG = Signaling Gateway

The OSS may have multiple of any of the AS, MS or database (db) components, i.e., any of these functions may be distributed across multiple physical elements. The OSS may be hosted by or resident within a VoIP Provider network different from the one serving the caller.

There may be multiple types of operator services MSes with different capability sets; for example, one that does simple speech recognition and another that does more sophisticated recognition; or one that specializes in domestic calls and another that specializes in international calls. Calls may be moved from one type of MS to another.

Human attendant resources (e.g., the functions currently performed by an operator position in the PSTN, involving both a workstation and a human agent) are logically considered part of an MS. An MS may include zero, one, or more pools of human attendant resources, or an MS may include a single human attendant resource. This concept is explored in more detail in sections 4.1.1 through 4.1.3.

The OSS components may be distributed or co-located. They may reside within a VoIP network, or some or all of the components may be third-party. The figure as shown assumes that the OSS relies on a VoIP network to route and distribute calls among the various MSes. However, it may also be the case that the OSS includes its own internal fabric/network for such routing.

A set of MSes could be part of a pool of media servers shared with other applications.

The figure also shows a PSTN (circuit switched) OSS, because interworking between VoIP OSSes and PSTN OSSes may be necessary (e.g. for internal call transfers between an operator and a supervisor).

Call origination and completion may occur from/to VoIP or PSTN endpoints, so the ability to interwork across a VoIP/PSTN boundary is critical.

While the data network used by the AS, MS and db to communicate is shown as being separated, it could be part of the overall IP network that includes the VoIP network.

Discussion of the dbs is out of scope of this TR. Thus, interactions with databases for real time rating, rate quotes, card validation and management, line information (e.g., LIDB) etc. are not covered here. Additionally, discussion of generic MS-MS data communication is outside the scope of this TR.

4.1 General VoIP Call Flow Involving OSS

Figure 2 below provides an illustrative call flow to a VoIP OSS.

In this description, it is assumed that the OSS is within the trust domain of a VoIP service provider’s network (see section 4.1.15 for third Party OSS considerations).

Network details not essential to understanding the role of the OSS and operator services support are omitted. SIP signaling enables all of what is described in the call flow to occur. However, SIP signaling details are not provided in this call flow. Note that all SIP signaling among the AS, MSes and Border Elements (BEs) occurs through the Call Session Control Function (CSCF).

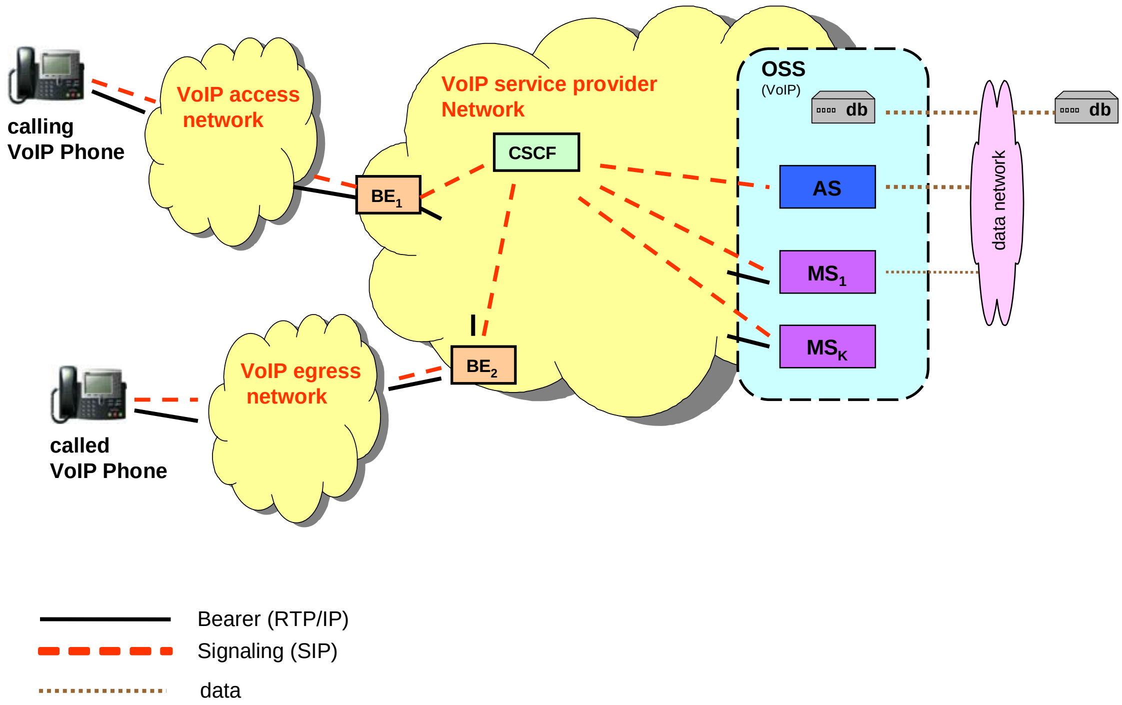

Figure 2: General Call Flow – Network Diagram

1. A calling party originates an operator services call from a VoIP phone, which sends a SIP INVITE message. SIP signaling proceeds through the VoIP access network and BE_1 to the CSCF, which determines based on the destination address that the call should be handled by an AS.
   The destination address could be, e.g., a telephony type number toll-free number in 'tel' url format, or it could be a SIP URI such as operator.svcs@mytelco.com.
   The CSCF may rely on some ancillary db or service broker function to help it determine the particular AS that is to be invoked.

2. The CSCF signals to the AS.
   Note that the following information currently signaled in the PTSN environment does *not* have analogies in VoIP signaling:
   - something that is specifically the caller's Billing Number (akin to the ISUP Charge Number)
   - originating line information (akin to ISUP OLI information)

3. The AS, acting as a SIP Back to Back User Agent (B2BUA) coordinates connecting a bearer path from the calling party to MS_1, in order to prompt the caller for additional information.

4. The AS instructs the MS_1 to perform a prompt and collect routine with the caller, e.g., a main menu through which the caller indicates what service and what language. MS_1 returns results to the AS.

5. The AS determines that a different MS, viz. MS_K, is required to further service the caller. (MS_K may, for example, have live operators that specialize in Swahili, which is what the caller requested.) The AS causes the bearer path to MS_1 to be dropped by sending a SIP BYE toward MS_1.
6. The AS causes the caller leg to be connected to MS_K by sending a SIP Re-INVITE toward the calling party and an INVITE toward MS_K.
   The Re-INVITE travels (through the CSCF) to BE_1, which handles it.
7. The AS provides instructions to MS_K regarding interactions with the caller, and MS_K returns results.
8. The AS causes the leg to MS_K to be dropped by sending a SIP BYE toward MS_K.

Assume that in this call flow the service requires call completion to a called party, e.g., this could be DA Call Completion, or card-billed, or call collect, etc.

9. The AS causes the caller leg (from BE_1) to be connected to the called party, using another SIP Re-INVITE toward the calling party and a new INVITE toward the called party.
   The CSCF routing function determines the egress BE, viz. BE_2, to be used reach the called party.
10. The INVITE reaches the called party VoIP Phone, which accepts the call.
    Upon exchange of SDP information in the SIP signaling messages, the media path is set up between the two end parties.

Note that from a signaling control point of view, the AS has remained on the call.

11. At some point the call ends. This call flow assumes that the called party clears, but the caller stays on the line. A SIP BYE is received by the AS through BE_2 for the called leg, and that call leg is cleared.[1]
12. The AS can decide what to do next, e.g., clear the calling party leg, or bring that leg back to an MS for some purpose such as giving the caller a menu of options.

4.1.1 Modeling the Workstation as a Media Server

Today's operator workstation (WS), typically including a human agent, interacts with a caller and performs application logic, e.g., in determining when to make a new call to a third party for billing verification. This section discusses how to model such a workstation within the framework of the ATIS NGN architecture as one type of Media Server. Note that a very simple prompt and collect MS, without involvement of a human agent, and a workstation-type MS, with a human agent, should be viewed as examples of MSes on an MS continuum of sophistication. Depending on the scope of service provided, a WS-type MS may be more or less sophisticated.

As such, it is recognized that

- A MS within the NGN architecture framework may contain application logic.
- Whether that application logic is carried out by a human agent or software or a combination is not relevant for NGN architecture concerns.
- How that application logic is obtained by the MS (e.g., fetched, cached or provisioned or comes in the brain of a human agent) is not relevant for NGN architecture concerns.

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[1] This example call flow does not illustrate the option that the AS could remove itself from the call path at some point after step 10 (when the call is established between the other two parties) and the end of the call.

- It is expected that any application logic in the MS does not work at cross-purposes with that in a controlling AS.

It is also recognized that extensions to the concepts of Media Resource Broker (MRB) and AS-MS control, as delineated in the NGN architecture standard [ATIS-1000018], may be needed to cover the WS case.

4.1.2 Call Distribution among MSes and among Agent Positions

In general, the VoIP architecture allows for efficient call distribution among pools of MSes. Depending on the level of sophistication desired, different methods can be employed. For example,

- An AS could provide different logical MS addresses to the Call Session Control Function (CSCF) functional element, which the CSCF would then use to route to calls to different MSes. The AS could use different logical addresses to implicitly identify different MS capability sets needed on calls.
- The routing capability within the CSCF itself could use the same MS address supplied by an AS for multiple calls to select among and route to mulitple MSes that are functionally equivalent.
- The network could employ a scheme that has knowledge of current resource utilization across the pool of MSes. For example, the AS could request assignment of an MS with particular attributes from an MRB (not shown in the figures). The role of the MRB is to know status of MS resources and support requests for MS resources. The AS would then route the call to the particular MS identified by the MRB.

Any given MS may employ local algorithms for call distribution among its internal resources, which, as noted earlier, may include pools of human attendant resources.

Attributes of a WS-type MS may include operator characteristics (e.g., a particular skill set representing the capabilities of the human operator currently using the particular workstation). The attributes of a WS-type MS may also change much more dynamically than for a typical automated MS, e.g. due to operator shift changes or breaks, or movement of a human operator from one physical workstation to another. The MRB concept acknowledges an operations support system feed to acquire knowledge of MSes and their attributes. Extension of the definition of the MRB to include other methods, such as SIP Registration or Presence techniques, should be explored as options to provide for the MRB to learn of changes in MS attributes.

Note that the MRB views sets of MS resources logically, not physically. Therefore, as a case in point, if there is a single operator/WS uniquely qualified to handle certain kinds of calls, that single ”position” could be viewed by MRB as a single MS, according to its unique set of attributes.

4.1.3 AS control of the MS

There are a variety of existing protocols/languages or protocol extensions that can be used to allow the AS to provide instructions to an MS and/or for the MS to return results, e.g., SIP “netann” [RFC 4240], VXML [VXML], or MSCML [MSCML]. (MSCML may become obsolete due to work currently under way in the IETF mediactrl working group.)

Standardization of the AS-MS control interface is an area that should be considered “work in progress”. These protocols and languages suffice for current operator services in terms of instructing the MS what to do for basic media handling and modes of interaction with the caller. However, it may be desirable to pass additional information between the AS and MS (even though this capability is not currently supported in existing protocols/languages) to support situations where:

- a more sophisticated WS-type MS with a human operator performs its functions using some application logic
- interaction between service functionality in the AS and in the MS must be supported, or
- The AS and MS are provided by a third Party network provider.

. Examples of additional information that may need to be conveyed are:

- Call type and status information (e.g., when moving a caller from MS1 to MSK, the AS has to be able to give MS_K information about the call so that it knows where to pick up in handling the call).
- Services rendered (e.g., listing number provided to the caller, or restaurant information and reservation provided for the caller).
- Allowed billing options for billing the caller and services the caller is authorized to request
- Call completion eligibility.
- An indication that a call needs to be transferred to a certain type of MS (e.g. a specialized operator position).
- An indication that a certain type of MS needs to be conferenced in.
- An indication of a recording (e.g., whisper recording of the calling party request for information or service) that needs to be passed from the recording MS to a second MS.

Depending on the extent to which the AS controls the MS, more granular information flow may be required. For example indication of specific customer requested services, target phone numbers, etc. may need to be signaled to the AS. The AS may need to signal granular information about the operator's next actions and potential choices. XML, HTTP, and other mechanisms could be investigated as a vehicle to perform this function.

Additionally, a correlation ID may need to be passed in order for the MS to know which control instructions over the control channel pertain to which SIP session.

4.1.4 DTMF/Tone Monitoring

Currently, some OS features, such as sequence calling or coin collection, require monitoring the calling party device for DTMF or other tones during call setup, or during a call, or after a call if the caller stays on the line. Thus, for example, sequence calling requires the capability for the AS to become aware that the caller has provided the expected signal (e.g., a long-duration # DTMF signal) so that the AS can connect the caller to an appropriate MS to collect the digits for the next call from the caller.

There are two basic ways to perform DTMF monitoring of the caller.

1. Keep an MS hairpinned in the media path between the calling and called parties. With this approach, the call flow above would then be different. After step 8, the AS would not drop the MS_K. Instead, it would instruct the MS_K to place the outgoing call toward the called party and instruct MS_K to monitor the caller for DTMF digits.

2. Have BE_1 perform the DTMF monitoring[2], which assumes that BE_1 is suitably capable. With this approach, in step 9 of the call flow above, when the AS re-INVITEs the caller toward the called party, the AS could also send a SIP SUBSCRIBE toward BE_1 with kpml content per [RFC 4730] requesting that DTMF monitoring be performed and notification sent to the AS when specified DTMF digits or patterns are detected. BE_1 would receive the SUBSCRIBE and act on it. The advantages of this second approach are that it affords a more direct connection between the end parties and it uses fewer network resources. Note that if BE_1 has the capability and authority, it may request that the previous network perform the monitoring, possibly eventually reaching the access network and obtaining key pad markup language (kpml) content from the access, rather than from monitoring of the RTP encoded signal.

4.1.5 Monitoring the Caller for Flash-hook

Some OS features in the PSTN environment allow the caller to perform a flash-hook on the telephone, causing a return to an OSS for interaction for additional service, such as return to a menu or operator.

A “Flash-hook” in the PSTN refers to the electrical signal that is produced when the electrical circuit of the access line is broken an immediately remade. For VoIP endpoints and access, such a technique is inappropriate because:

- VoIP does not use electrical methods for signaling information
- a VoIP access line can carry multiple calls and other traffic

For VoIP endpoints and access, it may still be desirable to preserve the ability of the caller to signal to the OSS with a request for additional service instead of, or in addition to, an existing particular call or call attempt. A potential solution could be the use of some as-yet to be standardized SIP-related signal generated by the VoIP endpoint, for an existing particular SIP dialog, to that effect. Note that KPML signaling supports a “register recall” indication. An ATIS implementation agreement would be required to use this indication for the caller to signal to the OSS with a request for additional service.

For PSTN endpoints, an OSS AS can still receive a flash-hook indication through ISUP-SIP interworking. In that case the caller’s flash-hook event is converted by the local office to an ISUP FACility message with a codepoint of “Network Service Recall” in a Service Activation parameter, per [T1.666]. That ISUP FAC message is relayed toward the VoIP network. At the ISUP/SIP interworking point, the ISUP FAC message is encapsulated in a SIP INFO message that is received by the OSS AS per [T1.679]

4.1.6 Return of the Caller to an MS

Some OS features require the caller to be temporarily returned to an MS during the call, for example to be alerted that a card balance is about to deplete or because the caller requested a return to a menu of options. To accomplish this, the AS may send a SIP Re-INVITE toward the caller, and a new INVITE toward an MS.

[2] Note that the calling terminal may support signaling of DTMF signals, e.g., using KPML (RFC 4730). If a standardized mechanism for passing this to the OSS AS in a different network were developed, this could take the place of BE monitoring for DTMF signals.

Meanwhile, the called party leg can be kept up, if desired, by sending a SIP Re-INVITE toward the called party with media description content that puts that leg on hold. If the called party leg instead were to be cleared, the AS would send a SIP BYE toward the called party.

The calling and called party legs can be rejoined by the AS sending a SIP BYE toward the MS and then sending coordinated Re-INVITES toward the calling and called parties.

4.1.7 Conferencing the Operator and Calling and Called Parties

Some operator services involve including the operator as an active participant in the call along with the calling and called parties; e.g., on a 911 call to assist the calling party, or on a Person-Person call with the ability of the caller to temporarily listen in.

Note that this functionality is different from the functionality required to support Busy Line Verification and Interrupt services described in sections 4.4 and 4.5.

To conference the three parties, the AS would utilize an MS with 3-way conferencing capabilities.

If the call is to persist without the involvement of the operator, the AS could drop the MS out of the path (a SIP BYE toward the MS) and reconnect the calling and called parties directly through SIP re-INVITEs toward each thereby causing the media connection to run directly between BE_1 and BE_2. Note: use of this procedure would cause a brief interruption in the media connectivity between the calling and called parties.

Some types of calls with the operator temporarily present, such as collect or person-to-person call, involve keeping the caller muted (i.e., maintaining bearer connections between the called party and the operator, and between the calling party and the operator, but not maintaining a bearer connection from the calling party to the called party) until the called party accepts the call. Muting can be performed through an instruction from the AS to the MS that is performing the conferencing; or, depending on the split in service logic, it could be something that an MS itself knows to perform.

4.1.8 Keeping Control of the Calling Party’s Line

Some OS features in the PSTN involve the OSS keeping control of the calling party’s line; i.e., the line is not freed up to place another call until the OSS permits it. (For example control of the calling party is retained for emergency service calls to ensure that the caller doesn’t accidentally hang up and for coin calls to ensure that entire payment for the call is collected.) Additionally, if the caller does hang up, the OSS can cause the caller’s phone to be rung (“Operator Ringback”).

In the PSTN, since an access line can only have one call, control of a line and control of a call on that line are effectively one and the same. In contrast, a VoIP access line and end device may inherently support multiple calls, and call control is performed on a per-call basis, not per access line. Moreover, SIP signaling procedures are peer-to-peer, where a SIP User Agent in the caller’s VoIP phone has as much right to clear the connection as an AS SIP User Agent in the network. It is difficult to even imagine an acceptable method in the VoIP world to parallel what is done in the PSTN.

4.1.9 Passing the Calling Party Whisper from One MS to another MS

In a situation where a caller is using speech to interact with an automated MS, there are occasions when the MS may not be able to understand the caller’s input, and the call will be transferred to a WS MS so that an operator can further assist the caller. The first (automated)

MS may have captured a segment of the caller's spoken input that must be passed to the WS MS so that it can be heard by the operator without having to prompt the caller again for the same input. That segment of speech is called a "whisper", because it is "whispered" into the ear of the operator.

Three options of passing this whisper follow:

- Option one: Convey the audio file with the caller's spoken input from the first MS to the second MS by firstly making the controlling AS aware of a URI for the audio file's location. That URI can be carried as payload in the BYE or 200 OK from the first MS (depending on whether the AS or the MS initiates clearing of that leg) to the AS using the technique described in [RFC 4483] that defines an extension to the URL MIME External-Body Access-Type to satisfy the content indirection requirements for SIP. Such an extension is aimed at allowing any MIME part in a SIP message to be referred to indirectly via a URI. The AS would then relay that URI as payload in the INVITE to the second MS. The second MS would finally retrieve the audio file from the first MS using the URI, and play the file to the operator.
- Option two: Pass a URI for the audio file between the first MS to the AS and thence from the AS to the second MS using the AS-MS control channels. This capability would need to be supported in AS-MS protocols and would require a protocol extension to do so.
- Option three: Establish a call between the two MSes, pass the whisper over the media path, and disconnect this call prior to transferring the call from the first MS to the second MS.

4.1.10 Passing the Original Called Address

In some scenarios the call may be sent to an address different from the one the caller provided.

For example a caller may call a global access address such as 800OPER8ER or joesoperatorservice@serviceprovider.com, which an intermediate system translates to another address for delivery of the call to the appropriate application.

In this section four potential approaches are listed in arbitrary order. Which one the industry adopts as a common practice remains to be seen.

1. The "final" destination address could be included in the INVITE Request-URI while keeping the original called address in the INVITE To header. This approach should suffice, since the intent of the To header is to carry the original "to" address. One potential concern is that some intermediate system may undesirably alter the address in the To header.
2. The original called address could be included in the INVITE as a Targeted-to-URI in a history-info header, by the SIP proxy or UA that changes the address per [RFC 4244]. Note that this is the most widely-available, RFC-based option in this list.
3. The original called address could be included in the INVITE as a previous Request-URI in a diversion header per [Diversion]. The Diversion draft will never be an RFC; however, it has been widely implemented and must be considered in the solution space.
4. The original called address could be carried as a URI as documented in [RFC 4458].

4.1.11 Service Support Across Multiple Networks

In the PSTN, the various parts of an operator service may be provided in separate networks. For example, an incoming call may be routed to an automated system in one network for an initial announcement and possible automation of the service. If full automation of the service is not possible for this call, the call will be routed to a live operator[3] in a separate network[4]. Multiple internetwork transfers are possible for some calls. The generic model of operator services shown in Figure 1 may be applied to these cases as shown in Figure 3. Figure 3 illustrates the basic signaling and bearer interfaces where multiple networks are involved in providing the service. For some services AS-A may act as a Back-to-Back User Agent (B2BUA) and remain in the signaling path while forwarding the call to AS-B (and its associated MSs). This is illustrated in the call flow in section 4.1.11.1. For other services, AS-A may drop out of the signaling path, e.g., using the REFER method to deliver the call to AS-B for further processing. This is illustrated in the call flow in section 4.1.11.2.

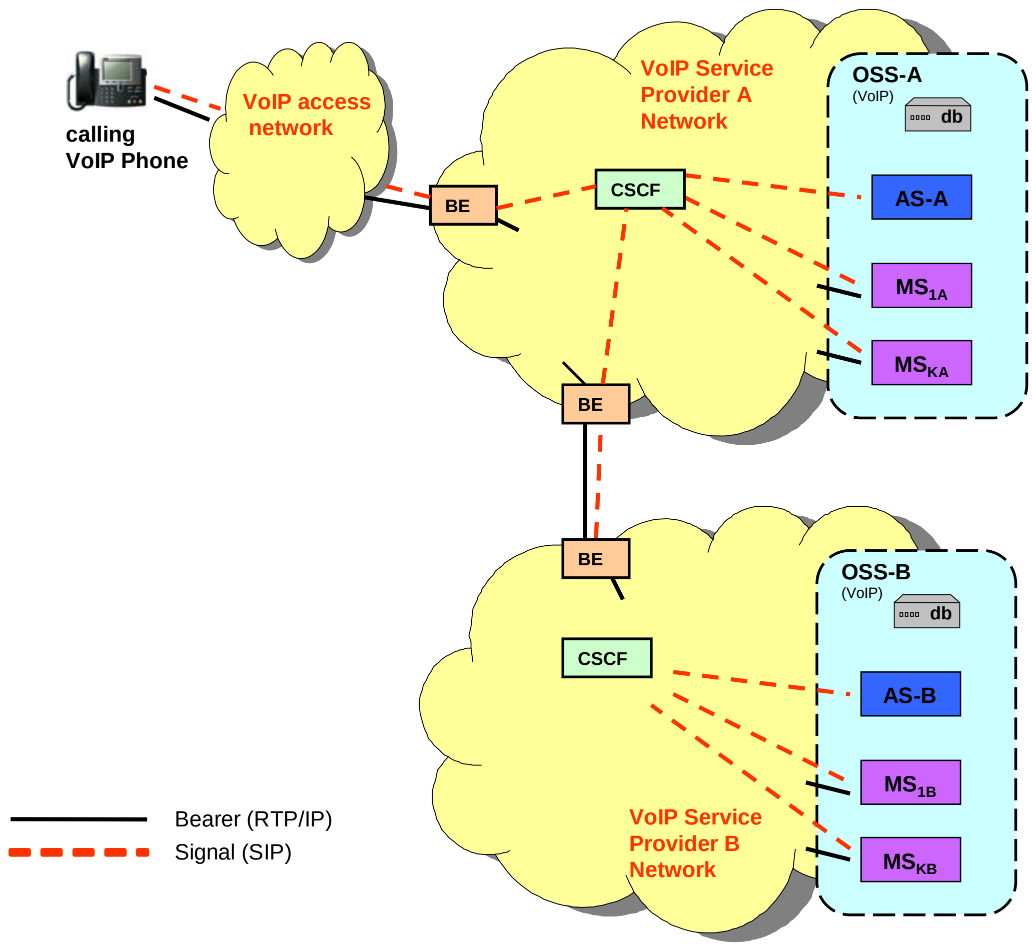

Figure 3: Operator Service Provided by Two Networks

[3] Note that, while this example shows transfer from an automated MS to an MS with a human operator, similar transfers are possible to a second automated MS.
[4] Note that it is also possible for this initial network to provide none of the actual operator service, but simply to redirect the call to the appropriate network for processing.

4.1.11.1 Internetwork Service, Service Provider Remains in the Signaling Path

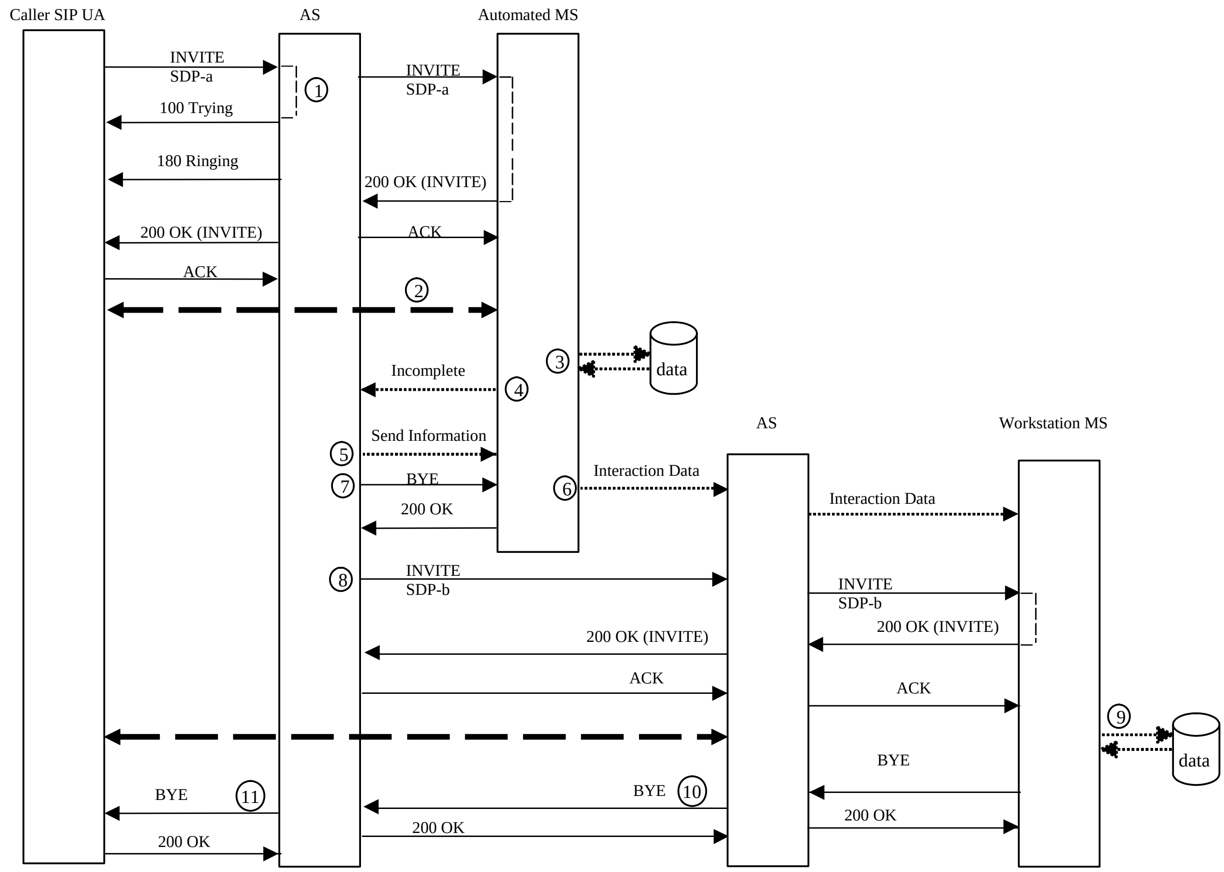

Figure 4: Internetwork Service, Service Provider Remains in the Signaling Path

Steps 1-11 of this call flow describe the signaling associated with the 12-step generic flow presented in section 4.1. In the internetwork case, steps 8 and 10 of Figure 4 involve internetwork signaling, including unambiguous identification of the Interaction Data that is sent in Step 6 and the handling of the messaging by the AS in the second network. The two involved networks will need to assess the need for additional headers or messaging to provide appropriate security and accounting functionality for the interaction.

4.1.11.2 Internetwork Service, Service Provider Does Not Remain in the Signaling Path

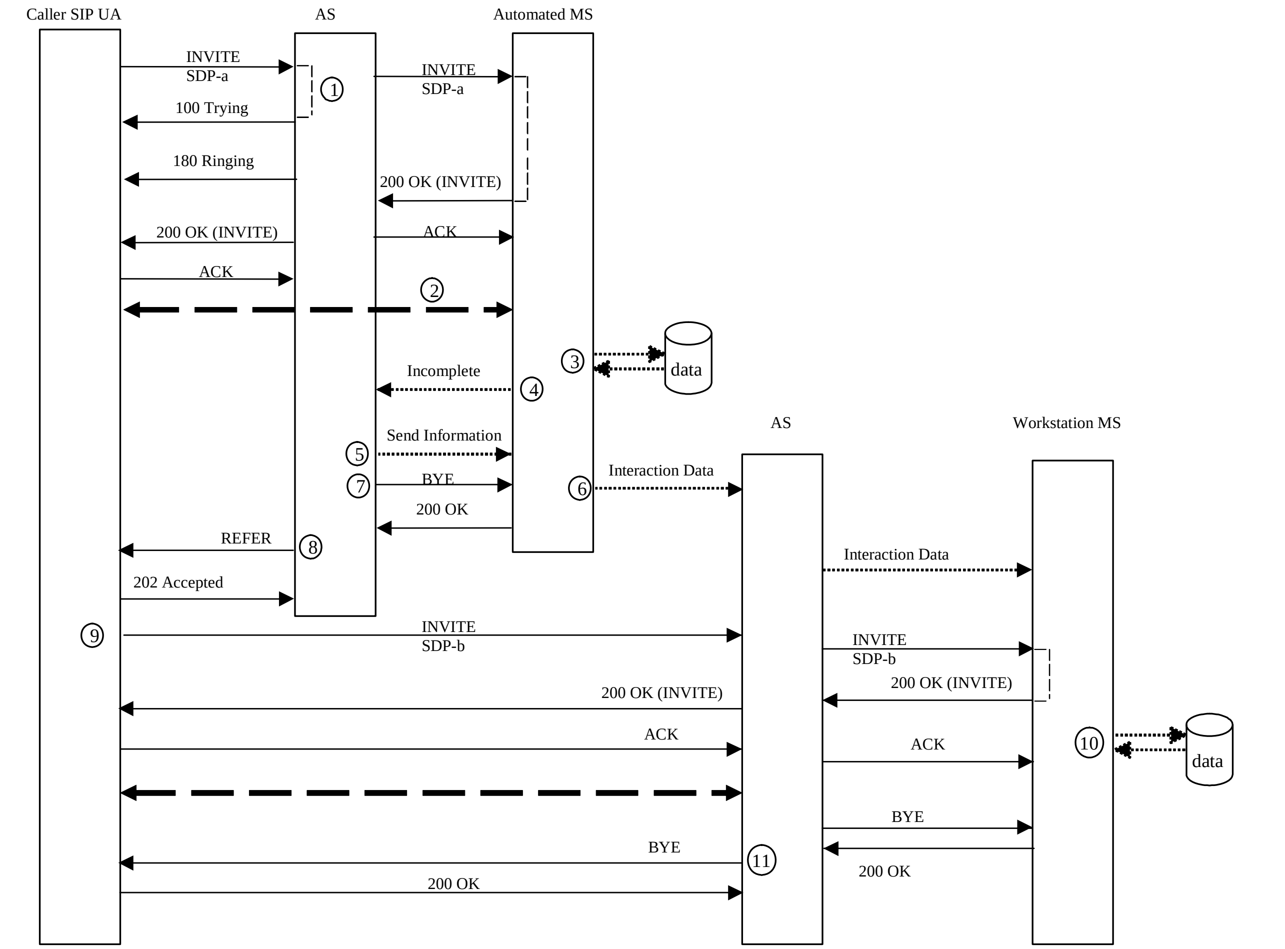

Figure 5: Internetwork Service, Service Provider Does Not Remain in the Signaling Path

Steps 1-7 of this call flow are the same as those shown in section 4.1.11.1. Note that the Interaction Data that is sent in Step 6 is sent internetwork. The two involved networks will need to assure that the call is identified unambiguously to the Workstation MS and will need to assess the need for additional headers or messaging to provide appropriate security and accounting functionality for the interaction.

At step 8, the AS in the first network uses the REFER method to redirect the call to the AS and Workstation MS in a different network. The call flow proceeds as follows:

8. The AS redirects the call to the WS MS using the REFER method which causes the AS in the first network to be removed from the call. The Caller SIP UA indicates successful completion by sending the NOTIFY (200 OK).

9. The Caller SIP UA sets up a call to the Workstation MS. Since the AS has been removed from the signaling path, security and accounting considerations are the same as for a new call to any other entity in the foreign network.
10. Based on the information from the Automated System and, if needed, further interaction with the calling party, the Workstation searches its associated database(s) (which may or may not be the same database(s) searched by the Automated System). In this illustrative example, the service is completed as the operator at the Workstation MS verbally provides appropriate information to the calling party.
11. The AS in the second network (and, therefore, the Workstation MS) disconnects from the SIP UA, by sending the 200 OK. Note that at this point the original OSS (the AS in the first network) is not in the signaling path and is not in a position to offer a call completion service. Depending on the details of the call setup signaling to the Workstation MS, the AS in the second network may or may not be in a position to offer a call completion service.

4.1.12 Addressing and Routing

Network-internal address translation and routing suffices for operator services. There is no need to use network-external address translation (e.g. ENUM) or routing functions for operator services because either the caller's service provider network is providing the services internally, or that network/provider has a contractual relationship with an external operator services provider. In that latter case, since the external provider is known, addressing and routing information can be provisioned into the caller's service provider network to send signaling/calls toward the external provider.

4.1.12.1 Addressing

Table 1 below illustrates a representative set of SIP URI formats for telephone numbers for Operator Services calls. These examples are taken from [ATIS-1000009].

Table 1 - SIP URI Formats for Telephone Numbers

URI	sip:+1NPANXXXXXX@host;user=phone
Description	NANP number
URI	sip:+18YYXXXXXXX@host;user=phone
Description	NANP 8YY number
URI	sip:+1NPA5551212@host;user=phone
Description	NANP Directory Assistance in global number format
URI	sip:0;phone-context=+1@host;user=phone
Description	NANP operator requested in local number format
URI	sip:0NPANXXXXXX;phone-context=+1@host;user=phone

Description	NANP operator requested in local number format
URI	sip:00;phone-context=+1@host;user=phone
Description	NANP LD operator requested in local number format
URI	sip:411;phone-context=+1@host;user=phone
Description	NANP special service code in local number format
URI	sip:613131;phone-context=+1@host;user=phone
Description	NANP directory assistance in local number format; illustration is for a call to Canadian NPA 613 and using Special Service Code 131
URI	sip:B;phone-context=+33@host;user=phone
Description	NWZ1 directory assistance in local number format; illustration is for a call to France (Country Code 33) for Directory Assistance (Code 11 (hex B)); whereas Code 12 (hex C) would be used for International Operator Services (non DA). This is based on the current utilization of hexadecimal digits in the Called Party number in ITU-T SS7 (Q.763).

SIP addresses that don’t embody traditional numbers, such as AcmeOperatorServices@host, could also be used, although obviously only from VoIP endpoints.

4.1.12.2 Routing

At the SIP signaling level, Operator Services routing may be to:

- An AS that is network-internal
- A third Party AS by way of Application Server Gateway (SIP third Party AS) or an Open Service Access Service Capability Server (Parlay Third Party AS)
- A MS (network-internal or external)
- An end-user
- Another network

and therefore can utilize the same routing mechanisms as other kinds of VoIP calls that may involve those types of entities.

> Note that it is assumed that neither an AS nor an MS in one network will directly route to an MS in another network.

At the IP network layer (IP addresses) there is no distinction between Operator Services signaling or media traffic and other kinds of VoIP traffic. Existing mechanisms are sufficient for routing at that level.

4.1.13 PSTN-SIP Interworking

As depicted in Figure 1, a VoIP OSS may receive calls from or complete calls to PTSN-based endpoints or an operator system within a PTSN. This section describes the basic PTSN-SIP signaling interworking framework. Other sections on particular operator services features discuss PTSN interworking for specific information as needed for those features.

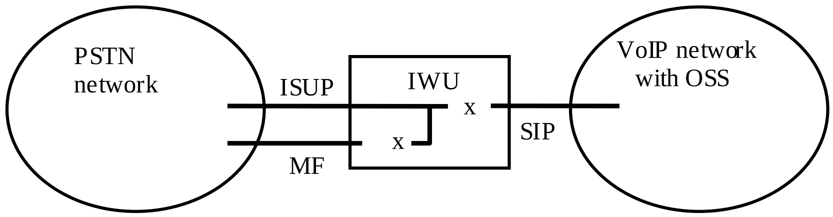

Figure 6: PSTN-SIP Signaling Interworking

Figure 6 illustrates that the point of signaling interworking is, logically, an Interworking Unit (IWU). IWU functionality may be implemented, for example, within a Gateway at the edge of the VoIP Network. The IWU interworks between ISUP (out of band) or MF (inband) signaling on the PSTN side and SIP on the VoIP network side.

4.1.13.1 ISUP-SIP Interworking

ISUP-SIP interworking is specified in [T1.679]. There are four key points to note:

5. Some ISUP information gets directly populated into SIP headers and fields; for example the ISUP IAM Called Party Number is populated into the SIP INVITE Req URI.
6. Since not all ISUP information can be populated directly into SIP headers and fields, entire ISUP messages may be conveyed as payload in SIP messages; for example an ISUP IAM may be carried as payload in a SIP INVITE.
7. Not all Operator Services-related ISUP message content is addressed in T1.679. See the discussion on T1.666 below.
8. When conveying the entire ISUP message is not necessary, specific ISUP parameters whose information content cannot be directly populated into SIP headers and fields may be conveyed using the Narrowband Signaling Syntax [ATIS-1000008].

With respect to point 4, the information content could be conveyed via as yet-to-be-proposed SIP headers or could be conveyed via as yet-to-be-proposed new parameters[5] in the Req URI. For example, the information content of the ISUP Operator Services Information parameter, which consists of the two fields Information Type and Information Value, could be conveyed as two new URI parameters. Where the information content may have broader application (e.g., the ISUP Charge Number), service-specific replication by standardizing new URI parameters may be an unattractive option for carrying the information content.

To provide similar operator services for calls initiated in the PSTN or in the VoIP network, a VoIP OSS that receives a SIP INVITE with information content in an encapsulated ISUP message, an NSS body, or operator services parameters in the Req URI should use the corresponding mechanism as needed in its reply messaging.

The VoIP OSS AS may also need to convey operator services-specific information toward a called endpoint, such as when transferring a call to another OSS. Some of that information may need to be carried along with the initial call setup (e.g., “Intercept” indication) but can’t be

[5] Recall that the syntax of a ‘parameter’ differs between SIP and ISUP, although in either case the purpose of the parameter is to convey information.

carried directly in SIP so must be conveyed in the SIP INVITE as encapsulated ISUP, NSS payload, or as new URI parameters. The VoIP OSS AS may not know a priori whether the destination is PSTN based, or, even if it does, what the SIP-ISUP signaling interworking capabilities of the egress gateway are. The AS would have to include appropriate SIP content to cover the set of unknowns.

T1.666 specifies ISUP signaling for a variety of Operator Services capabilities including coin station control, transfers, and connection control. The ISUP messages used for such capabilities can all be conveyed as SIP payload according to [T1.679]. However, not all of the parameters of those messages are available in T1.679. Table 2 below lists the parameters that may be used according to T1.666 and indicates which are listed in T1.679, which require the NSS extensions documented in ITU-T Recommendation Q.1980.1, and which require the further extensions documented in ATIS-1000008.)

Table 2 - Availability of T1.666 ISUP Parameters

T1.666 ISUP Parameter	ISUP Message	T1.679	Q.1980.1	ATIS-10000008
Called Party Number	IAM	X		
Carrier Identification	IAM			X
Carrier Selection	IAM		X	
Charge Number	IAM			X
Generic Address	IAM	X		
Generic Digits	IAM		X	
Operator Services Information	IAM			X
Optional Backward Call Indicators	ACM, ANS, CPG			X
User-Network Interaction Indicator	ACM		X	
Originating Line Information	IAM			X
Service Activation	IAM, ACM, Facility		X	

Note that the information content of any of those information elements could also be potentially passed by (or to, as appropriate) a SIP originating endpoint by encapsulating the entire ISUP message or by as-yet-undefined URI parameters.

4.1.14 MF-SIP Interworking

For MF signaling, as indicated in Figure 6, it is assumed that the IWU effectively internally uses ISUP to mediate between SIP and MF.

Support for MF is important on a practical basis especially for the case of coin phone-related signaling. Most coin phone-related signaling today is MF. T1.666 includes ISUP signaling for network control of coin phones. For a PSTN network-controlled coin phone, a VoIP OSS AS can use Narrowband Signaling Syntax encoded ISUP coin-related signals that cannot be encoded in SIP messages, and the IWU will interwork that ISUP coin control content to MF signals and inband tones; and vice-versa.

4.1.15 Third-Party Service Provider Considerations

In the PSTN environment today, operator services, including DA, are often provided by third party systems, and there is no reason to expect that to change in the VoIP environment.

Two characteristics of third party service support in the current PSTN environment are worth noting:

9. The third party systems are monolithic, i.e. there is no AS/MS-like split.
10. Providing operator services may actually be a cooperative effort between the third party system and service-layer functionality in the network. For example, the following kinds of information may be exchanged: identity of the caller's host network, eligibility for call completion (for DA), and recording information such as the listing number provided.

In the VoIP environment, the two most natural candidate interfaces for third party service support are:

- SIP, and
- Parlay or Parlay X

SIP and Parlay represent different approaches and even philosophies to a third party service interface. SIP is primarily a powerful call session control protocol, and the services model assumes the existence of separate Media Servers that the SIP AS sends call legs to and controls (via some control mechanism) in order to interact with a caller or to support a conference. The Parlay model assumes that the network has various resources that can be invoked – more so than SIP; some for call control, some for charging or account or profile management, some for interactions with a user, some for conferencing, etc. With Parlay, Media Server functions are assumed to be resources embedded in the host network. Note that one could in principle broaden the view of the Parlay model to include third party Media Servers external to the network and controlled by the Parlay Application Server via some kind of control mechanism, akin to the SIP third party service model.

4.1.15.1 SIP Third Party Interface

Consider the situation where the OSS is composed of a Third Party AS and one or more Third Party MSes, all of which are then outside the host network(s). Note that the case of a monolithic OSS – a combined AS & MS - can be considered a simplistic or degenerative case of that more general situation.

It is assumed that the AS-MS control communication does not involve the host network(s), although the host network(s) may be involved in setting up the control path or transporting control instructions. See section 4.1.

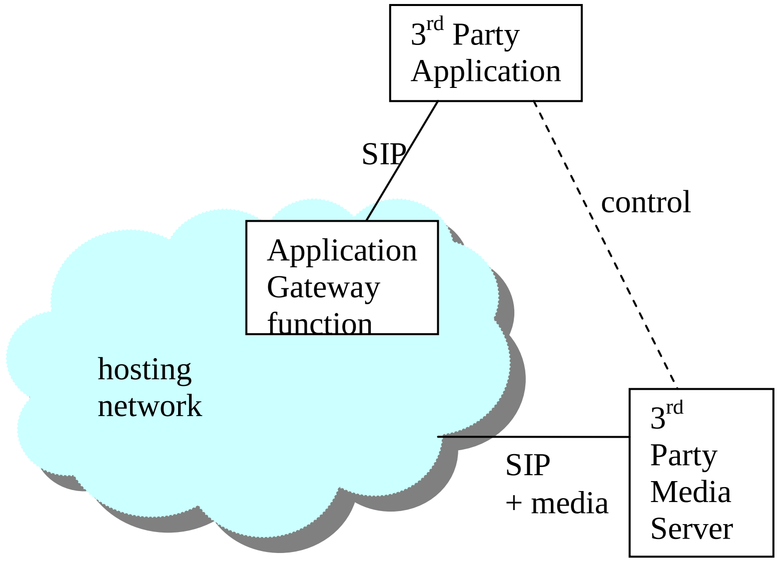

Figure 7: SIP Third Party Application Relationship to Host Network

A SIP third party interface could afford the third party system any and all of the capabilities that SIP supports, i.e., anything a network-internal AS could do. In practice, the host network would internally implement functionality, SIP Application Gateway functionality, that would restrict the SIP interface to support for services and capabilities as specified in a Service Level Agreement, and provide border security functions.

For the network-AS interface, the gaps in SIP support for operator services would be the same as for a SIP AS inside the network, which have been identified in previous sections and are summarized in section 7.

The network-MS interface could appear as a regular User-Network Interface, assuming that the AS and MS can pass control instructions and information over a separate control channel, with the following three additions:

11. Support of a “correlation ID”: This is to be, conveyed to the MS from the AS in a SIP INVITE and will be used to associate control instructions for the call that arrive over a separate AS-MS control path for that call.
12. Support of RFC 4240: This provides for support of simple MS-type instructions in a SIP INVITE from an AS.
13. Support for the establishment of and transport for an AS-MS control channel, e.g., per, [Mediactrl].

4.1.15.2 Parlay Third Party Interface

The set of Parlay APIs (available from the Parlay Group web site at http://www.parlay.org/en/index.asp) was designed to be a third party service provider interface, sophisticated enough to support a broad set of application-related capabilities without the extra complexity that would come with other protocols such as SIP or MEGACO.

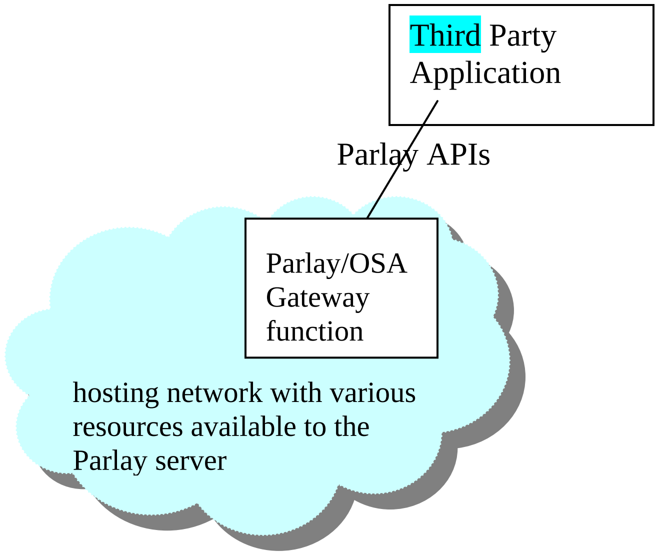

Figure 8: Parlay Third Party Application Relationship to Host Network

Architecturally, the Parlay application interfaces across the API with a Parlay/OSA (Application) Gateway function that resides in the host network. As mentioned above, that model basically assumes that resources for interaction with the caller (announcements, prompt and collect) are provided by the network. If third party Media Servers were used, one could extend the model so that either (1) the Parlay application server sends calls to them and controls them, or (2) the Parlay Gateway function to send calls to them and controls them.

In addition to the Parlay set of APIs, a set of web services referred to as Parlay X have been more recently defined. The service capability set of Parlay X is essentially a limited subset of the capabilities provided by the Parlay APIs. This Technical Report focuses on Parlay, for which the gaps to support operator services would be fewer, rather than on Parlay X.

The Parlay specifications that would primarily be used to support operator services include:

- Multi-Party Call Control and User Interaction,
- Charging
- Account Management (if user information is stored in the network), and possibly
- Mobility (for wireless).

Multi-Party Call Control supports call leg manipulation, including simple conferencing, and DTMF monitoring. User Interaction supports prompt and collect interactions with a call leg. Charging allows certain kinds of recording-related information to be passed to the network.

Regarding the network-AS Parlay interface and operator services needs, Parlay does not currently support the following, non-comprehensive list of functions:

1. Passing the following information to the Parlay application that would have come to the network, e.g., via ISUP on call origination:
   - Originating Line Information (e.g., residential/prison/hospital/coin)

- Carrier Identification Code (of the network serving the caller)
- Charge/billing number

2. Passing information back to the network for recording purposes
   - Listing number (provided by directory assistance)
   - More generally, a field for arbitrary information - network doesn't need to know what it is,
     just place it in the recording flow

3. Muting the caller leg temporarily (e.g., during the call setup phase on a collect call when the caller can hear but not speak while the operator announces the call to the called party
4. Passing information to the Parlay application on SIP originated calls – note that the following are not even currently supported in SIP
   - Caller's service providing network information
5. Conveying information on call type and status from the Parlay application to a network media server resource (e.g., when effecting an "operator transfer" so that the second operator position function knows where and how to pick up on interacting with the caller)

4.1.16 Information Elements Required

Support of the information-requested services requires that the OSS receive an indication of the providers (HSPs and Aggregation Services Providers) involved in an incoming call to the OSS. This information is used for branding, billing, routing, and additional call processing functions. DA providers use other information associated with the calling end user's equipment to influence call processing. Specifically, the DA provider may need the following additional signaling information:

- Provider Identifiers
- Originating Station (VoIP Calling Endpoint) Type
- Network Type Identifier
- Codec
- Dialed Digits

This section discusses providing the required information in an NGN.

4.1.16.1 Potential Provider Identifiers

The following are several possible ways to identify the providers (HSPs, Aggregation Services Providers) which are involved in a call. All the options are not applicable to all types of providers.

14. A unique identity for each provider, such as the National Exchange Carrier Association (NECA) Operating Company Number (OCN), which comprises 4 alphanumeric characters. The OCN may be delivered to the DA provider in the SIP INVITE message, if a standardized field is defined in the SIP message to carry it (for example: OCN: HSP1).

15. The right hand side of the values in "Via" headers inserted by the providers. The first Via would presumably be inserted by the serving HSP, while additional Via headers are inserted by Aggregation Services Providers. E.g. (irrelevant headers omitted):

```
INVITE sip:411@HSP1.net SIP/2.0
From: Alice <sip:alice@HSP1.net>;tag=1234567
To: sip:411@HSP1.net
```

Via: proxy1@HSP1.net
Via: proxy2@ASP1.net

16. The right hand side of the values in the “P-Asserted-Identity” headers inserted by the originating provider.
17. If the calling party's DN is signaled to the DA provider, then the HSP identity could be derived from the calling DN. Since this option requires HSPs to provide their end users' DNs to the DA providers, the use of this option depends on HSP business arrangements. This option is not expected to be used by all HSPs.
18. Implicit knowledge of the provider when it is served via a dedicated connection.

4.1.16.2 VoIP Calling Endpoint Type Identification

Operator services treatment can be tailored based on the type of originating station. For instance, calls from prison phones are typically restricted from accessing DA services. Example values include POTS, coin, hospital, prison/ inmate, and cellular. This information is signaled for SS7 calls using the Originating Line Information (OLI) parameter, and in MF calls using the ANI II digits. Additional information about the nature of the calling party, such as whether it is an operator or is requesting use of a specific language may be carried in the ISUP Calling Party’s Category (CPC) parameter.

OLI-type information could be stored in a VoIP network as companion information to the station address information used for authentication (see section 4.1.13.3 – “VoIP Calling Station Identity”). Such information could be provisioned into the database as part of service activation for a customer, or could be populated as part of registration/configuration. The OLI type information could be conveyed in SIP signaling as an oli parameter of the address URI, most appropriately in the P-Asserted Identity header. Likewise, calling party category information could be conveyed using a cpc parameter. Depending on trust considerations, an alternative approach could be to convey the OLI and Calling Party’s Category information using a Security Assertion Markup Language-based scheme per [SAML]

4.1.16.3 VoIP Calling Station Identity

Operator services rely on trustworthy identification of the calling station for screening or recording purposes.

In the VoIP environment, due to endpoint device mobility or multiple endpoint devices sharing an access line, it is the endpoint device (Terminal Adaptor, SIP Phone, etc.) that must be identified instead of the access line that the device is currently using or sharing.

In order to validate the identity of a calling endpoint device, networks use a challenge/authentication procedure on every originated call, where the calling endpoint supplies acceptable credentials along with its claimed identity (SIP or tel URI address). The SIP Digest Authentication scheme is one such standard procedure [RFC 3261]. In order to authenticate, the VoIP network must maintain a database of served legitimate endpoint addresses and their credentials.

The authenticated address could be passed in SIP signaling among SIP UAs within the same network (e.g., CSCF to AS) or between networks (e.g., when the host network is not the one providing the operator services). The authenticated address could be passed in the P-Asserted Identity header.

4.1.16.4 Network Type Identifier

Today, operator services providers perform call processing functions based on the type of originating network. For example, a unique operator queue can be selected if the call is wireless. In addition, the Automation recognition rates can be improved with the knowledge of the type of network. One way operator services providers can benefit from receiving an indicator that the type of network is VoIP is by setting up operator teams that are especially suited to handle calls from VoIP providers.

Ways to represent this information in SIP need to be explored.

4.1.16.5 Codec

By knowing the codec, the success rate for automated speech recognition can be improved.

This is currently signaled in the SDP.

4.1.16.6 Dialed Digits

For DA, the dialed digits continue to be important to differentiate 411 calls from 555 calls and direct dialed calls from 0+ dialed calls.

This information is carried in the initial INVITE, but due to retargeting or other reasons, it may be modified. It will be important to maintain this information along the entire path.

There may be existing mechanisms in SIP to handle this. Further study is needed. See section 4.1.10.

4.2 Service Call Flows

This clause provides basic call flows for an NGN-originated call to an NGN Operator Services System.

Note that the signaling flows in this section do not include all SIP messaging. For example, when transferring a call leg from one MS to another, it will be necessary to send a Re-Invite to temporarily put the caller on hold, then another Re-Invite to pass the new SDP info for the new MS.

4.2.1 Directory Assistance

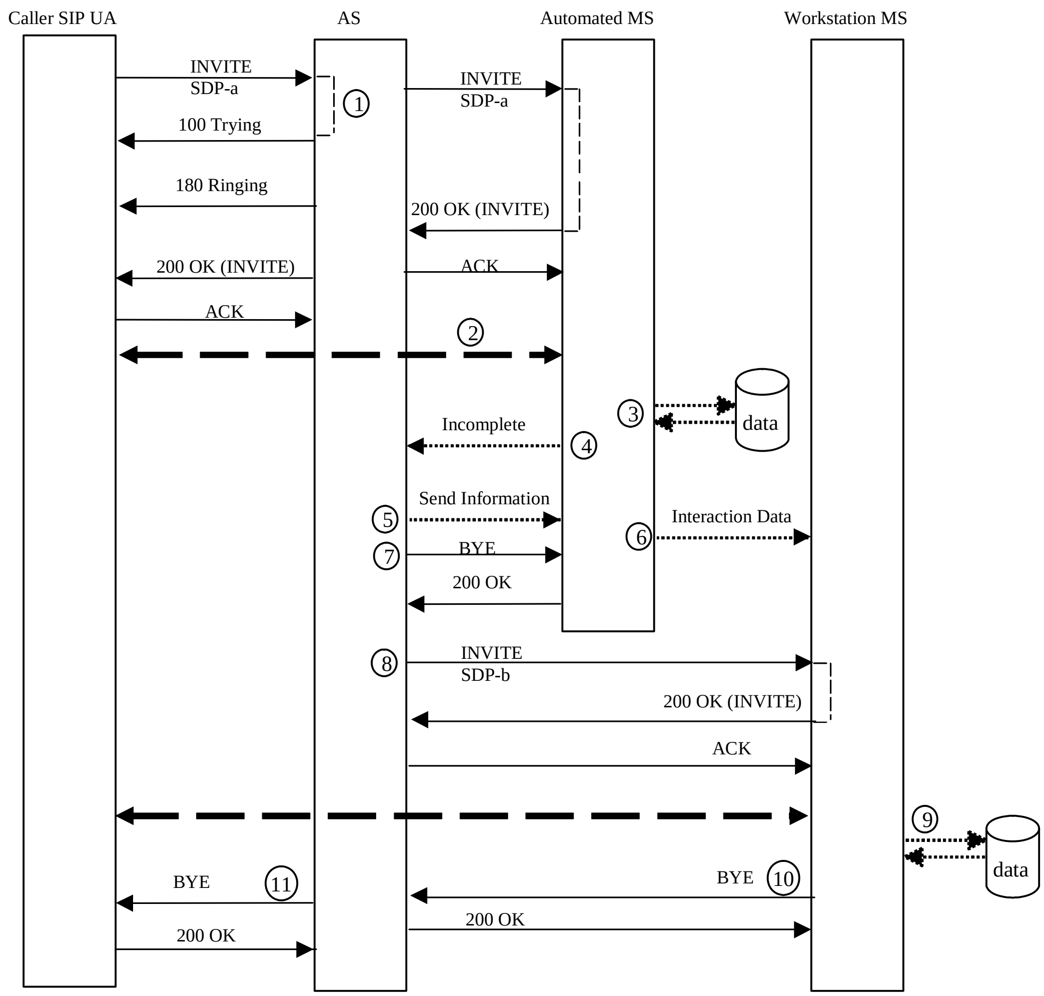

Figure 9: Directory Assistance Call Flow

Directory Assistance at the OSS involves the following steps:

1. A SIP INVITE for an incoming call arrives at the AS. The AS causes the call to be connected to the automated MS for initial processing.
2. The AS provides instructions to that MS on what to do, which may be carried in the SIP INVITE or over a separate control channel (not shown). In this illustrative example, the caller is prompted for information. The calling party provides the Automated System with the requested name and location for the Directory Assistance request.
3. The automated MS searches its associated database(s) for the requested listing. In this illustrative example, the Automated System is unable to unambiguously determine which listing is desired.
4. The automated MS notifies the AS that the listing has not been determined.

5. The AS or other network element selects an operator Workstation MS to continue call handling and requests the automated MS to forward relevant information (e.g., an audio clip of the request as given by the calling party) to the appropriate Workstation MS.
6. In response to the request from the AS in step 5, the automated MSforwards information to the Workstation MS. That information may be relayed through the AS using the MS-MS control mechanism or through an MS-MS data network.
7. The AS causes the call to the automated MS to be terminated.
8. In parallel with step 7, the AS connects the caller to the Workstation MS.
9. Based on the information from the automated MS and, if needed, further interaction with the calling party, the Workstation searches its associated database(s) (which may or may not be the same database(s) searched by the automated MS). In this illustrative example, the listing is unambiguously determined. The operator at the Workstation MS verbally provides the listing information to the calling party. The Workstation MS may also provide the listing information to the AS.
10. The Workstation MS drops from the call. Note that at this point the AS could offer a call completion service if it has the listing information.
11. At the completion of the Directory Assistance service, the AS ends the call.

4.2.1.1 Internetwork Directory Assistance, Service Provider Remains in the Signaling Path

In this example call flow, the Automated MS is in the service provider network, with AS-a, and the operator Workstation MS is in a different network, with AS-w. The service provider remains in the signaling path for the duration of the service.

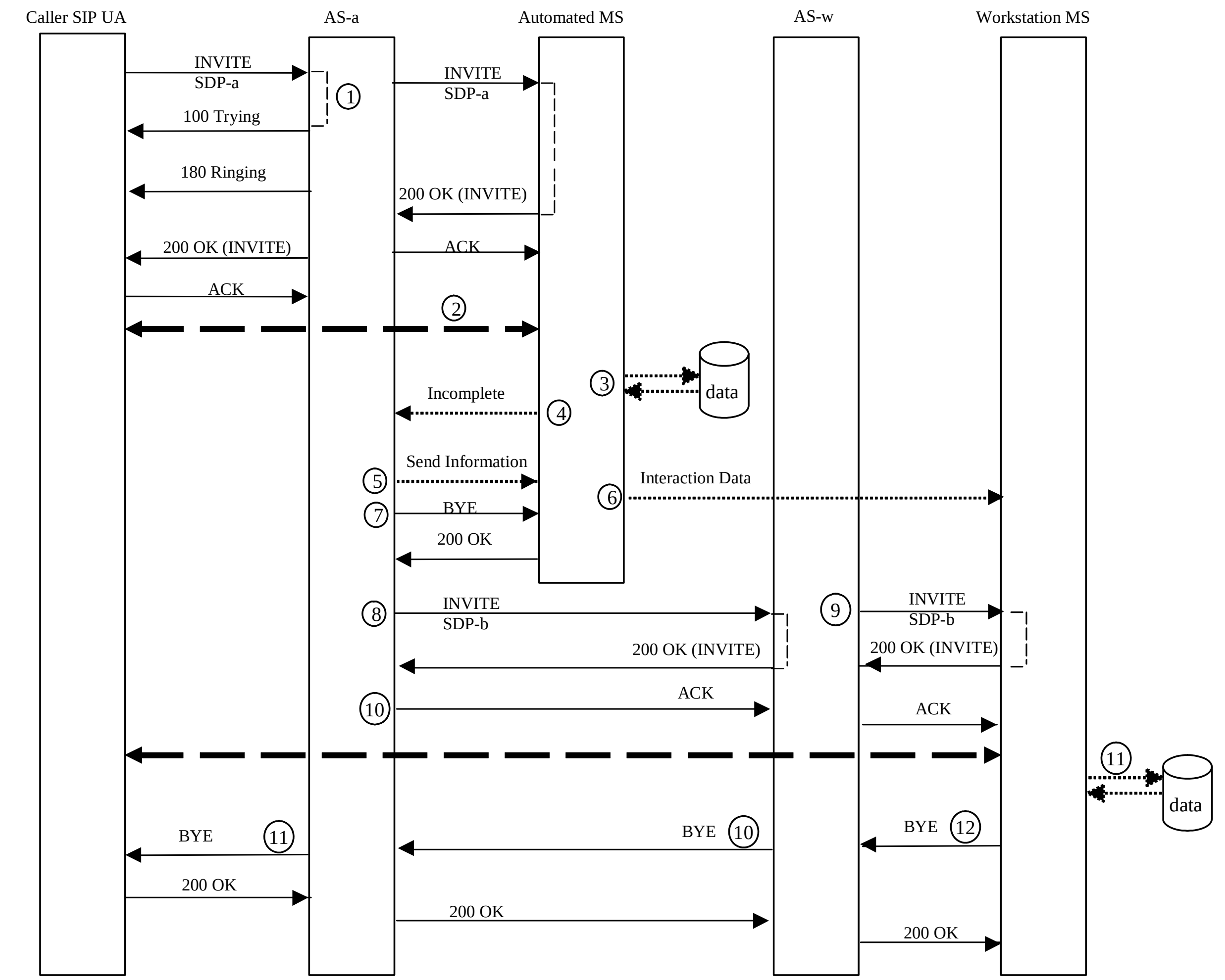

Figure 10: Internetwork Service, Service Provider Remains in the Signaling Path

Steps 1-8 of this call flow are the same as those shown in section 4.2.1. At step 9, AS-w sends the INVITE to the Workstation MS.

Steps 10, 11, and 12 correspond to steps 9, 10, and 11 in section 4.1, respectively.

Steps 6, 8, and 10 involve internetwork signaling. The two involved networks will need to provide appropriate security and accounting for the interaction.

4.2.1.2 Internetwork Directory Assistance, Service Provider Does Not Remain in the Signaling Path

In this example call flow, the Automated MS is in the service provider network, with AS-a, and the operator Workstation MS is in a different network, with AS-w. The service provider delivers the call to the second network and does not remain in the signaling path for the duration of the service.

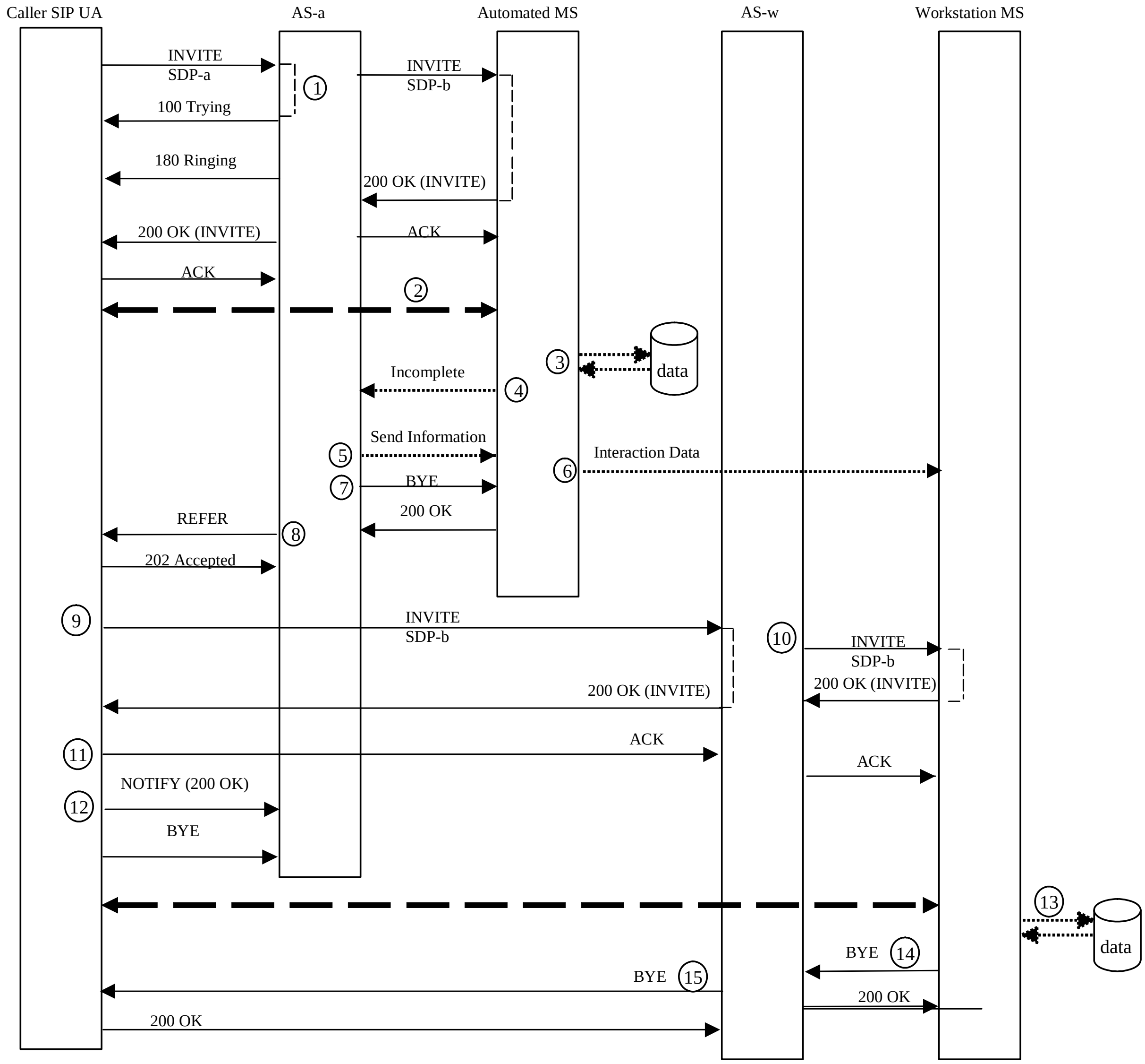

Figure 11: Internetwork Service, Service Provider Does Not Remain in the Signaling Path

Steps 1-7 of this call flow are the same as those shown in section 4.1. Note that the Interaction Data that is sent in Step 6 is sent internetwork. The two involved networks will need to assure that the call is identified unambiguously to the Workstation MS and will need to assess the need for additional headers or messaging to provide appropriate security and accounting for the interaction.

Starting at step 8, AS-a uses the REFER method to redirect the call to the Workstation MS in a different network. The call flow proceeds as follows:

8. AS-a redirects the call to the Workstation MS using the REFER method. The SIP UA indicates successful completion by sending the NOTIFY (200 OK).
9. The SIP UA sets up a call to the Workstation. Since the OSS UA has been removed from the signaling path, security and accounting considerations are the same as for a new call from the SIP UA to any other UA in the foreign network.

10. Based on the information from the Automated System and, if needed, further interaction with the calling party, the Workstation searches its associated database(s) (which may or may not be the same database(s) searched by the Automated System). In this illustrative example, the service is completed as the operator at the Workstation verbally provides appropriate information to the calling party.

The Workstation disconnects from the SIP UA. Note that at this point the OSS is not in the signaling path and is not in a position to offer a call completion service.

4.2.2 Operator Service Including a Call to a Third Party

4.2.2.1 Call Completion to a Called Party

Some operator services involve connecting the calling party to a called party. The call flow in Figure 12 illustrates this in the form of a Directory Assistance Call Completion call.

Steps 1 and 2 of this call flow are the same as those shown in section 4.1.

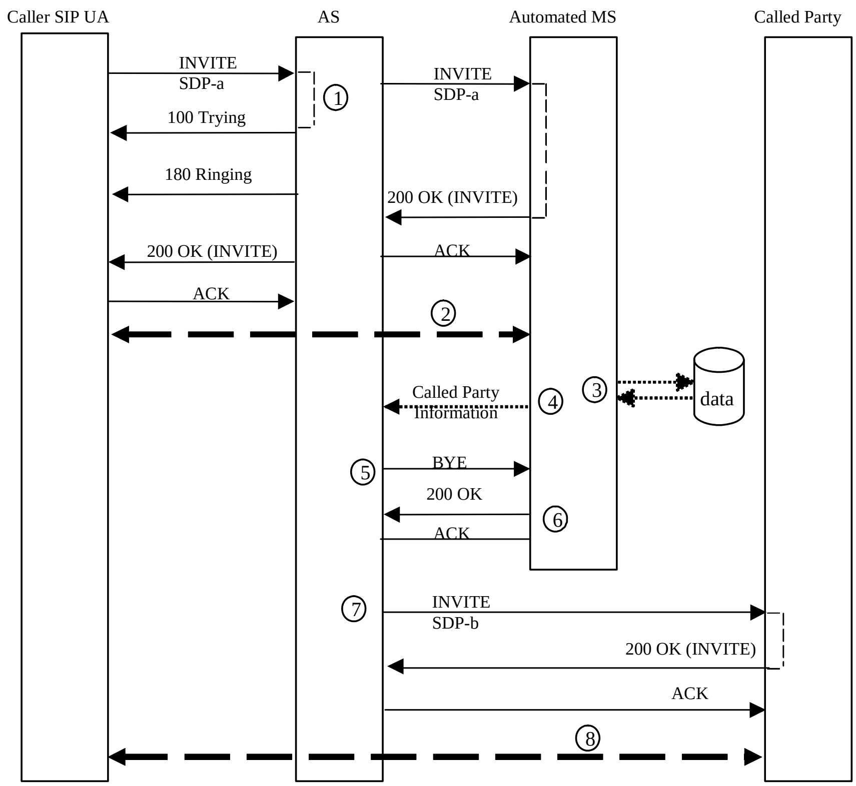

Figure 12: OSS Call to a Called Party

Figure10 illustrates call completion to an end party once the call has been removed from a Media Server. It represents various scenarios such as call completion to the provided number on a directory services call or call completion to the destination user on a card-billed call or a collect call. The involvement of one MS is shown, although without loss of generality it could be more than one, as depicted in Figure 6.

Steps 1-6 show call setup to the MS, successful identification of the called party, and eventual release of the MS, corresponding to steps 1-7 in Figure 6. Steps 7 and 8 show connection to the destination party, which is similar to connecting to a second MS in step 8 in Figure 6.

4.2.2.1.1 Operator Service Call to a Third Party

Some operator services require that the operator place a call to a third party. The example call flow shown in Figure 13 illustrates this in the form of a call billed to a third party. This illustration assumes that the MS initiates and then releases the call to the billed party; an alternative illustration could have shown that being coordinated by the AS.

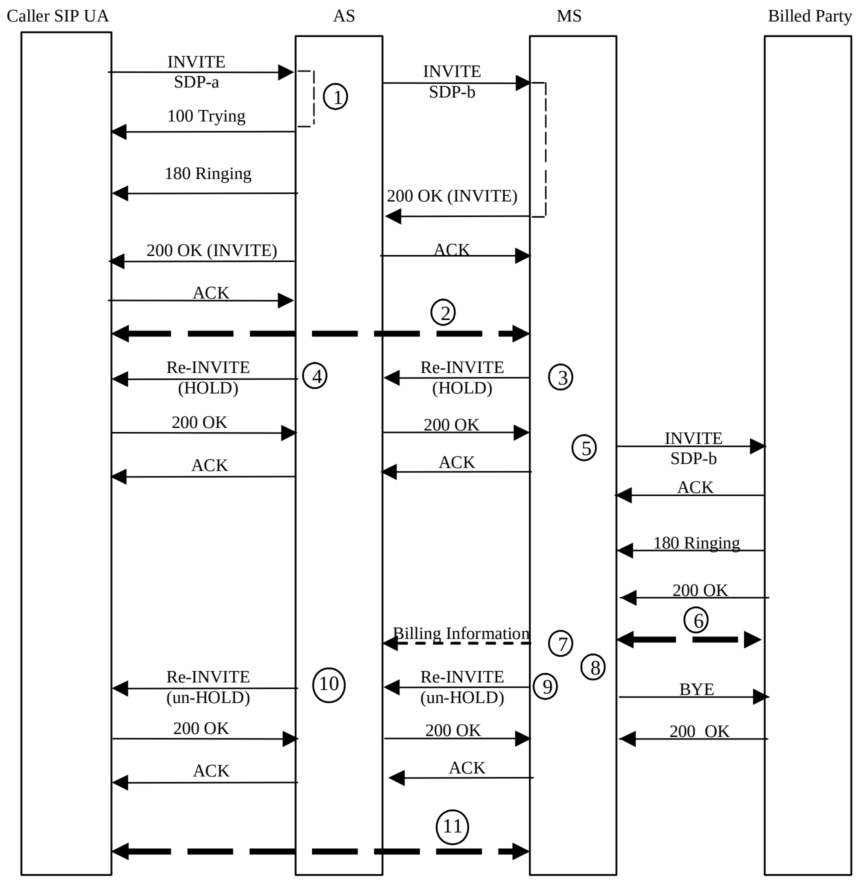

Figure 13: Bill to Third Party Call

Steps 1and 2 of this call flow are the same as those shown in section 4.1. The call flow then continues as follows:

3. Based on interaction with the Calling Party, the MS determines that a third party is required to authorize the call to the Called Party (not shown in the Figure.) Therefore, the MS sends a re-INVITE to place the Calling Party on hold.
4. The AS relays the re-INVITE to the Calling Party.
5. The MS sends an INVITE to the Billed Party to set up the call.
6. The MS interacts with the Billed Party and determines that the Calling Party should be allowed to make the desired call to the Called Party.
7. The MS informs the AS that billing has been satisfied.

8. The MS sends a BYE to the Billed Party to disconnect.
9. The MS sends a re-INVITE toward the Calling Party to reestablish the conversation between the MS and the Calling Party.
10. Based on the information from the MS that billing has been satisfied, the AS relays the re-INVITE to the Calling Party.
11. The MS announces to the Calling Party that the call to the Called Party has been authorized.
12. At this point, the call flow continues to call completion as shown starting at step 9 in Figure 2 in section 4.1.

4.3 Coin Telephones/Service

4.3.1 For VoIP Coin Telephones

This subsection assumes that a VoIP path exists between a VoIP coin telephone and the OSS. It is also assumed that the basic coin-related communication content that is used today in the PSTN environment suffices in the VoIP environment. Announcements would be played to the caller to indicate how much money to deposit, etc.

- One alternative for call setup and coin collection phase would be to convey the coin control signals in both directions as MIME payload in SIP signaling messages, such as SIP 200 OK and SIP INFO. A format for the control signals would have to be defined. Announcements may be played to the caller from an MS. The OSS and coin phone may wish to use the SIP “early media” capability in [RFC 3959] at this stage in the call, since the call is not yet being sent to a destination party outside the network.
- Another alternative for coin collection would be to convey the coin control signals in both directions as tone events using [RFC 4733].
- A less desirable alternative, because it is more resource-intensive, would be to pass coin control signals as inband tones, as is done today in the PSTN. In that case, the AS would utilize an MS to issue the coin collection tones and to listen for coin collection tones from the VoIP coin phone.

Later in the call, it may be necessary to play the caller more announcements (e.g., need to deposit more coins to continue the call), at which point the AS could orchestrate a temporary return to an MS for such announcements, while keeping the called party leg up, as described elsewhere in this TR.

4.3.2 For PSTN Coin Telephones

This subsection assumes a PSTN-based coin phone and that there is PSTN – VoIP interworking between the coin phone and the OSS. In the PSTN today, on the trunk side, the local office supporting the coin phone may utilize one of several signaling methods to an operator system:

- SS7
- Inband (inband)
- Multiwink (inband)
- Extendend Inband (inband)

Firstly, consider the case where the PSTN path involves an SS7 interface to the VoIP network.

Assuming SIP-I is employed, the AS could receive encapsulated ISUP messages in the SIP signaling, so would know that SS7 interworking is occurring and may know from the ISUP IAM that the line carries a coin line-class-of-service. Alternatively, the AS may have a list of PSTN-based coin phone numbers that it knows it serves. In this case, the OSS could pass coin control/ringback signals toward the coin phone as ISUP content in SIP messages. For example, the AS could create a Service Activation Parameter (SAP) with appropriate content as MIMEed payload using the NarrowBand Signaling Syntax [Q.1980.1] that may go in a 183 Session Progress message, or the AS could create an entire ISUP Facility message with a SAP that could be placed as MIMEed ISUP in a SIP INFO message, depending on the stage of the call flow. If the local office is not using SS7, it would be the responsibility of an interworking point within the PSTN to interwork between SS7 and the inband signaling method used by the local office.

Note that the PTSN-based coin phone would send coin collection signals inband, as today. An MS could be used to receive them, or the AS could use the SIP SUBSCRIBE method with a BE. See section “DTMF/Tone Monitoring”. Note that the current specification of KPML does not include coin collection tones. Therefore, converting the inband coin collection signals to SIP would require extensions to RFC 4730.

Secondly, consider the case where there is no SS7 in the PSTN path, or the PSTN interface to the VoIP network is not SS7 (which the AS might determine, for example, by the absence of an ISUP IAM as payload in the SIP INVITE request it receives on the incoming call). In this case, the AS does know what kind of signaling the coin phone’s local office is using on its trunk interface. Moreover, some inband signaling methods (PTSN) involve winks, for which there is no analogous signal, or need for one, in the VoIP environment.

The OSS cannot signal directly to the coin phone itself, because some signals involve battery polarity reversal on the phone line, which is provided by the local office when it receives appropriate signals on the trunk side.

4.4 Busy Line Verification

4.4.1 For VoIP Endpoints

Support of Busy Line Verification (BLV) in the VoIP environment for VoIP endpoints is first of all a business or service definition question for service providers, because the PTSN BLV feature does not carry over directly.

- In the PSTN, a POTS line can only carry one call at a time, whereas the analogous physical entity in the VoIP world (e.g., DSL or cable access) can carry multiple simultaneous calls/sessions at one time, as well as other types of traffic. This raises a number of questions regarding the definition of BLV service as it applies to a VoIP endpoint, e.g.,
  - What is to be verified?
  - What is the difference between an operator-initiated BLV call (if that is needed) and any other operator-initiated call?
- In the PSTN there is historically a stronger association between a particular line and the party that the caller is trying to reach, although that has been diluted in recent times by call forwarding and person locator type features. In the VoIP world, the SIP URI “address of record” (e.g., sip:mary.smith@anywhere.com) is intentionally typically a logical

address that can be associated with various VoIP endpoints, including multiple endpoints at the same time that can be attempted sequentially or in parallel. If a call attempt receives a “busy”, that means the target party can’t be reached at some assortment of VoIP lines. In that case, does it make sense to verify that any particular line is busy? Additionally, neither the caller nor an operator would necessarily know the identity of any of those lines.

- In the PSTN, the BLV service may be used to determine whether a line is “busy” or not based on detection of a human voice on the line. A scrambler may be used to obscure the actual words. Thus, a data call, when monitored with the use of a scrambler will not appear “busy.” Similarly, an active line without any conversation will not be perceived to be “busy.” In large measure, these definitions result from the tight coupling of the service to the Interrupt service; the call cannot be “interrupted” and the called party notified that the operator has joined the call unless the line is “busy” as defined above. In the VoIP environment, what constitutes a “busy” condition for a BLV-like service would need to be specified, including whether or not is it audio-content based (and if so then what kinds of audio-related media sessions qualify), signaling-based, and/or presence information based.
- In the PSTN environment, generally the caller has attempted to complete a call to the target line (perhaps multiple times) and has not been able to complete the call, but has received a busy signal, and then enlists the aid of an operator for BLV. In the PSTN a busy signal implies that the voice channel is in use and hence unavailable to satisfy a new request for service. (This may be the result of an active conversation, a phone accidentally left off-hook, or a network problem.). However, a VoIP endpoint generally has more flexibility in determining whether it is “busy” – a VoIP line can always receive signaling for an additional incoming call attempt and the called party can elect to receive it or not. A network problem may still cause a busy indication to the caller, and the VoIP endpoint could be configured to present a busy indication whenever there is an existing active voice session. In addition, unlike most PSTN lines, a VoIP endpoint can present a busy indication to the caller simply because the called party rejects the call attempt from that particular caller at that particular time. The BLV feature, applied to the VoIP environment, should account for this additional ambiguity in the interpretation of busy indication.

It is beyond the scope of this TR to propose any particular VoIP BLV-like service, and any such service would have to be evaluated for technical impacts. One could consider, for example, a VoIP BLV-like service that would basically allow an operator to determine whether there was any existing “conversation” involving a particular VoIP address. Alternatively, basing the definition of the BLV-like service in the VoIP environment on the capability of the target line to receive the signaling might be simpler.

4.4.2 For PSTN Endpoints

A VoIP OSS could still support BLV for PSTN endpoints. The OSS could provide the telephone number of the target line as destination address (SIP or tel URI) for the outgoing call, along with some kind of identifier that could cause the egress VoIP Gateway to egress the call on BLV-dedicated trunks. (BLV-dedicated trunks are what are used in the PSTN today.) From there the call would eventually route to and arrive at the end office serving the targeted line. That end office would trigger the BLV capability, involving conferencing the operator onto the call (if one exists) and allowing the operator to hear a scrambled version of the call media to determine whether there is conversation

There is no SS7 mechanism to support any aspect of BLV.

4.5 Operator Interrupt

4.5.1 For VoIP Endpoints

As with BLV, what an Interrupt-like service might be for VoIP endpoints is firstly a service provider question. It could be, for example, interruption of any conversation involving a particular VoIP address. Or, as another example, it could be the delivery of a new call attempt to a particular VoIP address where that new attempt would somehow be forced to the same termination as the existing call and be identified somehow to the called party as an Interrupt call. In any case, any potential Interrupt-like service would need to be assessed for technical impacts.

4.5.2 For PSTN Endpoints

As with BLV, a VoIP OSS could still support Interrupt for PSTN endpoints. After BLV and the detection of voice conversation as described above, the OSS could signal an inband tone that causes the end office to allow the operator to interrupt the conversation and speak to the called party.

There is no SS7 mechanism to support any aspect of Interrupt service.

4.6 Differentiated Treatment

4.6.1 Emergency Calls

A caller may access an operator and request the operator to complete an emergency call, which may be, e.g., 911 or directly to a police or fire department number. To the extent that any such calls are regularly marked in SIP for higher priority, the OSS when completing those calls should do likewise.

Government Emergency Telecommunications Service (GETS) calls do not go through an OSS in the PSTN today. It is assumed that they will also not go through an OSS in the VoIP environment.

4.6.2 Non-Emergency Calls

Non-emergency calls that pass through an OSS may effectively receive differentiated treatment in the PSTN world, because of dedicated access or egress in some cases, or use of different paths through the PSTN that may avoid chokepoints encountered on direct dialed traffic routes. It is assumed that this kind of incidental differentiated treatment is not within scope for Operator Services VoIP standardization.

5 Gap Summary

This section summarizes material covered in previous sections about what is needed to support existing Operator Services given the current state of VoIP architecture and interface standards.

There are basically three types of gaps:

1. Feature definition – a current PTSN-based feature doesn’t carry over naturally to the VoIP environment

2. Architecture – the VoIP architecture needs extensions
3. Interface/Protocol – an existing interface needs extensions

5.1 Feature Definition Gaps

The following PTSN-based features do not have a direct analogy in the VoIP environment. It needs to be decided what the nature of a meaningful substitute VoIP feature would be, if any.

1. Busy Line Verification (section 4.4)
2. Operator Interrupt of a busy line (section 4.5)
3. Operator control of a line (section 4.1.8)

5.2 VoIP Architecture Gaps

Regarding third party service interfaces:

1. If SIP is to be used, the architecture needs to recognize it as an allowed third party Interface
2. If Parlay is to be used with third party Media Servers, then the Parlay architecture model needs to recognize that; either with the Parlay application server controlling the MS, or the Parlay application Gateway controlling the MS.

In order to address WS-type media servers, this document assumes that the concept of the MS in the ATIS NGN architecture includes the potential that the MS may contain application logic, and that application logic may be embodied in software or a human or some combination. In addition, this document assumes that the concept of the Media Resource Broker (MRB) in the ATIS NGN architecture allows operator skill sets as an example among MS attributes, and approaches. Moreover, due to the relatively dynamic nature in the change of attributes for WS-type MSes, methods for populating such knowledge in MRB other than a provisioning interface should be explored; for example methods involving SIP registration or presence.

5.3 Interface/Protocol Gaps

5.3.1 SIP

This section summarizes SIP signaling protocol additions needed to support current operator services. Table 3 identifies the information that is not currently supported and that needs to be passed, and where/when in a call flow that might occur.

The listed needs are to pass additional information in existing SIP messages using existing SIP procedures; no new needs have been identified for additional SIP procedures or messages. Where to carry the information must be determined. The solution may possibly involve official extensions to SIP and/or informally carrying information in existing tags or parameters. Some specific alternatives have been identified in prior sections for carrying some of this information.

For PSTN interworking, any of the information currently available in ISUP can be forwarded to or sent from an AS in SIP using SIP-I [T1.679] (where an entire ISUP message can be carried as payload in a SIP message) or for those parameters that cannot be encoded into a SIP message as NarrowBand Signaling Syntax elements per [Q.1980.1] and [ATIS-1000008] (where specific ISUP parameters can be carried as payload in a SIP message).

Note: In the cases below involving one or more Application Servers, any AS could potentially be a third party AS.

Table 3- SIP Protocol Gaps

	Information	Where/When	Comments
1.	Allowed billing and service options	From one AS to another AS when a call is initiated toward the 2nd AS (i.e., SIP INVITE)	
2.	Call completion eligibility Yes/No	From one AS to another AS when a call is initiated toward the 2nd AS	
3.	Listed directory number	From one AS to another AS during a call redirect or clearing phase	For recording or call completion purposes
4.	"HSP" id (Voice Services Provider)	• From the caller's service provider network to an AS providing operator services, during call initiation	The AS may potentially be elsewhere than inside the HSP network. Would need to decide which provider network element provides this information.
5.	"ASP" id (Aggregation Services Provider)	• From the aggregating network (what element?) to an AS providing operator services, during call initiation	The AS may potentially be elsewhere than inside the Aggregation Services Provider network. Would need to decide which provider network element provides this information.
6.	Nature of calling source	• From an originating side Border Element to an AS during call initiation • From one AS to another AS during call initiation	Like OLI
7.	Correlation ID	• From an AS to a MS when a call leg is initiated toward the MS	for third party AS-MS call leg and control correlation
8.	Charge number	• From one AS to another AS during a call redirection or call clearing	For accounting and charging purposes
9.	ID of Carrier preferred for call completion	• From one AS to another AS during call redirection or call clearing • From an AS to CSCF during call initiation	

5.3.2 Parlay

Gaps in Parlay as a third party service provider interface for operator services are summarized in section 4.1.15.2.

5.3.3 AS-MS Control Protocol

Gaps in the area of AS-MS control protocol for operator services are summarized in section 4.1.1 (AS control of the MS). In particular, note that it is expected that any application logic in the MS does not work at cross-purposes with that in a controlling AS.

5.3.4 Flash-Hook Signal

A signal from VoIP endpoints to indicate the equivalent of flash-hook would need to be specified.

Whether to define it as an inband signal or a SIP signal or as a payload signal within a SIP message needs to be decided.

It is worth noting that, historically, wireless technology faced the same issue – the mobile handset cannot electrically make and break the access connection. There, a tone-based signal of 400 ms sent on the reverse voice channel (typically triggered by depressing the Send key on the mobile handset) was originally defined to be the equivalent of a flash-hook.

6 References

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7 Acronyms

AS	Application Server
ANI	Application-Network Interface
API	Application Programming Interface
BE	Border Element
BLV	Busy Line Verification
CPC	Calling Party Category
CSCF	Call Session Control Function
DA	Directory Assistance
DS	Directory Services
GW	Gateway
IM	IP Multimedia
IMS	IP Multimedia System
IP	Internet Protocol
ISUP	ISDN User Part
kpml	key pad markup language
MRB	Media Resource Broker
MS	Media Server
NGN	Next Generation Network
OI	Operator Interrupt
OLI	Originating Line Information
OSA	Open Service Access
OSS	Operator Services System
PSAP	Public Safety Answering Point

PSTN	Public Switched Telephony Network
RFC	Request For Comments
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
SCS	Service Capability Server
SDP	Session Description Protocol
SS7	Signaling System #7
UNI	User-Network Interface
VoIP	Voice over IP
WS	Workstation (Media Server)