ATIS-1000064

Transitional Circuit Switched/Packet Switched Services

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ATIS-1000064, *Transitional Circuit Switched/Packet Switched Services*

Is an American National Standard developed by the **PSTN Transition (PSTN)** Subcommittee under the **ATIS Packet Technologies and Systems Committee (PTSC)**.

ATIS-1000064

Technical Report on

Transitional Circuit Switched / Packet Switched Services

Alliance for Telecommunications Industry Solutions

Approved October 2014

Abstract:

This Technical Report (TR) analyzes a basic set of services commonly provided in the circuit-switched Public Switched Telephone Network (PSTN), and addresses the technical issues surrounding possible support of these services in a mixed circuit-switch/packet-switched environment and in the future all-packet-switched Next Generation Network (NGN).

Foreword

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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, the PSTN Transition Subcommittee, which was responsible for its development, had the following leadership:

M. Dolly, PTSC Chair (AT&T)

V. Shaikh, PTSC Vice-Chair (Applied Communications Sciences)

M. Dolly, PTSC PSTN Chair (AT&T)

B. Bethea, PTSC PSTN Vice-Chair (Time Warner Cable)

W. Downum, Technical Editor (Ericsson)

The PSTN Subcommittee was responsible for the development of this document.

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1   Introduction

As telecommunication networks migrate from the Circuit Switched (CS) PSTN to the IP-based NGN, there is a need to migrate regulatory mandated and market-driven call management services. During this transition, call management services need to be supported in both environments without disruption to the customer experience.

To assist with that transition, this Technical Report analyzes a broad sample of services, identifying currently deployed and standardized methods for implementing the same services/functions in the IP network. More importantly, the report outlines the gaps in standards that may cause interoperability issues or degradation of service quality. The report offers high level recommendations on addressing these gaps in future standards. It should be noted that not all services that are technically feasible will migrate and be supported in IP networks, due to lack of regulatory requirement or market demand.

In the next generation of services, the emphasis on privacy-enhancing services will continue to grow as consumers seek to identify and manage unwanted calls, especially with the rise of malicious spoofing traffic. Some of the services analyzed in this TR will be impacted by the ease of manipulating the caller's identity in the CS environment and in the IP environment. This issue is independent of migration from CS to IP and is not analyzed further in this TR.

2   Definitions

For a list of common communications terms and definitions, please visit the *ATIS Telecom Glossary*, which is located at < http://www.atis.org/glossary >.

2.1 Hub Provider (Hub): A provider that facilitates the exchange of IP and Time Division Multiplexing (TDM) traffic. Hubs provide multiple services including, but not limited to: protocol conversion, least cost routing, and ENUM mapping.

2.2 Service Transactions Gateway: An extension of the ATIS-1000051 TCAP Gateway functionality responsible for:

- Associating the TCAP Response[1] to the original Query that was launched in support of a value-added service, such as AC or MWI.
- Translating TCAP Components into SIP headers, and vice versa and maintaining the TCAP Component state machine [2]. An example would be translating an Invoke component containing an operation requesting monitoring the busy/idle status of a line into a SIP SUBSCRIBE, as well as translating the TCAP Return Result (idle) [3] into a SIP NOTIFY.
- Performing the necessary service logic to align the functions and communications of the service trigger points and the end users between different networks (NGN and PSTN).

2.3 Trigger Point: An entity or a function that is programmed to detect certain event(s) upon which it executes a series of actions resulting in a service manageable by the end user. The Trigger Point can reside in a gateway, a PSTN end office, an NGN Application Server, or any physical location of the service provider's choice.

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[1] Or a TCAP Conversation or Abort in the Transaction Portion.

[2] Note that a TCAP Unidirectional Component does not require maintaining a TCAP Component state machine at the Gateway.

[3] Or Return Error or Reject component.

3 Scope

This Technical Report (TR) analyzes a basic set of services commonly provided in the circuit-switched Public Switched Telephone Network (PSTN), and addresses the technical issues surrounding possible support of these services in a mixed circuit-switch/packet-switched environment and in the future all-packet-switched Next Generation Network (NGN).

4 Abbreviations & Acronyms

ABS	Alternate Billing Services
AC	Automatic Callback (service)
ACR	Anonymous Call Rejection (service)
ALI	Automatic Location Identification (database)
ANI	(IMS) Application Network Interface
APRI	(SS7) Address Presentation Restriction Indicator
AR	Automatic Recall (service)
ATIS	Alliance for Telecommunication Industry Solutions
BLV	Busy Line Verification (service)
BLV/I	Busy Line Verification and Interrupt (service)
BNS	Billed Number Screening
CC	Calling Card
CF	Call Forwarding (service)
CFB	Call Forwarding Busy (service)
CFNR	Call Forwarding No Reply
CFU	Call Forwarding Unconditional (service)
CFV	Call Forwarding Variable (service)
CIC	(SS7) Carrier Identification Code
CIDW	Calling Identity Delivery on Call Waiting (service)
CNAM	Calling Name (delivery service)
CND	Calling Number Delivery (service)
COT	Customer Originated Trace (service)
CPN	Calling Party Number
CPE	Customer Premises Equipment
CS	Call Screening (service)
DN	Directory Number
DNS	Domain Name System
E9-1-1	Enhanced 9-1-1
EAEO	Equal Access End Office
ENUM	tElephone NUmber Mapping
ESIF	(ATIS) Emergency Services Interconnection Forum
ESN	Emergency Service Number
ETS	Emergency Telecommunications Service

FCC	Federal Communications Commission
GETS	Government Emergency Telecommunications Service
GETS-AN	GETS-Access Number
GETS-NT	GETS-Number Translation
GMLC	Gateway Mobile Location Center
GN	(SS7) Generic Number
IAM	(SS7) Initial Address Message
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IMSESINET	(ATIS) IMS Emergency Services IP Network
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	(SS7) Integrated Services User Part
ITFDB	Industry Toll Free Database
IXC	Interexchange Carrier
IWF	[ATIS-1000679] Interworking Function
KPML	Key Press Markup Language
LATA	Local Access and Transport Area
LDAP	Lightweight Directory Access Protocol
LEC	Local Exchange Carrier
LIDB	Line Information Database
LSDB	Listing Services Database
MF	Multi-Frequency
MGCF	Media Gateway Control Function
MPC	Mobile Positioning Center
MSC	Mobile Switching Center
MSR	Message Storage and Retrieval
MWI	Message Waiting Indication (service)
NANP	North American Numbering Plan
NGES	Next Generation Emergency Services
NGN	Next Generation Network
NNI	Network-Network Interface
NOA	(ISUP) Nature of Address
NS	(ENUM) Name Server
OLNS	Originating Line Number Screening
OSS	Operator Services System
PAI	(SIP) P-Asserted Identity
PC	Personal Computer
POTS	Plain Old Telephone Service
PSAP	Public Safety Answering Point

PSTN	Public Switched Telephone Network
PTSC	(ATIS) Packet Technology and Services Committee
RBOC	Regional Bell Operating Company
RFC	(IETF) Request for Comments
RESPORG	Responsible Organization
SIP	(IP) Session Initiation Protocol
SBC	(IMS) Session Border Controller
SCP	(SS7) Service Control Point
SLE	Screening List Editing (service)
SMDI	Simplified Message Desk Interface
SR	Selective Router
STIR	(IETF) Secure Telephone Identity Revisited (Working Group)
SS7	Signaling System 7
STG	Service Transactions Gateway (function)
TCAP	(SS7) Transaction Capabilities Application Part
TFAS	Toll Free Application Server
TFSP	Toll Free Service Provider
TR	Technical Report
TWC	Three Way Calling (service)
UNI	User-Network Interface
URI	(IP) Uniform Resource Identifier
VoIP	Voice over IP
VMSR	Voice Message Storage and Retrieval
VMWI	Voice Message Waiting Indicator (service)
VPN	Virtual Private Network
XML	eXtended Markup Language

5   PSTN Service Migration Analysis

5.1 Database Services

5.1.1 Emergency Telecommunications Service

[ATIS-1000057] defines Emergency Telecommunications Service (ETS) as a national service, providing priority telecommunications to the ETS-authorized user in times of disaster and emergency and provides descriptions of the government-sponsored services under the umbrella of ETS and their relationships:

- Government Emergency Telecommunications Service (GETS) is one facet of the U.S.A. instantiation of ETS using public telecommunications networks, offered by the government to authorized users for NS/EP purposes. GETS is a circuit-switched form of ETS for voice (and voiceband data) using PIN authorization, in which a user can invoke the service by dialing a GETS-AN or GETS-NT from most phones served by the Public Switched Network (PSN). GETS provides priority treatment across originating, transit, and terminating networks.
- Wireless Priority Service (WPS) is a circuit-switched form of ETS for voice (and voiceband data) using subscription-based authentication, in which a user can invoke the service by dialing a feature code from a WPS-subscribed mobile phone served by a public wireless network. WPS provides

priority treatment across originating and terminating public wireless networks, including priority radio resource assignment upon call origination and termination.

- NS/EP NGN Priority Services (NS/EP NGN-PS) are the evolution of Legacy GETS and WPS to achieve service continuity in the packet-switched NGN and leverage the NGN to offer new features and priority multimedia services.

Signaling for ETS calls includes the marking for the higher priority for call completion (relative to other calls) that has been authorized and call setup procedures are based on these priority markings. Standards for providing ETS include, but are not limited to:

- [ATIS-1000005] Service Description of ETS
- [ATIS-0100009] Overview of Standards in Support of Emergency Telecommunication Service (ETS)
- [ATIS-1000010] Support of Emergency Telecommunication Services in IP Networks
- [ATIS-1000011] ETS Packet Priority for IP NNI Interfaces – Use of Existing DiffServ Per Hop Behaviors
- [ATIS-1000020] ETS Packet Priority for IP NNI Interfaces – Requirements for a Separate Expedited Forwarding Mechanism
- [ATIS-0100022] Priority Classification Levels for Next Generation Networks
- [ATIS-1000023] Network Element Requirements
- [ATIS-1000053] Emergency Telecommunications Service (ETS) Profile and Tests for IP Network-to-Network Interconnection

[ATIS-1000049] provides example call flows for (among other use cases) interworking between the SS7-based PSTN and the IMS-based NGN.

Signaling for ETS calls is similar to ordinary calls but includes an indication that the call should receive higher priority in call setup and optionally includes a priority level. SS7 ISUP and SIP signaling information in support of ETS is described in [ATIS-1000010] and the ISUP/SIP interworking is specified in [ATIS-1000679].

5.1.1.1 Identified Gaps

None identified.

5.1.2 Toll Free Calling
5.1.2.1 PSTN Implementation

The toll free paradigm in the PSTN relies on originating switched access to deliver calls from the originating service provider of the caller to the Toll Free Service Provider (TFSP) serving the toll free number subscriber. The TFSP may be a Local Exchange Carrier (LEC) or an Interexchange Carrier (IXC).

Toll free numbers are portable so originating service providers determine which toll free carrier to route a toll free call to by querying the Industry Toll Free Database (ITFDB), which returns a Carrier Identification Code (CIC) and potentially a translated Plain Old Telephone Service (POTS) number[4]. The Industry Toll Free Database (ITFDB) is not a single entity but a set of SS7 TCAP databases operated generally by the Regional Bell Operating Company (RBOC) descendants. All instances of the ITFDB get its information from a common number management system known as the SMS/800. The SMS/800, in turn, derives routing information for any given toll free number from the Responsible Organization (RESPORG) representing the subscriber that has been assigned that toll free number.

The ITFDB supports a number of vertical features including routing of a toll free number to multiple different carriers on a geographic or time of day basis.

Figure 5.1 below shows the current toll free service architecture, call flow, and provisioning of the ITFDB and SMS/800:

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[4] If the ITFDB does not return a translated POTS number, the CIC identifies the TFSP network that will receive the call setup and separately derive the translated POTS number allowing delivery of the call setup to the subscriber CPE.
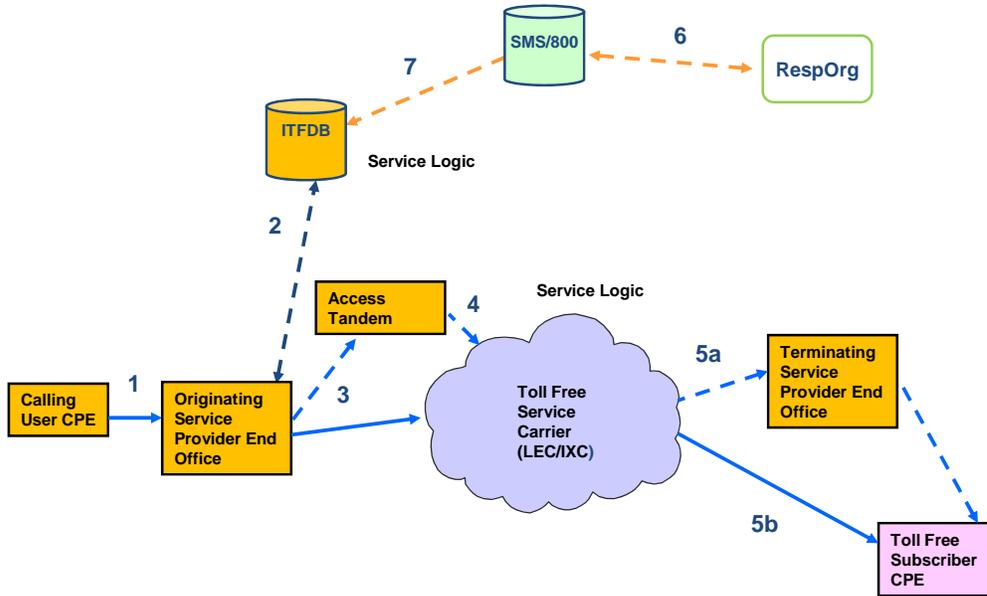
Figure 5.1 - Present Toll Free Service Architecture

Call Flow Steps

1. Caller dials a toll free number.
2. Originating Service Provider queries the ITFDB for routing instructions. (If the Originating Service Provider End Office does not have the capability to query the ITFDB, the Originating Service Provider routes the call to an Access Tandem in its own network or in an Aggregator network.) Note that if the Originating Service Provider does not own an ITFDB, it may make arrangements to query an ITFDB operated by another entity.
3. The Originating Service Provider routes the call to the indicated Toll Free Service Provider. If the Originating Service Provider does not have trunks to the indicated carrier, it routes the call to an Access Tandem that does.
4. The Toll Free Carrier optionally performs additional service processing. This may, for example, include an additional query to another database.
5. The Toll Free Carrier routes the call to the Toll Free Subscriber, either directly (5b) or through the Toll Free Subscriber's local service provider (5a).

ITFDB Provisioning

6. The RespOrg enters routing instructions into SMS/800.
7. SMS/800 downloads routing instructions into ITFDB SCPs.

5.1.2.2 All IP Network Implementation

5.1.2.2.1 Interconnection

Historically, interconnection for delivery of toll free calls from the originating service provider to the toll free service provider made use of Feature Group D switched access trunks with the handoff in the LATA of origination[5]. These same trunks are also used for delivery of 1+ originating long distance traffic and for termination of toll calls. As aggregators of originating toll free traffic entered the market they handed off traffic at a smaller number of points and the hand off points no longer necessarily reflect the LATA of origin.

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[5] If the local CIC 0110 is returned, the originating service provider may complete the call directly to the translated POTS number.

Industry discussions and evolving practice of IP interconnection for general (non-Toll-Free) call termination suggest that IP interconnection between a pair of service providers is likely to take place at an order of magnitude fewer points – maybe just a handful – than is the case with today's LATA-based interconnection. Delivery of originating toll free calls to the toll free service provider should take place over the same IP interconnection fabric and architecture used for termination of geographic number calls.

5.1.2.2.2 Routing

Ultimately, post transition, the industry may leverage the capabilities of Internet Protocols to move the execution of service logic (in this case, the determination of the Toll Free Service Provider based on vertical features and/or the derivation of the translated number) out of the originating service provider network to a locus designated by the toll free number subscriber via its RESPORG, simplifying the common industry infrastructure required.

For example, an ENUM query could point to a Toll Free Application Server (TFAS) in a Toll Free Service Provider's network or some other network selected by the toll free subscriber or their RESPORG. The selected TFAS could execute the type of service logic that today is executed in the originating service provider's ITFDB and then return a URI which would identify the Toll Free Service Provider SBC to which to route the SIP INVITE[6].
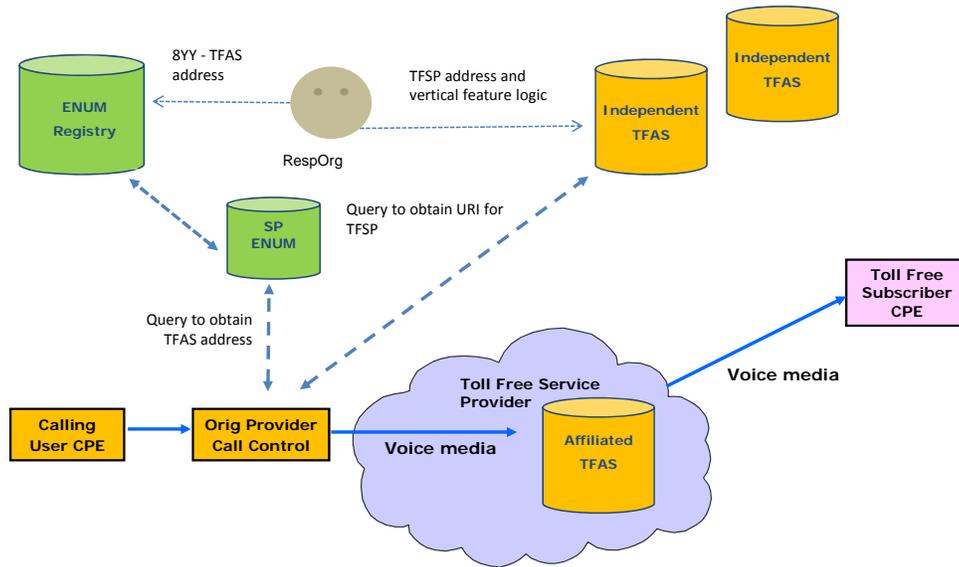
The approach is shown in Figure 5.2 below:

Figure 5. 2 - Toll Free Service Architecture in an All IP Network

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[6] Analogous to the PSTN implementation, this URI could contain full routing information to route the SIP INVITE to the subscriber CPE or could only identify the TFSP assuming further server logic in that network will derive the routing information to the subscriber CPE.

As shown in the figure, the RESPORG would populate the ENUM registry with a Name Server (NS) record as for a geographic number and registry data would be replicated in an originating service provider's ENUM name server. The RESPORG would also provide routing instructions to the selected TFAS which could be one that resides in a Toll Free Service Provider's network –"Affiliated TFAS" (as might be expected where the TFSP is the RESPORG for its customer's toll free numbers) or an independent one.

When a call to a toll free number is initiated, the originating service provider would query its ENUM server which would return a NS record(s) as in the case for a geographic number. The originating service provider uses the NS record to query the TFAS and receive a URI to which it would address a SIP INVITE. As with geographic numbers, although the query protocol is ENUM, the responding server need not be a simple DNS. Instead it could be capable of applying service logic similar to the ITFDB SCP to affect different routing policies including the support of multicarrier routing.

It may be noted that current toll free processing in the ITFDB may be based on information about the calling party location and that standard [RFC 6116] ENUM queries do not include calling number. Proposals have been made, however, that would include such information [draft-kaplan-enum-source-uri-00]. Moreover, the effectiveness of calling number based routing is diminishing. In response to this situation, independent parties have developed services that provide more accurate location information. (See < http://www.locationsmart.net/platform/ >.) These types of services could be utilized by a TFAS outside the originating SP network to achieve location based routing.

5.1.3 Emergency Calling

In North America today, emergency calling is typically provided by a service referred to as Enhanced 9-1-1 (E9-1-1) Service. E9-1-1 Service is a public safety feature that allows customers to report an emergency or request emergency assistance by dialing the 3-digit telephone number "9-1-1." Emergency calls originated in this manner are then routed to the appropriate Public Safety Answering Point (PSAP) based on the location from which the call was originated. Depending on the municipal requirements and procedures, a PSAP may transfer the call to the proper public safety agency (e.g., police, fire, emergency medical service), collect and relay emergency information to the proper agency, or dispatch Emergency Service personnel directly for one or more participating agencies.

E9-1-1 Service is broken down into a number of critical service capabilities that must be supported in any E9-1-1 Service architecture. First, the network node (i.e., end office switch or Mobile Switching Center [MSC]) serving the originating end user must be able to recognize when a customer is requesting E9-1-1 Service. That is, the originating network must be able to detect that the customer has dialed the digits "9-1-1," and identify the call as an Emergency Call. Using the architecture implemented most widely in the PSTN today, upon recognizing an originating call as an Emergency Call, an end office/MSC will use normal trunk selection procedures to select an outgoing dedicated trunk group toward a specially equipped tandem switch referred to as an E9-1-1 tandem or Selective Router (SR). The E9-1-1 tandem/SR is responsible for selectively routing Emergency Calls to the proper PSAP based on caller identification/location information (e.g., the caller's Directory Number [DN], billing number, or location identification information). The Selective Routing function includes the ability to perform default and alternate routing of Emergency Calls based on the time of day or network conditions (e.g., traffic busy, failure). Currently, an E9-1-1 tandem/SR interacts with a Selective Routing database for assistance in selecting a PSAP destination. The PSAP destination is expressed in terms of an Emergency Service Number (ESN).

Once the PSAP has been identified, the call must be delivered to that PSAP, along with the necessary caller location and/or callback information. The PSAP will use the information received with the call to query an Automatic Location Identification (ALI) database. The ALI provides location information (e.g., street address, in the case of a wireline call) that can be used to identify the incident site. For legacy wireless emergency originations, the ALI may have to query a Mobile Positioning Center [MPC]/Gateway Mobile Location Center [GMLC] to obtain location information. The location information initially returned by the MPC/GMLC will typically provide the x,y coordinates of the cell tower that handled the emergency call. The location information is conveyed to the appropriate emergency service agency to support the dispatch of personnel to the incident location.[7]

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[7] In the case of wireless emergency originations, the PSAP may subsequently request location updates in an effort to gain more precise caller location to support the dispatch of emergency personnel.

Upon receiving an Emergency Call, a PSAP may determine that the call must be transferred to a different destination. The network node serving the PSAP must be capable of supporting the transfer of an Emergency Call to a destination served by the same or a different network node, if a PSAP determines that such a transfer is necessary.

Today, emergency calls are commonly delivered by wireline end offices or MSCs to E9-1-1 tandems/Selective Routers via SS7-supported trunk groups, although there are still areas where Multi-Frequency (MF) trunking is used. [ATIS-1000628] describes the SS7 protocol and procedures to support the delivery of wireline and wireless emergency call originations to E9-1-1 tandems/SRs with callback and/or location information. While [ATIS-1000628] supports the delivery of geodetic location in the ISUP IAM, current implementations of E9-1-1 Service only support the delivery of a "location key" that is in the form of a 10-digit North American Numbering Plan (NANP) number. Details regarding MF-based implementations can be found in [GR-350-CORE].

5.1.3.1 Transition to Next Generation 9-1-1 (NG9-1-1)

Transition to Next Generation 9-1-1 is documented in [ATIS-0700015].

Any further modifications to transition for this service will be developed by the joint ATIS IMSESINET project led by the WTSC Systems and Networks subcommittee and coordinated with ESIF NGES and PTSC SAC.

5.1.4 Line Information Database (LIDB) Services

A Line Information Database is a transaction-based distributed database system containing information associated with working telephone numbers (landline, wireless, and VoIP). Each LIDB contains a unique set of telephone numbers and associated data attributes. The LIDB, its administrative systems and interfaces are described in a number of Telcordia Generic Requirements. The key reference is [GR-1158-CORE]. LIDB data is used to support fraud reduction, identity management, and communications-related services. The following are a few examples of LIDB line-level attributes:

- Identification of the customer's service provider/carrier for branding and billing.
- Billing name and address to verify a customer or to reduce billing inaccuracies and fraud.
- Preferred language of the customer/household for providing customized services.
- The name of the customer and the privacy status (private name or not).
- The type of service or equipment for the line number, e.g., hospital, prison, business, residential, etc.
- Indication of how long the telephone number has been in service which may help reduce billing fraud.
- Information indicating whether or not third party charges can be included on the end user's bill.

Only authorized clients/Query Originators having contractual agreements with the LIDB provider or with a hub provider can access a LIDB. A hub provider (or hub) is a communications entity that acts as a gateway to all LIDBs, providing interconnection, transport, access, and protocol conversion services to LIDB clients. The hub supplies or provides access to a transaction gateway function. The hub provides transparent conversion of messages from one format to another in order to eliminate the need for making frequent and significant interface changes across multiple networks. These translations may occur between standard or non-standard protocols.

LIDB services, such as GetData [GR-2838-CORE], Originating Line Number Screening (OLNS) [GR-1158-CORE], and CNAM [GR-1188-CORE], are accessed via predefined SS7 TCAP queries. However, over the past 13 years, LIDB companies have provided a variety of architectures that support convenient TCP/IP access to LIDB (in addition to the SS7 interfaces). One such example is the Lightweight Directory Access Protocol (LDAP). Various other proprietary IP queries are also in use today (e.g., XML queries, or queries delivered as payload within SIP INFO messages). A hub provides access and translation between the various proprietary protocols and the set of protocols supported at any specific LIDB making access to LIDB data virtually protocol-agnostic. Clients are typically provided Virtual Private Network (VPN) access to the hub provider's applications platform. Different types of queries are handled by the servers on these platforms. Connections to the applications platform are mainly IP.

Alternate Billing Services (ABS)

In the 1980s, LEC-owned LIDBs enabled the LECs to validate billing numbers using their own network elements. One set of services using LIDB data was ABS, which permits end users to obtain service (e.g., place a call) and have the charges billed to an account other than the account that is associated with the originating station. These billing options include collect, calling card, and third number billing. LIDBs contain information about valid calling cards and other billing numbers. There are two types of LIDB queries under the "ABS" umbrella: Calling Card (CC), and Billed Number Screening (BNS) to support collect and third number billing and they are both described in [GR-1158-CORE].

Over the past decade, ABS queries have diminished due to the widespread adoption of wireless services, Internet voice dialing plans, and a variety of economical pre-paid calling cards.

Identity Verification Services

Identity Verification is a service where LIDB supports businesses in the detection of suspicious/potentially fraudulent credit card transaction activity. The information supplied by the end user is compared to the data in LIDB. The initial identity verification of end user data generally includes data such as name, billing address, and length of telephone service. The LIDB Identity Verification service then sends a response code indicating the results of the match to the business that launched the query. If there is a discrepancy when the data is compared, it is an indication to a business that further investigation is warranted. Based on that response and several other criteria the business applies, the transaction is either accepted or rejected.

Multiple industries, including Retail, Banking, and Healthcare, rely on LIDB to perform this "pre-screening/validation" of the consumer-provided data for online transactions and other purposes.

An Identity verification query is an example of a "proprietary" query that a hub provider facilitates. Different industries (healthcare, retail, etc.) may have different preferences for the protocols they use and the data they need from LIDB. Custom conversions facilitated by the hub providers – as described in Section 5.1.5.1 – will ensure a seamless migration of the Identity Verification service.

5.1.4.1 All IP Network Implementation

In an all IP network, the LIDB owners will continue to serve their IP clients with many of the same capabilities that are offered today. As new queries-responses are introduced access to LIDB will continue to be supported through the hub.

5.1.4.2 PSTN-IP Interworking

LIDB query processing via hub providers has proven to be protocol agnostic for over two decades without service interruption or critical failures. In order to maintain the flexibility of serving new clients with their rapidly changing applications, the LIDB owners rely on the hub providers to provide the protocol conversion and query-response customization.

Therefore, it is not anticipated that significant additional requirements are needed to migrate access to the LIDBs from the PSTN through the mixed PSTN/NGN environment and on to the IMS-based NGN environment.

5.1.4.3 Identified Gaps

None identified.

This Technical Report concludes that there is no need to develop any further evolution plan to migrate these services from the PSTN through the mixed PSTN/NGN environment and on to the IMS-based NGN environment.

5.1.5 Calling Name Delivery (CNAM)

CNAM is a Caller ID-based service that allows the subscriber to receive the calling party's name and the date and time of the call on a specialized display device before the call is answered. The calling party name

information is retrieved from a database via SS7 or IP. Calling Name Delivery is described in [GR-1188-CORE].

In the SS7 network, the information that is finally presented to the CNAM subscriber is determined through a combination of values of the ISUP Calling Party Number, the Generic Name Parameter (in the IAM) and the TCAP Generic Name Parameter for that calling party. Similarly, in the NGN the presentation of the name to the subscriber will depend on the values received in the P-Asserted Identity (PAI) header field.

5.1.5.1 CNAM Display Rules

In the SS7 network, service providers offer CNAM under a strict regulatory framework governing the display of the calling number and name based on the privacy status of the originating party. If the originating party number is private, the name – even if available from the database or otherwise – should not be displayed. This is in accordance with the FCC's Rules and Policies regarding Calling Number Identification Service – Caller ID, CC Docket No. 91-281 which requires that blocking a number also blocks name delivery. This document assumes the same rules apply to the IP offering of CNAM.

Table 5.1 summarizes the possible combinations of ISUP parameters and the corresponding Calling Number/Name Display outcome.

Table 5. 1 - Summary of CNAM Display Rules

ISUP CPN	ISUP GN Presentation	TCAP GN Presentation	Number Display	Name Display
Not received	-- n/a --	Query not launched	Unavailable	Unavailable
Restricted	-- n/a --	Query not launched	Private	Private
Allowed	Not received	Allowed	Number	Name
		Restricted	Number	Private
		No indication / Other	Number	Unavailable
	Allowed	Allowed/ Restricted/ No indication / Other	Number	Name
	Restricted	Query not launched	Number	Private
	Toggle	Allowed	Number	Private
		Restricted	Number	Name
		Not indicated/ other	Unavailable	Unavailable

5.1.5.2   All IP Network Implementation

In an all IP network, the name may be derived from the SIP INVITE. Alternatively, the terminating IP service provider may query a Name Database if:

  a) The name information is not received in the SIP INVITE, or
  b) The provider chooses to retrieve trusted name information from a reliable database.

The rules for displaying the calling name to the terminating SIP phone should follow the essence/intent of Table 5.1, preserving the Calling Name privacy end to end, in congruence with the privacy of the calling number.

End to end SIP signaling will require the establishment of a trusted model that will verify or guarantee the name information is accurate over the entire signaling path.

5.1.5.3 PSTN-IP Interworking

In a PSTN-IP interworking scenario, the relevant entities in a CNAM feature offering are:

  a) The CNAM database,

b) The Gateways between the SS7 and the IP networks (ISUP and TCAP as defined in [ATIS-1000679] and [ATIS-1000051]),
c) Hub providers, and
d) The IP and SS7 CNAM feature trigger points are the points where the CNAM feature is activated, queries are launched to the database and caller ID data (number + name) is assimilated and sent toward the subscriber. The CNAM feature trigger point may or may not reside at either Gateway. The Trigger Point is assumed to reside in the same network as the subscriber (IP or SS7).

The calling name is assumed to be derived at the terminating side of the session by querying a database.

Requirements governing the name database (structure, data catalog, security, etc.) are described in [GR-1188-CORE] and [GR-1158-CORE], respectively.

SS7 to SIP Interworking Scenario

- On calls from an SS7 caller to the SIP CNAM subscriber, the ISUP Gateway will map the IAM Calling Party Number parameter to the SIP P-Asserted-Identity header fields according to [ATIS-1000679], Section 7.1.3, where the 'privacy' header content or presence will be determined from the Address Presentation Restriction Indicator (APRI) value of the Calling Party Number only.
- If the IP CNAM Trigger Point determines the Caller Name should be retrieved (i.e., the CPN is available and its APRI is not restricted), it will launch a query to the LIDB/CNAM database. This query may transit an [ATIS-1000051] TCAP Gateway.
  a. Alternatively, the IP CNAM Trigger Point may forward the query – in any protocol of their choosing – to a hub provider service that could launch the query on their behalf. This is a common practice among IP entities when accessing the LIDB/CNAM database. An IP CNAM Trigger Point may launch queries in XML, LDAP or other protocols and the hub provider applies the necessary protocol translations.
- The CNAM database will return the calling name (via the hub provider, if one was used for the query) to the IP CNAM Trigger Point.
- The IP CNAM Trigger Point will map the contents of the received Name into the display-name portion of the P-Asserted ID sent to the CNAM subscriber.

SIP to SS7 Interworking Scenario

- On calls from SIP callers to the SS7 CNAM subscriber, the ISUP Gateway will map the SIP P-Asserted-Identity header fields to ISUP Calling Line Identification parameters according to [ATIS-1000679], Section 6.1.3.6.1[8].
- The SS7 CNAM Trigger Point will determine the presentation status of the Calling Number and Name based on the APRI values of the CPN and Generic Name received in the IAM and the rules provided above in Table 1.
- If the SS7 CNAM Trigger Point determines the Caller Name should be retrieved to be displayed to the CNAM subscriber, the SS7 Trigger Point will launch a TCAP query to retrieve the name directly, according to [GR-1188-CORE].
- The SS7 CNAM Trigger Point will deliver the name information received from the database to the SS7 CNAM subscriber via the interface defined in [GR-30-CORE].

5.1.5.4 Identified Gaps

None identified.

5.2 Line-Based Services

Line-based services provide customers with various methods to interact, and manage both incoming and outgoing calls. Some line-based services provide customers with basic call control functionality (e.g., Call Waiting (CW) and Call Forwarding (CF)). Others allow more control by taking advantage of the incoming CPN

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[8] [ATIS-1000679] does not provide mappings for a Remote Party ID header, therefore, if a RPID is received, the P-Asserted Identity header values in the INVITE message should take precedence.

information (e.g., Calling Number Delivery (CND) and SLE services). As technology and the network evolved, so have these services, allowing customers to receive more information, protect their privacy, and manage their lifestyles. With the migration to an all IP network, these services will again undergo a transformation.

In addition to the gaps in standards that may cause interoperability issues or degradation of service that need to be resolved if the services were to transition from the PSTN to the NGN, there are potential operational issues (not gaps) which could impact the integrity, service functionality, and economic viability of services that make use of the CPN. The services discussed in this TR that rely on the CPN include ACR, AC/AR, COT, CIDCW, CNAM, CND, and SLE services. These operational issues are summarized below for informational purposes. The identification of potential resolutions for these issues is outside the scope of this TR.

- In some ISUP-SIP interworking scenarios, unverified CPN data received from the PSTN, as determined by the Calling Party Screening Indicator, will result in the absence of the PAI in the NGN. This could reduce the perceived value of the service to the NGN end user for a period of time until more calling numbers undergo verification. Fraud and incorrect service functionality are likely to occur if the calling number and its presentation status were to be delivered to the end user from unverified sources.
- Similarly, the absence of a complete E.164 in the CPN received from the PSTN (e.g., a blank calling number) will result in the absence of the PAI – according to ATIS-1000679, Table 28 – which would reduce the service functionality and the customer's perceived value of the service.
- Furthermore, the number of malicious callers relaying false CPN information providing a false Caller ID with the intent to defraud the called party (identity theft, eliciting sensitive personal information) has increased because it is much easier to disguise the caller's original phone number with VoIP; do-it-yourself tools for the layman are plentiful in today's market. As a result, Caller ID-based services have become more susceptible to such fraud. Congress is considering legislation to protect consumers against such fraud (i.e., caller ID spoofing). Telecom service providers are investigating verification solutions to increase the reliability and authenticity of the delivered calling numbers, but those solutions are outside the scope of this TR. It should be noted that there are legitimate reasons for changing the original CPN in some cases. However, from a purely operational perspective, the capability to change the original CPN may still lead to incorrect service operation and fraud.

5.2.1 Anonymous Call Rejection (ACR)

Anonymous Call Rejection allows customers to reject calls from lines that have a privacy feature preventing the delivery of their CPN to the customer. Calls with a CPN that is anonymous are routed to a network announcement and the customer does not receive alerting for the call. The announcement informs the calling party that the customer does not accept anonymous calls and the caller would have to unblock his/her number to reach the customer. The NANP activation/deactivation codes for ACR are *77/*87, respectively.

ACR is described in [GR-567-CORE].

5.2.1.1 All-IP Network Implementation

There is an equivalent mechanism to ACR described for SIP users in [RFC 5079], *Rejecting Anonymous Requests in SIP.* This RFC introduced the 433 (Anonymity Disallowed) response code to specifically indicate the reason for rejection. This allows for proper automated machine handling; not just human interpretation.

5.2.1.2 PSTN-IP Interworking

This feature relies on the delivery of the presentation status of the calling party number to the called party. [ATIS-1000679] attends to the SIP-ISUP requirements handling the translation of the necessary presentation status indicators. However, feature-specific elements necessary for interworking are currently not supported, as explained below.

Interworking with an IP ACR Subscriber

On calls from an SS7 caller to the SIP ACR subscriber, the ISUP Gateway will map the IAM Calling Party Number parameter to the SIP P-Asserted-Identity header fields according to [ATIS-1000679], Section 7.1.3, where the 'privacy' header content will be determined from the APRI value of the Calling Party Number.

If the Calling Party Number was omitted/not available, or if the APRI value is "presentation allowed," the incoming call/session is offered to the ACR subscriber. If the APRI value is "presentation restricted," the user agent serving the ACR subscriber is expected to refuse to fulfill the request by sending the 433 response code defined in [RFC 5079] for this purpose. The mapping for the 433 response code is not included in the list of 4xx codes recognized by the [ATIS-1000679] IWF. It is likely that this message would be rejected at the PSTN Gateway.

Interworking with a PSTN ACR Subscriber

On calls from SIP callers to the PSTN ACR subscriber, [ATIS-1000679] IWF will map the SIP P-Asserted-Identity header fields to ISUP Calling Line Identification parameters.

The SS7 switch serving the ACR subscriber is expected to follow procedures in [GR-567-CORE] and the call will not be terminated to the ACR subscriber. In addition, the switch plays an audio announcement indicating that the called party does not accept anonymous calls. Since the announcement is played from the switch serving the ACR subscriber, there is no signaling interworking back to the caller.

5.2.1.3 Identified Gaps

A SIP 433 response code to an anonymous PSTN caller is expected to cause release treatment on the PSTN side, causing confusion on the part of the PSTN caller, potential reattempted calls that will result in repeated failures.

To avoid inconsistent implementations of ACR between the PSTN and IP, it is strongly recommended that an audio announcement be generated BEFORE sending the 433 response code. This way, anonymous callers would receive a clear indication of why their calls are not being completed.

5.2.2 Automatic Callback (AC) and Automatic Recall (AR)

[GR-215-CORE] defines Automatic Callback as:

> An outgoing call management feature that allows customers to perform an activation procedure to set up a call to the last station that the customer called without the customer having to redial the telephone number. If the called party is busy when AC is activated, call setup is performed automatically when the called station becomes idle. The NANP activation/deactivation codes for AC are *66/*86, respectively.

[GR-227-CORE] defines Automatic Recall as:

> An incoming call management feature that allows a customer to perform an activation procedure to automatically set up a call to the last incoming number. The AR customer does not need to know the telephone number or the calling party of the last incoming call.

The NANP activation/deactivation codes for AR are *69/*89, respectively.

[ATIS-1000611] defines AC and AR in the context of a Multi-Location Business Group, noting that:

> AC and AR are very similar MBG features. The AC feature attempts to call the number associated with the most recent outgoing call made by the customer. The AR feature attempts to call the number associated with the most recent incoming call received by the customer.

5.2.2.1 All-IP Network Implementation

Comparable AC/AR monitoring of the busy/idle status of the terminating party could be accomplished through SUBSCRIBE and NOTIFY requests as described in [RFC3842] and [RFC 6665], Session Initiation Protocol (SIP)-Specific Event Notification. The calling user agent sends a SUBSCRIBE requesting to be notified when the terminating user agent becomes available. The terminating user agent would send a NOTIFY response when it becomes available. If the calling user agent is available, a new INVITE is sent and the session is established.

In addition, [RFC 6910] defines "Completion of Calls for SIP," which uses SUBSCRIBE-NOTIFY to allow the caller of a failed call to have the call completed without having to make a new call attempt. [RFC 5359]

provides a call flow example using the name "Automatic Redial". The feature closely resembles the PSTN AC/AR services and offers more functionality that is suitable for SIP implementation.

> NOTE: Currently, there is no SIP EVENT type describing voice media explicitly. It is possible to assume EVENT: available for all media types, including voice, thus supporting the use of SUBSCRIBE-NOTIFY to implement AC and AR.

5.2.2.2 PSTN-IP Interworking

In the PSTN, the monitoring of the busy/idle status of the line to be called may be carried out by (1) the originating switch launching periodic TCAP queries to the terminating end requesting the target status, or (2) the originating switch requesting that the terminating switch perform terminating scanning and inform it only when the called party becomes available.

Comparing the capabilities available in the SIP protocol to those necessary to deliver an equivalent AC experience to its end users, the SUBSCRIBE and NOTIFY methods appear to satisfy most of these capabilities effectively. SUBSCRIBE-NOTIFY method is expected to be the prevalent method for implementing an event-based service, such as AC, between two SIP users.

This TR is not the appropriate vehicle for feature requirements, therefore the following text is provided on an informational basis, highlighting potential areas for investigation. A Service Transactions Gateway will potentially perform the TCAP-SIP mapping and provide the necessary monitoring and notification. Standardizing the translation function (TCAP Components into SIP headers and vice versa), managing transaction IDs, busy/idle scanning requests, and handling errors would be some of the key responsibilities of the Service Transactions Gateway.

5.2.2.3 Identified Gaps

Interworking an AC request between SIP and SS7 is not defined in standards today. Informative interworking guidelines need to be added to [ATIS-1000051].

AC services based on [GR-215-CORE] include an initial TCAP Query requesting information about the services, features, and restrictions on the terminating line prior to initiation of scanning for the line to become idle. The functionality provided by this initial Query is not available in NGN functionality provided by the SUBSCRIBE-NOTIFY method. SIP extensions, service-specific TCAP-to-SIP interworking, or AC use of TCAP encapsulation as described in [ATIS-100051] may need to be standardized.

5.2.3 Customer Originated Trace (COT)

Customer Originated Trace allows the recipient of harassing calls to initiate a trace of the last call presented to the COT customer's line. COT information (i.e., CPN, date, time associated with the call, and the date and time of the trace activation) is sent to a designated retention location. The information is therefore potentially available for further use by the service provider and/or law enforcement. If the captured incoming number is invalid (i.e., unavailable or partial), the customer is informed that a trace cannot be done and an announcement is given to indicate why. In some offerings, the announcement includes the contact information for a local Law Enforcement Agency. The NANP activation code for COT is *57.

COT is described in [GR-216-CORE].

The interface that delivers the trace information to the retention/processing location is outside the scope of this TR.

This TR assumes the agency (Law Enforcement Agency or other) that receives and retains the COT data resides locally, within the same network (PSTN or IP) as the COT customer.

5.2.3.1 All IP Implementation

Commercial VoIP offerings of COT service already exist. While there are no standard descriptions of COT in an IP environment, [RFC 5503] introduces SIP extensions that support COT.

5.2.3.2 PSTN-IP Interworking

For this feature to be successfully activated, the CPN of the caller must be available (anonymous CPNs are captured for the Trace log but are not revealed to the COT subscriber).

SS7 to SIP Interworking Scenario

On calls from an SS7 caller to a SIP COT subscriber, the ISUP Gateway will map the IAM Calling Party Number parameter to the SIP P-Asserted-Identity header fields according to [ATIS-1000679]. Therefore, the required Calling Party Number and privacy indication will be delivered to the terminating end and will be available for logging as a COT is activated by the called party.

SIP to SS7 Interworking Scenario

On calls from SIP callers to the SS7 COT subscriber, the ISUP Gateway will map the SIP P-Asserted-Identity header fields to ISUP Calling Line Identification parameters according to [ATIS-1000679]. Therefore, the required Calling Party Number and privacy indication will be delivered to the terminating end and will be available for logging as a COT is activated by the called party.

5.2.3.3 Identified Gaps

[RFC 5503] SIP uses an INVITE to deliver the COT data to the designated agency, except as noted in the excerpt from [RFC 5503] below, for a subset of cases. If the INVITE is forwarded within the IP network, the COT request will complete as intended. However, if the INVITE traverses a PSTN gateway (i.e., the COT customer and the designated agency are served by different networks – one SIP-based and the other SS7 – the COT data and call content will be lost and the request will fail.

> "To initiate a customer-originated-trace from an untrusted User Agent Client (UAC), an additional header is defined for the INVITE request...The entity addressed by the Request-URI performs the service-provider specific functions of recording and reporting the caller identity in the P-DCS-Trace-Party-ID for law enforcement action. It then forwards the call to either an announcement server or to the service provider's business office to collect further information about the complaint. A trusted UAC does not use this header, as it initiates this action locally."

5.2.4 Calling Number Delivery (CND)

Calling Number Delivery is an incoming feature that provides the CPE with the calling party's number, date, and time of the incoming call, during the silence between the first and second ring. If the calling party number is "anonymous" or unavailable, an indication is displayed in the CPE in place of the number. CND is described in [GR-31-CORE].

5.2.4.1 All IP Implementation

The basic CND function was built into the SIP From header. However, since this header is populated by the user, the calling ID is not guaranteed to be accurate; hence, the P-Asserted-Identity header which provides a value analogous to that of the Calling Party Number in the PSTN.

5.2.4.2 PSTN-IP Interworking

The mapping of P-Asserted-Identity and Privacy headers to ISUP Calling Party Number parameter is documented in [ATIS-1000679].

5.2.4.3 Identified Gaps

None identified.

5.2.5 Message Waiting Indication (MWI) / Visual Message Waiting Indicator (VMWI)

The Message Waiting Indication (MWI) service enables the network to indicate to the user that there is at least one message waiting. [ATIS-1000611] notes subscription options for the alert provided to the end user to be a message waiting light or stutter dial tone, but does not specify the protocol at the User-Network Interface (UNI). [GR-1401-CORE] extends the description of the service beyond the [ATIS-1000611] intranetwork signaling, adding the UNI signaling to the CPE and adding a text display option.

In the PSTN, the service is administered by the service provider (i.e., is activated on a subscription basis for the end user). From that point forward, the MSR communicates the notifications to the switch serving the subscriber via a number of interfaces, including but not limited to the following:

a) SS7, or
b) Proprietary interface, or
c) Simplified Message Desk Interface (SMDI) described in [GR-283-CORE], or
d) Dedicated Data Link Interface described in [GR-1193-CORE], or
e) ISDN Message Service Generic Switching and Signaling Requirements described in [GR-866-CORE]

The switch then relays the notification to the end user via another interface described in [GR-30-CORE].

5.2.5.1 All-IP Network Implementation

[RFC 3842] describes "A Message Summary and Message Waiting Indication Event Package for SIP" which expands the MWI service to multimedia messages such as email, fax, voice and video mail.

- Typically, a SIP user will launch a SUBSCRIBE request to be notified when the status of messages changes.
- The Notifier (the MSR or an entity acting on its behalf) sends a message summary in a NOTIFY each time the account messages change, showing the number of old and new messages, as well as possibly the type of message (voice, fax, other).
- The capabilities and tools necessary for implementing Message Waiting Indicator in a SIP network are described in [RFC 3842] and [3GPP 24.606].

5.2.5.2 PSTN-IP Interworking

If the MSR server and the MSR subscriber are in the same (PSTN or NGN) environment, there is no PSTN / NGN interworking necessary.

IP-based MSR - PSTN Subscriber

If the MSR system resides in an IP network and the MWI subscriber is served by the PSTN, the end user is not expected to be able to launch a SIP SUBSCRIBE. Therefore, administrative steps may need to be taken to ensure the subscription of the PSTN user is in place, even if the IP-based MSR does not receive a SUBSCRIBE request. This is the case of an implicit SUBSCRIBE. Translation between the TCAP Message Waiting Notification message and the SIP NOTIFY will also be necessary. The Service Transactions Gateway is expected to carry out that role, as follows:

- Map the SIP NOTIFY request with "Messages-Waiting: yes," into a TCAP Query With Permission containing an Invoke component of "Report Event-Voice Messages Available" which will cause the subscriber alert to turn on.
- Map the SIP NOTIFY request with "Message-Waiting: no" into a TCAP Query With Permission containing an Invoke component of "Report Event-Voice Messages Retrieved," which will cause the subscriber alert to turn off.

However, context information (priority level and type of message available to NGN users) will be lost at the Service Transactions Gateway. Ideally, the MSR would be aware that the subscriber is in the PSTN and would not forward context information, thus avoiding unnecessary error conditions.

PSTN MSR – IP-based Subscriber

An IP subscriber launches a SUBSCRIBE request to initiate (and periodically renew) the Message Waiting Indicator service. Since a SIP SUBSCRIBE does not have an SS7 TCAP equivalent, additional procedures may be necessary at the STG to ensure the IP customer experience is preserved.

1. According to [GR-1401-CORE], when a change in the MSR status takes place, the MSR server sends a TCAP Query With Permission containing a Message Waiting Notification (MWN) control request. This request, which contains the destination number and the calling number, is sent in either a "Report Event-Voice Messages Available" or "Report Event-Voice Messages Retrieved" Invoke-Last component.
2. Since the subscriber is in an IP network, the TCAP message needs to be routed to an STG to be mapped to a SIP NOTIFY request.
3. [RFC 3842] provides detailed descriptions on formatting the corresponding NOTIFY request that should be sent by the Notifier to the subscriber. A SIP NOTIFY request with statusMessages-Waiting: yes will cause the subscriber alert to turn (or remain) on; a NOTIFY request Messages-Waiting: no will cause the subscriber alert to turn off.

5.2.5.3 Identified Gaps

Interworking an MWI activation/deactivation request between SIP and SS7 is not defined in standards today. Informative interworking guidelines and the role of the Service Transactions Gateway could be added to [ATIS-1000051].

5.2.6 Call Forwarding (CF)

[ATIS-1000611] defines three types of Call Forwarding service. [GR-586-CORE] describes several types of Call Forwarding Subfeatures, including the three types of redirection listed here:

1. Call Forwarding Unconditional (CFU) permits a served user to have the network send all incoming calls for the served user's number to another number. The served user's originating service is unaffected. If this service is activated, calls are forwarded no matter what the condition of the access. This is also known as Call Forwarding Variable (CFV), which is described in [GR-580-CORE]. The NANP activation/deactivation codes for Call Forwarding Variable are *72/*73, respectively.
2. Call Forwarding Busy (CFB) permits a served user to have the network send all incoming calls for the served user's number that meet Busy to another number. The served user's originating service is unaffected. This is also known as Call Forwarding Busy Line (CFBL), which is described in [GR-586-CORE].
3. Call Forwarding No Reply (CFNR) permits a served user to have the network send all incoming calls for the served user's number that meet No Reply to another number. The served user's originating service is unaffected. This is also known as Call Forwarding Don't Answer (CFDA) which is described in [GR-586-CORE].

5.2.6.1 All IP Implementation

- User A sends a SIP INVITE request toward user B.
- Call forwarding may be invoked:
    - At the terminating AS (due to CFU);
    - At the UE (due to user interaction);
    - At the terminating AS (due to timeout or busy).

5.2.6.2 PSTN-IP Interworking

When a call terminates at a PSTN station and encounters a CF service, a new IAM is sent to the forwarded-to destination. A call may be forwarded multiple times and the IAM conveys information about the first and last instances of forwarding in addition to a redirection counter.

Similarly, CF is handled in an IP environment via a SIP 302 - Moved Temporarily, and a new INVITE to the forwarding destination.

> NOTE: the forwarding could also be accomplished by having a proxy server rewrite the Request URI and forwarding the INVITE to a Gateway, similar to the example in [RFC 5359].

SS7 to SIP Interworking Scenario

Scenario: Party A dials the Original Called Number (party B), which resides in the PSTN and encounters forwarding. The "forward to" number (party C) resides in an IP network.

- When the call attempts to terminate to B and encounters CFV, a new IAM is generated by the switch serving B containing the Original Redirecting Reason (unconditional) and the Redirection Counter (the number of times the call has been forwarded). The Called Party Number (CdPN) is reset to the "forward to" number.
- The ISUP Gateway will map the IAM parameters to the SIP header fields as well as derive the Hop Counter from Max-Forwards, according to [ATIS-1000679]. The resulting INVITE is sent to party C.

SIP to SS7 Interworking Scenario

Scenario: Party A sends a SIP INVITE to party B, and encounters forwarding to a party C which resides in the PSTN.

- Given the forwarding is active at B, the response to the INVITE will be a SIP 302 Moved Temporarily, which includes instructions to retry the request to a new address (party C) given in the Contact header field.
- The SIP server sends a new SIP INVITE to party C.
- Since party C resides in the PSTN, the ISUP Gateway (or I-IWU) will map the INVITE header fields to an IAM according to [ATIS-1000679] to attempt call setup to party C.

5.2.6.3 Identified Gaps

None identified.

[ATIS-1000679] includes the mappings between all of this information and the corresponding SIP History-Info header defined in [RFC 4244] and the Diversion header defined in [RFC 5806]. Therefore, all of the information needed to support this service is carried transparently between the PSTN and the NGN.

5.2.7 950+ Call

The stage 1 service description of 950+ calling in [ATIS-1000611] notes that

> This service is provided by one network for another network and is not visible to the end user. This service is the transfer from the originating network to a transit network of the information that the 950+ call originated from either a local exchange carrier public phone, a hotel or motel line, or a non-equal access end office (non-EAEO).

The information is carried in the Nature of Address Indicator field in the Called Party Number parameter. In particular, the usual value of 000 0001 (subscriber number) is replaced by the value 111 0110 (950+ call from local exchange carrier public phone, hotel or motel line, or non-EAEO). Thus, although the Nature of address indicator value is in the Called Party Number parameter, this value reflects the calling party, not the called party.

The service only applies to 7-digit Feature Group B dialing. The Called Party Number is of the form 950-XXXX, where XXXX identifies the interexchange carrier where the call will be delivered for further processing (e.g., verifying billing information and obtaining a final destination for the call.)

5.2.7.1 NGN Implications

Support for Feature Group B dialing is increasingly rarely used in the PSTN and is not expected to be supported in the NGN. Therefore, [ATIS-1000679] does not include a mapping for this Nature of Address Indicator field in the Called Party Address parameter. Specifically, the Nature of Address Indicator field in a

received SS7 Initial Address Message is not conveyed in the SIP INVITE and when the SIP INVITE is received this Nature of Address Indicator field is populated to "follow the current ISUP standards." Since there is no indication in the SIP INVITE, the field should typically be populated with the value of 000 0001 (subscriber number).

Based on this observation, this TR concludes that there is no need to develop an evolution plan to migrate this service from the PSTN through the mixed PSTN/NGN environment and on to the IMS-based NGN environment.

5.2.8 Assist

[ATIS-1000611] defines the Assist service in the PSTN as allowing:

> A serving exchange without the required capabilities to provide certain call processing functions to the user by requesting the assistance of another exchange with the required capabilities.

In the example service in [ATIS-1000611], a switch requiring Play Announcement & Collect Digits service functionality is instructed by Intelligent Network service logic to set up a call to a different switch with the capability. The second switch needs to access the same service logic as the first to determine how to proceed. The correlating information is passed from the first switch to the second in the ISUP Transaction Request parameter.

There is no reason for the PSTN switch to invoke the Assist service and forward the call into the NGN; the Assist service assumes the second switch is a PSTN Intelligent Network switch and that the service will remain entirely within the PSTN. Therefore, [ATIS-1000679] does not provide a mapping for the ISUP Transaction Request parameter. Therefore, if the forwarded call were to reach an [ATIS-1000679] IWU, the Assist service would be terminated and the call setup would continue according to normal call setup procedures.

5.2.9 Carrier Selection

[ATIS-1000611] defines the Carrier Selection service in the PSTN as:

> The transfer from one network to another of the reason for the selection of the carrier identification code used for delivering a call to a transit network.

At the O-IWU (mapping ISUP to SIP) [ATIS-1000679] provides for transmission of the ISUP Carrier Selection Information (CSI) parameter in encapsulated ISUP (SIP-I).

Note that in the PSTN the CSI parameter is used in association with the Carrier Identification parameter, which indicates the transit network selected by the originating subscriber. The Carrier Identification parameter and CSI are not used for routing the call, but only for transmitting the identity of the selected network and the reason for the selection of the network. The corresponding routing information is carried as far as the selected network in the ISUP Transit Network Selection (TNS) parameter. The Carrier Identification parameter and the TNS parameter are each populated with a Carrier Identification Code (CIC), typically the same CIC.

The [ATIS-1000679] IWF maps the CIC from the Carrier Identification parameter into the "cic=" field of the Userinfo component of the SIP Request-URI and vice versa.

At the O-IWU (mapping ISUP to SIP) The CIC from the TNS is mapped into a corresponding value for the domain of the SIP Request-URI. At the I-IWU (mapping SIP to ISUP) If the host portion of the Request-URI includes a domain name that is not that of the I-IWU and is not that of the SS7 network at the I-IWU, then the I-IWU will populate the Transit Network Selection parameter with the CIC corresponding to that domain name.

The O-IWU does not map the CSI parameter into SIP. If SIP-I is not used to carry the CSI information through the NGN the information is not provided to the NGN.

At the I-IWU SIP-I information may be used to construct the CSI parameter if the PSTN needs the CSI parameter along with the Carrier Identification parameter. If SIP-I is not received at the I-IWU, the CSI information is not received and the I-IWU codes the ISUP CSI parameter, if needed, as with a default value of "selected carrier identification presubscribed and not input by calling party".

Since [ATIS-1000679] defines interworking of the CSI parameter, this TR concludes that there is no need to develop any further evolution plan to migrate this service from the PSTN through the mixed PSTN/NGN environment and on to the IMS-based NGN environment.

5.2.10 Three Way Calling (TWC)

Three Way Calling is a service that allows a customer engaged in a stable two-way call to add a third party to the conversation without operator assistance. TWC is described in [GR-577-CORE].

5.2.10.1 All IP Implementation

- User B sends a SIP INVITE to user A requesting a call session.
- A sends a SIP 200 OK indicating the connection is made and B sends a SIP ACK.
- To add party C, B places A on hold via the a *sendonly* SDP.
- B sends a SIP INVITE to user C.
- C responds with the SIP 200 OK, thus establishing a two-way connection between B and C.
- B takes A off hold by sending a SIP INVITE with an SDP attribute parameter *sendrecv*.
- B acts as the focus or the bridge mixing the channel between A and B, plus the channel between B and C.

[RFC 4353] provides informational outlines for different scenarios for implementing a conference and the role of the focus.

5.2.10.2 PSTN-IP Interworking

The success of the TWC feature depends on the ability of the TWC customer to perform the "bridging" function, whether the customer resides in the PSTN or the IP network.

For TWC customers in the PSTN, the capability already exists and, as long as the customer can establish a two way connection with any given SIP user, at any time, the customer will be able to create a three way bridge, as needed.

For SIP users attempting to create a three way connection, the capabilities may reside in their SIP phone, or the nearest proxy. SIP allows for the user to place existing sessions on hold (via the SDP *sendonly* attribute). That is the first step in setting a TWC. The TWC customer can bridge or mix the two channels once the second leg/channel is established with the third party (a PSTN user). In SIP, this role is referred to as the focus. The focus is the centralized manager of the conference that maintains the SIP signaling relationship with each participant.

5.2.10.3 Identified Gaps

None identified.

The TWC feature described above is limited to phone and voice. This feature does not offer mixing with any other media.

Since the "bridging" happens at the node where the service is invoked, there does not appear to be a need for an interworking solution, therefore no other gaps are identified at this point.

5.2.11 Call Waiting (CW)/ Calling Identity Delivery on Call Waiting (CIDCW)

PSTN Call Waiting is described in [GR-219-CORE] as a feature that informs the customer who is already on a call that another call is being made to the line[9]. The alert is in the form of a special tone. The customer may choose to answer the new call by placing the original call on hold.

CIDCW works in place of Call Waiting to allow calling identity information – normally sent in the on-hook state – to be delivered in the off-hook state. While on an existing call, a CIDCW customer can receive the identity of the calling party (number and name) on the waited call. The transmission occurs almost immediately after the

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[9] This is distinct from Internet Call Waiting, which provides the user with the name and number of a caller in a pop-up window, while they are engaged in an Internet session.

customer is alerted to the new call so they can decide whether to take the new call. CIDCW requires a special CPE that interacts with the end office to deliver the data in the off-hook state. The CIDCW alerting utilizes different alerting (different frequencies from the traditional CW tone) to communicate with the CPE.

CIDCW is described in [GR-575-CORE].

5.2.11.1 All-IP Network Implementation

CW is a service that is implemented on single-line telephones to give them some of the benefits of a second line. Since there is no concept of "lines" in a SIP network, there may not be an equivalent or a need for CW. However, the functionality could be replicated via a "hold" capability whereby the SIP user (B) places the original party (A) on hold, then establishes a new connection with the second caller (C), and back.

- B could place the first party (A) on hold, by sending a new INVITE with SDP: a=*sendonly.*
- B sends a SIP INVITE to establish a 2-way connection with the second caller (C).
- To tear the connection down, B sends a SIP BYE to C.
- B takes A off hold and re-establishes the call by sending a SIP INVITE to A with the SDP parameter *sendrecv.*

CIDCW is not expected to need any special CPE signaling in the IP network, since the special CPE signaling that facilitates Caller ID delivery described in [GR-575-CORE] is intended for special, non-SIP, non-IP devices. The caller ID for the new incoming call is expected to be delivered with the new INVITE. Although there are no internetworking issues, there might be some potential UNI issues for VoIP customers where CIDCW alerts need to be transmitted through the network adapters in order for the CIDCW CPE to function properly.

Guidelines would be needed for VoIP offerings where the network adapter is responsible for emulating CIDCW signals and interacting with and delivering the Caller ID to the CIDCW CPE.

5.2.11.2 PSTN-IP Interworking

The feature does not require any internetwork signaling to inform the customer that another call is being made to the line. Similarly, the feature does not require any internetwork signaling to place the original call on hold. Therefore, there are no anticipated interworking issues.

5.2.11.3 Identified Gaps

None identified.

Call Waiting is expected to function properly, as long as the SIP users have the capability to place a call/session on hold.

5.2.12 Screening List Editing (SLE)

[GR-220-CORE] defines Screening List Editing as:

> A set of supportive procedures that allow customers to activate and deactivate features, and create and modify lists of DNs. Each list is associated with a certain feature, such as Selective Call Acceptance/Rejection or Selective Call Forwarding, to identify calls that should receive special treatment.

In the PSTN, the end office maintains the screening lists as part of the subscriber's profile and applies the necessary treatment on each call terminating to the subscriber's DN. The services enabled by SLE are:

- Selective Call Acceptance (SCA) is an incoming service that allows customers to accept call attempts from a limited number of calling lines (only the DNs matching the SCA list). SCA is described in TA-TSY-001034.
- Selective Call Rejection (SCR) is an incoming service that prevents termination of call attempts from specified DNs (those on the SCR list). Instead, those calls are routed to special announcements. SCR is described in [GR-217-CORE].

- Selective Call Forwarding (SCF) is a service that forwards incoming calls from DNs on the SCF screening list to a predetermined forward-to station. SCF is described in [GR-218-CORE].

Visual SLE, defined in [GR-1436-CORE], provides similar functionality through a visual interface.

5.2.12.1   All-IP Network Implementation

SLE may be implemented in the SIP user's device or serving proxy. The list of DNs and the special announcements could be programmed in a SIP phone or a PC, etc.

5.2.12.2   PSTN-IP Interworking

SLE does not require interworking signaling because the editing of the lists is an interaction between the CPE and the terminating PSTN switch or NGN Application Server (AS).

5.2.12.3   Identified Gaps

None identified.

5.2.13 Call Screening Service (CS)

The Call Screening service was intended to permit a customer to monitor and intercept an active call that has been forwarded from the customer's line to a remote party (e.g., voicemail). When CS is active, if a call is forwarded by one of the available Call Forwarding features, the call is available to be monitored. To monitor the forwarded call, the customer simply goes off-hook and can then listen, via a one-way listening-only connection, to the communication between the calling party and the forwarded-to-party. The customer may choose to end monitoring by going on-hook or pressing the switch hook-flash. When a call is intercepted, a three-way call is established between the calling party, the CW customer and the remote party.

Call Screening is described in [GR-1512-CORE].

5.2.13.1   Identified Gaps

None identified.

CLASS^{SM} Call Screening is a terminating feature; therefore, no interworking signaling capabilities are needed.

5.3 Operator Services Related Services

5.3.1 Busy Line Verification (BLV) & Busy Line Verification and Interrupt (BLV/I)
5.3.1.1 PSTN Functionality

Busy Line Verification service in the PSTN is an operator-mediated service that provides information to the calling party regarding the status of a target number. The typical use case involves a calling party that has been unable to reach the called party, receiving a busy signal over a period of time. The calling party contacts the operator, who invokes BLV, setting up a call toward the target number over a specialized trunk group (Terminating Inward Service trunks).

At the terminating switch, incoming calls on the Terminating Inward Service group that complete to an idle called number do so normally. On receiving ringing, the operator may bridge the calling party on the call or may simply verify to the calling party that the line has been verified with no trouble found.

If the incoming call encounters a busy condition on the terminating line, the terminating switch bridges the incoming call onto the existing call (if possible). The operator hears a scrambled version of the content of the existing call. Therefore, the operator can inform the calling party that the line has been verified as being busy with an existing call (if the scrambled content appears to be conversation) or that some other condition exists (e.g., that the line appears to be busy with a fax call or there is no sound on the line).

If the incoming call encounters a network error condition or the scrambled content indicates noise or trouble on the line, the trouble may be reported to the network and to the calling party.

As a follow-on service to BLV, if the line is verified as being busy with an existing conversation, the operator may invoke the Interrupt service (with the sequence of services being referred to as Busy Line Verification and Interrupt or BLV/I). The conversation-scrambling function is removed from the call, the parties to the conversation are alerted (by a tone) that an operator has joined the call, and the operator alerts the participants in the conversation that there is a calling party trying to contact the called party and optionally providing a short message. If the called party indicates a willingness to end the existing conversation, this information may be passed to the calling party.

5.3.1.2 NGN Implications

There are four main reasons why BLV and Interrupt are not expected to be supported in the IMS-based NGN:

a) Philosophically, there is an issue surrounding the definition of a 'busy' line in the NGN. SIP-based user equipment typically has the ability to accept multiple simultaneous calls. Unlike the PSTN case, the existence of one communication stream terminating at the user equipment does not necessarily make the user equipment 'busy' (i.e., unable to accept another call). Therefore, the question of what is being 'verified' becomes more difficult than in the PSTN.

b) In the PSTN, BLV and Interrupt rely on specialized Terminating Inward Service trunking to invoke custom call processing in the target switch. In particular, calls on ordinary trunks do not bridge onto existing conversations. BLV and Interrupt require that the target switch bridge the incoming call from the operator/BLV system onto any existing call on the target line (so that the operator system may provide the operator with a scrambled version of the ongoing conversation). There is no corresponding capability in IMS to force the user equipment to accept the call from the operator/BLV system, nor is there a capability to force the call (if it were accepted) to be a three-way call. There is an additional difficulty in IMS in selecting which bearer stream to attempt to connect to, in the event that the user equipment has multiple bear streams in use.

c) In the PSTN, the status of a line is verified by having the human operator listen to a scrambled version of the activity on the line being verified. Possible line states that are expected to be able to be distinguished include:

Table 5. 2 - PSTN BLV Conditions Encountered and Operator Interpretation

Condition Encountered	Interpretation
Scrambled conversation is heard	Conversation on the line
Background noise is heard	No conversation on the line
Tone, ringing, or no sound is heard	No conversation on the line
Busy signal or reorder signal is heard	No conversation on the line
Data tone, computer, fax, etc. is heard	Data Transmission on the line
Static, clicking, humming sound, roaring sound, etc., or trouble announcement recording is heard	Trouble on the line
Vacant Code Announcement display or recording is received at the operator workstation	The number cannot be verified

It is not clear how all of the conditions encountered in Table 5.2 would manifest themselves in an IP environment and whether IP transport network problems would manifest as static, clicking, etc. or would manifest as no sound heard. Therefore, at least some of these conditions may not map easily into the NGN.

In addition, SIP signaling in the NGN provides several different indications that a destination number is "busy", including:

486   Busy Here

480   Temporarily Unavailable

403   Forbidden

488   Not Acceptable Here

While the SIP 486 code can be mapped to the 'busy' status of BLV in the PSTN, it is not clear how the other codes should be interpreted in the context of BLV. For example, should 480 Temporarily Unavailable be taken to mean that the status of the terminating end cannot be determined or that there is an equipment trouble preventing completion of the call? In addition, determination of 'Trouble' in the PSTN is assumed to imply a network error condition; in the NGN it may be difficult to determine from the signaling where the error exists.

In the PSTN, a significant fraction of BLV/I attempts do not succeed. Terminating Inward Service trunks may not be available on an internetwork call (especially if the target number is in a cellular network), not all end switches support BLV service, and end user services such as Call Forward on Busy may interfere with the capability to Interrupt an existing conversation. Diminishing user expectations regarding the level of support in the PSTN may diminish the value in attempting to provide a similar service in the NGN.

It should also be noted that some user needs that are met by BLV are addressed by other functionality in the PSTN and in the NGN. Call Waiting service alerts a busy called party of an additional incoming call attempt. Alternative near-real time services (Instant Messaging or email) may be employed to attempt to establish contact with a called party whose telephone service appears to be consistently busy. The IMS Presence service provides a mechanism for determining whether the target party is potentially available to accept new communication requests.

Based on the technical and practical challenges to providing BLV/I service in the NGN and the availability of alternatives to address at least a portion of the functionality, this TR concludes that there is no need to develop an evolution plan to migrate this service from the PSTN through the mixed PSTN/NGN environment and on to the IMS-based NGN environment.

5.3.2 User-Network Interaction

The User-Network Interaction network capability is defined in [ATIS-1000611] as allowing:

A network to temporarily halt call set-up and play tones or announcements, or both, and collect additional in-band information from the calling user before routing the call further.

Specifically, this network capability allows a network to establish a two-way talking path before Answer. The assumption is that the call has not been set up to its destination and that the network requires the ability to play tones or announcements in the backward direction and to collect additional in-band information from the calling user before routing the call further.

In the SS7 network, the functionality is mediated via the User-Network Interaction Indicator in the Optional Backward Call Indicators parameter in the ISUP Address Complete Message.

The mapping of the Optional backward call indicators parameter with the User-Network interaction indicator set to "user-network interaction occurs, cut through in both directions" to the corresponding Early Media functionality in the NGN is described in [ATIS-1000679].

Therefore, this TR concludes that there is no need to develop an evolution plan to migrate this network capability from the PSTN through the mixed PSTN/NGN environment and on to the IMS-based NGN environment.

5.3.3 Operator Assistance and Directory Assistance

5.3.3.1 Service Description

[ATIS-1000036] defines the Operator Assistance service in the NGN as follows:

> Dialing instructions and rate information are provided to the calling party by an OSS[10].

[ATIS-1000036] also defines the Directory Assistance in the NGN service as follows:

> The telephone number for a given name and address is provided to the calling party by an OSS. Call completion to the telephone number is an option available to the calling party. In the PSTN, Operator Recall may be signaled by depressing the flash-hook.

In each of these services in the PSTN, an ordinary call is set up to the Operator Services System, and the service is entirely managed by the OSS Application Server. The OSS may need to monitor the MGCF for the # signal that is used in some PSTN implementations by the calling party to request Operator Recall (e.g., after Call Completion has been invoked).

Since these services are managed entirely by the Operator Services switch in the PSTN and in the OSS Application Server in the NGN, they do not introduce any new requirements at the PSTN-NGN interface beyond those of normal call setup. Therefore, this TR concludes that there is no need to develop an evolution plan to migrate these services from the PSTN through the mixed PSTN/NGN environment and on to the IMS-based NGN environment.

The services may also include database queries to obtain needed information (e.g., rate information for Operator Assistance or the telephone number for Directory Assistance).

5.3.3.2 Identified Gaps

As noted in the previous section, some PSTN implementations use the hook-flash signal rather than the # signal for reconnection to the operator. The hook-flash is signaled inband and thus cannot be detected by the MGCF. Therefore, the PSTN Operator Recall feature is not supported in the NGN; a corresponding feature could be supported by changing the end user signaling, e.g., by using the # signal.

In the PSTN, Directory Assistance is implemented with a query to the Listing Services Database (LSDB) described in [GR-1157-CORE] to map the given name and address to the desired telephone number. Interworking guidelines would need to be developed.

5.3.4 Call Completion

[ATIS-1000036] defines the Call Completion service in the NGN as follows:

> The OSS completes a call for the calling party. The called number may be determined by an operator service (e.g., Directory Assistance), or by the calling party alone (by providing the called number to the OSS.) This service includes an optional feature (Operator Recall) to return the calling party to an operator when signaled by the calling party.

In the PSTN, Operator Recall may be signaled by depressing the flash-hook or the #.

5.3.4.1 PSTN-IP Interworking

For Call Completion service originating in the PSTN but served in the NGN, an ordinary call is set up to the NGN Operator Services System, and the service is entirely managed by the OSS Application Server. The NGN OSS may need to monitor the RTP bearer for the # signal[11] that is used in some PSTN implementations by the calling party to request Operator Recall. Note that the flash-hook signal is not reliably transmitted from

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[10] Operator Services System, in [ATIS-1000036].

[11] Alternatively, if NGN implementations use Key Press Markup Language (KPML) as described in [RFC 4730], the NGN OSS would be able to send a SUBSCRIBE request to the calling party in order to receive a NOTIFY request when the end user signals the #. Interworking between the inband PSTN signal and the NGN has not been standardized.

the PSTN to the NGN, nor is it mapped by KPML. Therefore, this signal is assumed to be lost at the PSTN-NGN boundary.

Since these services are managed entirely by the Operator Services switch in the PSTN or in the OSS Application Server in the NGN, they do not introduce any new requirements at the PSTN-NGN interface beyond those of normal call setup. Therefore, this TR concludes that there is no need to develop an evolution plan to migrate these services from the PSTN through the mixed PSTN/NGN environment and on to the IMS-based NGN environment.

5.3.4.2 Identified Gaps

As noted in the previous section, Operator Recall may be signaled by depressing the flash-hook or #. The flash-hook is not passed from the PSTN into the NGN; therefore, the PSTN Operator Recall feature using the flash-hook is not supported in the NGN. A corresponding feature could be supported by changing the end user signaling, e.g., by using the # signal and defining interworking at the [ATIS-1000679] IWF, but this could imply changes to some PSTN service implementations.

5.3.5 Person Handling (Person-to-Person Call)

[ATIS-1000036] defines the OSS function of a Person-to-Person Call as follows:

> The OSS completes a call for the calling party. In addition, before the calling and called parties are connected, the OSS confirms that a specific individual has been contacted, rather than just the requested end user equipment.

[ATIS-100036] notes that NGN call processing is similar to Directory Assistance with Call Completion, in that an ordinary call is set up to the OSS, an ordinary call is set up from the OSS to the called party, and connecting the calling and called parties is entirely managed by the OSS AS. This is similar to the PSTN functionality, where the signaling and the bearer transit the PSTN OSS for the duration of the call.

Note that in the NGN the bearer may be redirected to the new destination while retaining signaling through the Application Server. This does not, however, introduce any unusual requirements in the NGN, nor any new requirements at the PSTN-NGN interface beyond those of bearer redirection for any other service. Therefore this TR concludes that there is no need to develop an evolution plan to migrate this network capability from the PSTN through the mixed PSTN/NGN environment and on to the IMS-based NGN environment.

5.3.6 Operator Intercept Service

5.3.6.1 PSTN Service

The PSTN Intercept Call Request is defined in [ATIS-1000666] as:

> An operator services call request initiated by the network because of the service condition of the line of the called party (e.g., line out of service, etc.).

A call that is set up to an intercepted number is routed as an ordinary call to the destination (intercepting) switch. At this point a simple announcement may be played (indicating the number is out of service or has been changed). In this case, the call is then terminated.

Alternatively, the Intercepting switch may redirect the call to an Operator Services Switch to provide the announcement and any potential additional services (e.g., Call Completion). Two ISUP protocol mechanisms have been standardized for routing these calls to the Operator Services Switch in the PSTN. They are distinguished by the information populated in the SS7 Initial Address Message, Called Party Number parameter, Nature of Address (NOA) field. In both mechanisms, the intercepting switch populates the Called Party Number in the Initial Address Message with the intercepted number.

In the more widely implemented 'Basic NOA Field' mechanism detailed in [GR-1277-CORE], section 3.4.1, the intercepting switch routes the IAM as received (with the [ATIS-1000113]-standard Nature of Address (NOA) field populated as received – usually 'national (significant) number') directly to the Operator Services Switch that will provide the Intercept service (i.e., the IAM does not transit any tandem switch that would read the Called Party Number parameter and route the call back to the intercepting switch). The Operator Services Switch never acts as a tandem switch; therefore, it can safely assume that the Called Party Number contains

information about a call to a number that requires Operator Services handling. The Operator Services Switch determines that the Called Party Number is on its list of Intercepted numbers and provides the service.

In the 'Modified NOA Field' mechanism detailed in [GR-1277-CORE] section 3.4.2, the [ATIS-1000113]-standard NOA field is populated "national number, operator requested" and intermediate switches are expected to route these calls toward the Operator Services Switch rather than toward the switch associated with the digits in the Called Party Number parameter.

In addition to routing the call to the Operator Services Switch, the Intercepting switch populates the Service Activation Parameter/Feature Code indicator field with a value indicating the reason for intercepting the call as shown in Table 5.3:

Table 5. 3 - Regular Operator Intercept Feature Codes

Feature Code	Meaning	Interpretation
10010000	Intercept – Regular	Number has been changed
1001 0110	Intercept – Blank	Number is not assigned
1001 0111	Intercept – Trouble	Network trouble reaching this number

5.3.6.2 NGN Service

[ATIS-1000050] describes Operator Regular Intercept service as follows:

> When a call termination attempt is addressed to such a destination (the 'intercepted number'), then Operator Regular Intercept service indicates (e.g., by playing a verbal announcement) the reason the call cannot be completed. Additional information may also be provided to the caller – e.g., providing a new destination address.

The service description in [ATIS-1000050] only addresses Regular Intercept (where the called number has changed). Assuming that there will be a 'default' Application Server designated to handle calls to numbers that have never been assigned to a customer (or which have been assigned but any Regular Intercept service has ended and the number is, therefore, not assigned to any customer), Blank Intercept service should function similarly. The informative call flows for Regular Intercept processing that are given in [ATIS-100050] show the Intercepted Number Application Server sending a 301 Moved Permanently response to redirect the call/session to the Application Server that will provide the Regular Intercept treatment.

Trouble Intercept is not expected to be supported in the NGN. There is no standard NGN requirement that the network be able to test session setup functionality. In addition, there is significant ambiguity surrounding the question of what would correspond to the PSTN 'network trouble', since the SIP session reaches past the UNI to the User Equipment. Therefore, the NGN is not expected to support Trouble Intercept service (i.e., is not expected to provide a Trouble announcement to the caller).

5.3.6.3 PSTN to NGN Interworking

Because of the differences in the service functionality between the PSTN and the NGN, it is necessary to consider the two interworking scenarios separately. This section considers the use case where the intercepted number is in the PSTN and the service is provided in the NGN. This section addresses the use case where the intercepted number is in the NGN and the service is provided in the PSTN. The use case where the intercepted number is in the PSTN and the service is provided in the NGN is addressed in section 5.3.6.4.

5.3.6.3.1 Basic NOA Field

As noted above, the Basic NOA Field mechanism requires the Intercepting switch to route the call directly to the Operator Services Switch. No intermediate routing (including routing through an [ATIS-1000679] Interworking Function is allowed. (If the IAM using the Basic NOA Field mechanism were to be routed to an [ATIS-1000679] Interworking Function, one of two events would follow:

- The O-IWU might fail the call at this point, as the O-IWU recognizes that the Called Party Number is in the PSTN network that just routed the call to the PSTN Gateway, or

- The O-IWU might route the call back into the PSTN network to the intercepting switch, which would once again provide the Intercept service and complete the circular route back to the PSTN Gateway.

Therefore, this use case requires that interworking be avoided. Separate PSTN and NGN Intercept functionality would need to be supported for the duration of transition until the PSTN functionality is no longer needed. Note that separate PSTN and NGN Intercept functionality also circumvents the need for implementation of signaling to the NGN Intercept Application Server related to Trouble Intercept calls[12].

5.3.6.3.2 Modified NOA Field

[ATIS-1000679] does not consider the option of a Called Party Number in the Initial Address Message with the NOA field populated "national number, operator requested" and "international number" (modified NOA). Therefore, behavior of the Interworking Function is not defined. In this use case, if the [ATIS-1000679] Outgoing (from ISUP) Interworking Unit (O-IWU) were to ignore the NOA field (i.e., treat the Called Party Number as a national (significant) number), one of two events would follow:

- The O-IWU might fail the call at this point, as the O-IWU recognizes that the Called Party Number is in the PSTN network that just routed the call to the PSTN Gateway, or

- The O-IWU might route the call back into the PSTN network to the intercepting switch, which would once again provide the Intercept service and complete the circular route back to the PSTN Gateway.

If the SS7-side processing of the PSTN Gateway, where the Interworking Function resides, were to process the NOA field and derive some NGN-operator-services-related Called Party Number for the Interworking Function to use to route the call/session to the Intercept Service in the NGN, it would also need to populate an Original Called Party Number parameter with the intercepted number for processing in the NGN. The resulting call into the Interworking Function would appear to have been subject to call forwarding. This is well beyond the usual functionality of a PSTN Gateway and might be better modeled as a unique expansion of the Gateway or as a PSTN analog to the NGN Application Network Interface. In addition, the forwarded call would need to identify the specific Intercept service being invoked, e.g., by routing each to its own address[13]. If separate addresses are used, Regular Intercept and Blank Intercept could share an address; when the Intercept Application Server look-up for the Intercepted Number does not return a Referral Number to provide to the calling party, the information given to the calling party is the same whether the number has been changed (Regular Intercept, but no longer announcing the Referral Number) or the number has never been assigned to a customer (Blank Intercept). However, for Trouble Intercept there is no need to attempt to obtain a Referral Number and the information provided to the calling party is characteristically different from Regular Intercept and Blank Intercept.

It is also possible to avoid the potential failure events without developing interworking functionality that is unique to the Intercept service. As with the Basic NOA Field use case, interworking may be avoided by maintaining separate PSTN and NGN Intercept functionality for the duration of transition until the PSTN functionality is no longer needed.

5.3.6.4 NGN to PSTN Interworking

This section addresses the use case where the intercepted number is in the NGN and the service is provided in the PSTN. When the informative call flows in [ATIS-1000050] are followed, the 301 Moved Permanently response should direct the call into the PSTN. However, the address in the 301 is the address of the entity that provides the service (the Operator Services System in the PSTN), not the called number[14].

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[12] Note that signaling related to Trouble Intercept calls in the PSTN may also be avoided by providing the Trouble Intercept announcement at the intercepting switch, while still forwarding Regular Intercept and Blank Intercept calls to the Operator Services Switch.

[13] Note that signaling related to Trouble Intercept calls in the PSTN may also be avoided by providing the Trouble Intercept announcement at the intercepting switch in the PSTN, while still forwarding Regular Intercept and Blank Intercept calls to the Intercept Application Server in the NGN.

[14] If the Intercept Service is provided in the NGN, the NGN could administratively maintain its routing tables to route calls/sessions directly to the Intercept Application Server based on the called number in the Request-URI. However, if that

[ATIS-100050] provides informative examples of routing an intercepted call/session in the NGN to the Intercept AS in the NGN with a SIP address of the form 'sip: intercept-regular@provider.com'. The original dialed number is carried in the Diversion and History-Info headers of the INVITE message. Note that also implies that text-based addressing is introduced into the NGN. If this mechanism is used and the Intercept service is provided in the PSTN, the [ATIS-1000679] NGN-PSTN Interworking Function will need to be extended to map from the text-based 'intercept-regular' address (and, potentially a separate 'intercept-blank' address) to an E.164-based address in the PSTN. Since Trouble Intercept is not expected to be supported in the NGN, there is no need for a mapping leading to an SS7 Service Activation Parameter/Feature Code of Intercept – Trouble (e.g., a mapping for an address of the form 'sip: intercept-trouble@provider.com'). Thus, enhancements to existing implementations are focused on the [ATIS-1000679] IWF with this mechanism.

Alternatively, the address in the 301 Moved Permanently header could be E.164-based[15]. [ATIS-1000679] specifies that a 3xx header is handled according to the SIP protocol, resulting in invocation of the local routing function. Presumably this would result in an INVITE that includes the original dialed number in the Diversion and History-Info headers. The [ATIS-1000679] IWF would then map this into an SS7 Initial Address Message. This mechanism would require assignment of separate E.164 numbers to the PSTN Operator Services System and changes to the Operator Services System to provide the service as if the incoming calls had been forwarded from a PSTN switch rather than using either of the standard Operator Intercept protocols. Thus, enhancements to existing implementations are focused on the PSTN Operator Services System with this mechanism.

As with the other use cases for Regular Intercept service and Blank Intercept service, interworking may be avoided entirely by maintaining separate PSTN and NGN Intercept functionality for the duration of transition until the PSTN functionality is no longer needed.

5.3.6.5 Identified Gaps

Because intercepted numbers in the PSTN lead to Initial Address Messages using the Basic NOA or Modified NOA, and because PSTN Operator Services Switches assume one of these mechanisms is used, interworking for the case where the intercepted number is in one domain (PSTN or NGN) and the service is provided in the other is complicated.

This TR identifies the following options for transition of Operator Intercept service:

1. Maintain separate PSTN and NGN Intercept functionality for the duration of transition until the PSTN functionality is no longer needed, avoiding interworking.
2. Extend the [ATIS-1000679] IWF, possibly at a limited number of gateways designated as interworking points for Operator Intercept calls, to accept Basic NOA and Modified NOA signaling on the PSTN side and calls that are effectively forwarded (e.g., via the 301 Moved Permanently header) on the NGN side.
3. Replace Operator Intercept service in the PSTN with a service based on forwarding the call from the intercepting switch to the PSTN Operator Services switch, assuring that the protocol used will interwork successfully through an existing [ATIS-1000679] IWF to the protocols that are eventually implemented in the NGN. Note that this TR does not assume that changes to PSTN service functionality is feasible.

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routing leads to the PSTN Gateway, ordinary [ATIS-1000679] mapping of the address would either fail the call or start a circular route as noted for the Modified NOA Field case of PSTN to NGN interworking.

[15] Note that assigning an E.164 number to the NGN Operator Services AS would avoid the need to support the text-based routing in the NGN in the informative examples in [ATIS-1000050].

6 Summary

The following Tables summarize the analysis performed in this TR on the expected transition gaps for each service. The tables also identify support available for implementing those services in the NGN – at the time of publishing this TR – as evidenced by the availability of standards, IETF RFCs and known implementations. In addition, the tables note the status of some current (early) implementations and the potential for long term support for these services in the NGN.

Note that issues surrounding the integrity and utility of the original calling party number – that many of these features rely on – are not the result of PSTN to IP transition, per se. Therefore, those issues are not listed as gaps.

Table 6. 1 - Database Services

Service	Any Gaps identified?	What capabilities/ tools exist to implement this service in the NGN?	Is this service offered in early NGN implementations today?	Long term outlook[16] for this service or equivalent functionality in the NGN
Emergency Telecommunication Service (ETS)	No	Existing and emerging ANSI standards and implementations	Yes	ETS is expected to be supported in the NGN.
Toll Free Calling	No	ENUM	Yes	Toll free calling is expected to be supported in the NGN.
Emergency Calling	No	Existing and emerging ANSI standards and implementations	Yes	Emergency calling is expected to be supported in the NGN.
LIDB Data Services (e.g., Verification)	No	Hub and Service Transactions Gateway Function	Yes	These services are expected to continue supporting identity management and fraud prevention applications for telecom and non-telecom services in the NGN.

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[16] This column offers preliminary views of the potential for each service – in the NGN environment – based on known end user needs, behaviors, and upcoming technologies that may supplant existing services.

Service	Any Gaps identified?	What capabilities/ tools exist to implement this service in the NGN?	Is this service offered in early NGN implementations today?	Long term outlook[16] for this service or equivalent functionality in the NGN
Calling Name	No	SIP INVITE or IP-based queries (LDAP, SIP, XML, etc.) to the Name database	Yes	End users familiar with PSTN Calling Name service are likely to expect and continue utilizing it in the NGN. Device-based name displays that are based on the contact list stored in the device are limited compared to network provided CNAM.

Table 6. 2 - Line-Based Services

Service	Any Gaps identified?	What capabilities/ tools exist to implement this service in the NGN?	Is this service offered in early NGN implementations today?	Long term outlook for this service or equivalent functionality in the NGN
950+ Calling	Only applies to FG B signaling	None	No	The corresponding functionality is not expected to be used in the NGN.
Anonymous Call Rejection (ACR)	No	Anonymity Disallowed as described in RFC 5079	Offered by several VOIP providers and designed in several IP vendor products.	End users familiar with PSTN ACR service are likely to expect and continue utilizing it in the NGN, given the number of customers not wishing to accept unidentified calls is rising.
Assist	Only applies to PSTN Intelligent Network services	None	No	PSTN Assist calls are not expected to be routed to NGN network elements.
Automatic Call Back/ Recall (AC/AR)	No gaps for IP implementation but TCAP-SIP mapping needs to be defined for interworking scenarios.	SUBSCRIBE-NOTIFY as described in RFC 6665 RFC 6910 Completion of Calls	Offered by several VOIP providers and designed in several IP vendor products.	End users familiar with the PSTN AC and AR services are likely to expect and continue utilizing them in the NGN, given the services offer a convenient, hands-free way to reach a busy party instead of physically redialing. In the long term, device-based capabilities may perform the same function.

Service	Any Gaps identified?	What capabilities/ tools exist to implement this service in the NGN?	Is this service offered in early NGN implementations today?	Long term outlook for this service or equivalent functionality in the NGN
Busy Line Verification (BLV) and Busy Line Verification and Interrupt (BLV/I)	Only applies in the PSTN	None	No	Technical and practical challenges suggest BLV and BLV/I will not be supported in the NGN.
Call Completion	No	Basic call setup	Yes	End users familiar with Call Completion (by the operator) are likely to expect and continue utilizing it in the NGN.
Call Forwarding	No	SIP History-Info header [RFC 4244] and SIP Diversion header [RFC 5806]	Offered by several VOIP providers and designed in several IP vendor products.	End users familiar with PSTN Call Forwarding service are likely to expect and continue utilizing it in the NGN.
Call Screening (CS)	No	"Bridging" capabilities similar to those used by Three Way Calling	No	CS faces competition from devices such as answering machines.
Call Waiting	No	The ability to place an entity on and off hold in SIP (via *sendonly* and *sendrecv*, respectively)	Offered by several VOIP providers and designed in several IP vendor products.	End users familiar with PSTN CW service are likely to expect and continue utilizing it in the NGN.
Calling Identity Delivery on Call Waiting (CIDCW)	No gaps, but possible UNI issues	RFCs 3261 and 3325 defining PAI and privacy extensions.	Yes	End users familiar with PSTN CIDCW service are likely to expect and continue utilizing it in the NGN. Customers will no longer need special CPE to receive the Caller ID on CW in the NGN, though.
Calling Number Delivery (CND)	No	RFCs 3261 and 3325 defining PAI and privacy extensions.	Yes	End users familiar with PSTN CND service are likely to expect and continue utilizing it in the NGN. Possible variations and enhancements to the NGN-based service could include providing the address, current location, or photo, etc., along with the calling number.

Service	Any Gaps identified?	What capabilities/ tools exist to implement this service in the NGN?	Is this service offered in early NGN implementations today?	Long term outlook for this service or equivalent functionality in the NGN
Carrier Selection	No, though PSTN Carrier Selection Information (CSI) parameter value is not mapped into the NGN	Generation of a default CSI value if needed when entering the PSTN	Yes	Carrier selection may not apply in an all-NGN environment.
Customer Originated Trace (COT)	No gaps for IP implementation but potential issues for interworking scenarios.	SIP extensions in RFC 5503	Offered by several VOIP providers and designed in several IP vendor products.	End users and law enforcement familiar with PSTN COT service are likely to expect and continue utilizing it in the NGN for similar purposes (tracing harassing or threatening calls).
Directory Assistance	No	Basic call setup	Yes	End users familiar with Directory Assistance are likely to expect and continue utilizing it in the NGN.
Message Waiting and Visual Message Waiting	No gaps for IP implementation but TCAP-SIP mapping needs to be defined for interworking scenarios.	MWI Event Package in RFC 3842	Offered by several VOIP providers and designed in several IP vendor products.	End users familiar with PSTN MWI and VMWI services are likely to expect and continue utilizing them in the NGN. Messaging and unified communications services will continue to require a notification mechanism (audio or visual).
Operator Assistance	No	Basic call setup	Yes	End users familiar with Operator Assistance are likely to expect and continue utilizing it in the NGN.
Operator Intercept	Yes Request for an operator system to provide service will not pass an [ATIS-1000679] IWF if called number and operator system are in different environments	Basic call setup	Unknown	End users familiar with Operator Intercept service are likely to expect and continue utilizing it in the NGN. Changing the end user's number may be less frequent in the future.

Service	Any Gaps identified?	What capabilities/ tools exist to implement this service in the NGN?	Is this service offered in early NGN implementations today?	Long term outlook for this service or equivalent functionality in the NGN
Person Handling (Person-to-Person Call)	No	Basic call setup	Yes	End users familiar with Person-to-Person Call service are likely to expect and continue utilizing it in the NGN.
Screening List Editing	No	None needed	Yes	End users familiar with PSTN SLE-based services are likely to expect and continue utilizing them in the NGN. The services and maintenance of the Screening List may continue to be implemented in the nearest proxy serving the end user or in an Application Server. NGN SLE-based services, their lists, and terminating treatments may also be managed in the end user's device.
Three Way Calling	No	"Bridging" is effectively performed by the "focus" or party in charge (Best practices are described in RFCs 4579 and 5359)	Offered by several VOIP providers and designed in several IP vendor products.	The ability to conference multiple voice calls or sessions (beyond simple 3-way calling) is likely to continue to be in demand for a long time.
User-Network Interaction	No	[ATIS-1000679] defines interworking between ISUP Optional backward call indicators parameter and NGN Early Media functionality	Yes	Early Media is expected to be used in the NGN.

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