ipdr.org

Network Data Management – Usage (NDM-U) For IP-Based Services

Service Specification – Voice Over IP (VoIP)

Version 2.5-A.0

April 13, 2001

Preface

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Abstract

This document is a companion to NDM-U, which specifies the overall business requirements and protocol generic to all services. The content herein is compliant to those requirements and specifications and is particular to the service specified.

Change History

2.5-A.0 April 13, 2001 – First Production Release

Table of Contents

1. Introduction

1.1. Purpose

This document is intended to specify the business use case and formal XML Schema for the IP-based voice telephony service.

1.2. Scope

This document is limited to the discussion of issues as defined by the mission statement of IPDR.org, namely:

The IPDR Organization (the "Organization") is organized and operates as a non-stock not for profit organization for the following purposes:

(a) To develop, agree upon and publish a non-proprietary, open specification for the representation and encapsulation of Internet Protocol (IP)-based events for use by business, operations and decision support systems. Such events include, but are not limited to, IP-based network services, application services and e-commerce transactions;
(b) To develop, agree upon and publish a non-proprietary, open specification for the representation and encapsulation of IP-based network and service elements provisioning events;
(c) To promote work accomplished and uniform specifications to the industry and submit approved published specifications to the appropriate standards bodies for acceptance in the public domain; and

To have and exercise all powers necessary or convenient to affect any or all of the purposes for which the Organization is organized.

1.3. Compatibility

Future revisions are expected to make every attempt to preserve investments made by service providers and solution vendors by considering backward and forward compatibility whenever it is practical.

1.4. References

[1] NDM-U 2.5, IPDR.org.

[2] XML Schema Part 1: Structures, W3C Working Draft 7 April 2000.

[3] XML Schema Part 2: Data Types, W3C Working Draft 7 April 2000.

1.5. Overview

This specification is divided into two major chapters:

- Service Specification – description of the specific requirements and business use case for the service in question.
- Formal Specification – XML Schema description of the IPDR Record for this service.

1.6. Terminology and Glossary

Terminology

Term	**Definition**
Accounting	The process of collecting and analyzing **service** and **resource usage** metrics for the purposes of capacity and trend analysis, cost allocation, auditing, and billing, etc. Accounting management requires that resource consumption be measured, rated, assigned, and communicated between appropriate business entities.
Mediation	In view of network reference model, Mediation refers to the combination of the logical entities IPDR recorder, IPDR transmitter, and IPDR store.
Resource	A quantifiable asset employed by a **Service Provider**, or on behalf of a **Service Provider** by another Service Provider, to fulfill a request of a **Service Consumer**. (Examples include: files, communications, goods, etc).
Roaming	Service usage initiated by a service consumer and provided by a service provider other than the one with which the service consumer have business relationship.
Service	Network and/or application operation that provides the **Service Consumer** with the requested **resource.**
Service Consumer	The beneficiary (human or system) of a **service**.
Service Element	Any element that is responsible for fulfilling a **Service Consumer** request. (Examples include: network equipment and system processes)
Service Provider	An enterprise that provides communications-based **services**.
Session	A set of related service usages; service usages may or may not be time based in the unit of measurement.
Usage	Consumption of **resources** and **services** by a **Service Consumer**.
Usage Attribute	A parameter whose value indicates some aspect of **usage** of a given **service** and/or **resource.**
Usage Entry[1]	A **Service**-specific trigger resulting in the generation by a **Service Element** of a set of **Usage Attribute** values related to **Usage** specific to a given **Service Consumer**

[1] Because of legacy issues, a Usage Entry from a given Service Element will not initially conform to an IPDR specification or, in some cases, may never conform. To be considered a Usage Entry the information presented or made available by inference from the Service Element must minimally contain attributes from some of the general attribute categories.

Glossary:

ANI - Automatic Number Identification
ASP - Application Service Provider
BSS - Business Support Systems
CCI - Call Clarity Index
CRM - Customer Relationship Management
DSS - Decision Support Systems
DTD - Document Type Definition
DSL - Digital Subscriber Line
EP - End Point
ESN - Electronic Serial Number
ETSI - European Telecommunications Standardization Institute
FoIP - Fax over IP
GK - Gate Keeper
GPRS - General Packet Radio Service
GSM - Global System for Mobile Communications
IETF - Internet Engineering Task Force
IMSI - International Mobile Subscriber Identity
IP - Internet Protocol
IS - IPDR Store
ISDN - Integrated Services Digital Network
ISO - International Standards Organization
ISP - Internet Service Provider
IT - IPDR Transmitter
ITU-T - International Telecommunications Union – Telecommunications Standardization Section
MOS - Mean Opinion Score
NDM - Network Data Management
NSE - Network Service Element
OSS - Operations Support System
PLMN - Public Land Mobile Network
PSTN - Public Switched Telephone Network
QoS - Quality of Service
RADIUS – Remote Access Dial-In Usage Server
RAS - Remote Access Server
SC - Service Consumer
SCN - Switched Communications Network
SE - Service Element
SIP - Session Initiation Protocol
SMS - Short Message Service
SP - Service Provider
TIPHON- Telecommunications and Internet Protocol Harmonization over Networks
TMF - TeleManagement Forum
TOM - Telecommunications Operations Map
UA - Usage Aggregators
UC - Usage Collectors
VoIP - Voice over IP
VPN - Virtual Private Network
WAP - Wireless Application Protocol
xDSL - Digital Subscriber Line of type x
XML - eXtensible Markup Language

2. VoIP Specification

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2.1. Definition

VoIP is voice (or voiceband) communications between two or more parties over a partial / complete Internet-based connection. The “call” is initiated by a calling party (call initiator) and received by one or more call parties (call recipients). The participating elements for the call include service elements, gatekeepers, and endpoints (end-users).

The transmission path of the call is realized at VoIP switch by a VoIP gatekeeper (GK) and at each customer location by an Endpoint (VoIP/PSTN). At customer locations, the user speaks and listens into a device that carries the voice signals. Alternatively, voiceband signals such as fax and modem signals can be transmitted.

The intent is for SEs to generate and transfer IPDR records to a BSS, which represent each voice call transparently between all SEs involved in the VoIP call. Additional information on the call such as QoS (Quality of Service) attributes can be measured by independent probes and provided to the mediation system and the BSS system. The per-call QoS rating information can be compared against customer’s subscribed QoS grade for Service Level Agreement (SLA) confirmation and possible billing adjustments. Generally, an IPDR is generated at the end of each call/connection that results in either normal or abnormal completion (end-of-call driven). Alternatively, an IPDR may be generated during the progress of call/connection in response to certain significant events such as call start, call answer, SCP/LNP dip, very long-duration, detected fraud, call termination, etc. (event driven).

2.1.1. Requirements

1. An IPDR must contain the identifiers of all call participants (calling party & called parties).
2. An IPDR must contain the time that the call was started and ended.
3. The call progress state (as defined in the use case) that describes the phases that each SE goes through in a call must be contained by an IPDR
4. An IPDR must contain final call completion codes for each call for billing purpose.
5. Times in IPDR should be expressed per ISO 8601 format for the purpose of facilitating data exchange. The specific time precision requirements vary with applications (e.g. IP packet time as opposed to billing time) and are individually specified in the attribute list. For billing purpose, time stamp accuracy should be 1 second or better. Local time zone offset with reference to GMT should be provided and should reflect local time of calling party for correct billing.
6. An IPDR must contain call payment type for the call (toll-free, charge to calling party, charge to called party, prepaid, charged to third party etc.).

2.1.2 Usage Attribute List

2.2 IP to IP Use Case

This use case covers a VoIP scenario where the participating parties use completely Internet-based connections.

2.2.1 Basic Flow

1. A VoIP calling party (IP based EP1) signals a GK1 for a call activation and passes in a called party (IP based call recipient – EP2) id / phone number.
2. The GK1 who owns the call may contact (via directory lookups) another GK (GK2) to complete the call if the called party is not a subscriber in the GK1's domain.
3. The GK with the called party as a subscriber acknowledges the request for service activation and proceeds to ring/signal the called party.
4. The called party answers the call and enters into the call connected state. The EP's deliver their voice or voiceband content for a finite amount of time (call duration), and then disconnect the call (call complete). Alternatively, an error may occur or the called party does not respond, and the call is disconnected.
5. In duration of the call, various QoS changes or degradations may occur in real-time. The QoS information including call setup success/failure, call setup time, call drop-out and the quality of the contents (voice, fax and data) can be measured by an IP monitor (IP probe), and provided to the mediation system.
6. The mediation system collects and aggregates the call connection, call type and QoS information about the call from GK's and probes.
7. Upon call completion, the mediation system provides the information collected about the transaction to the BSS system.

Figure 1 depicts an example of the VoIP service for the IP-IP scenario (complete IP-based connection)

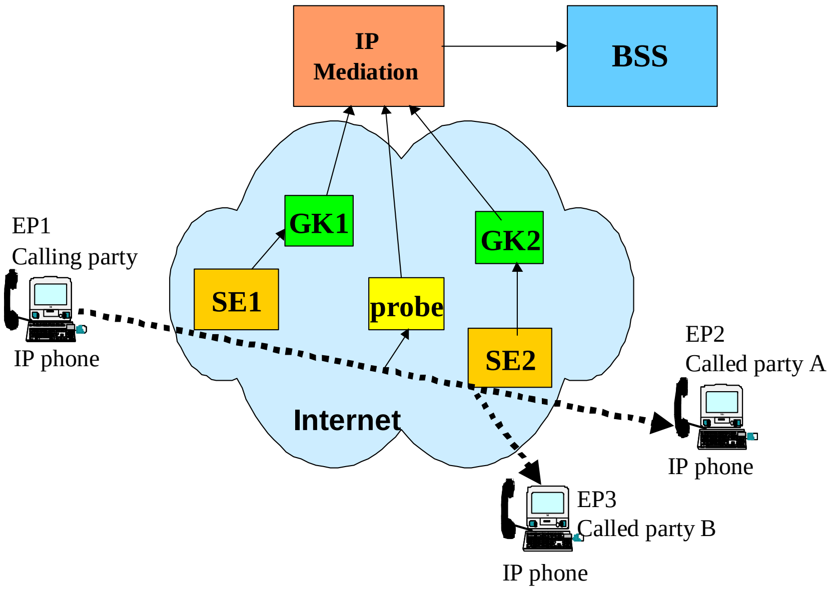

Figure 1 VoIP Service (IP-IP)

2.2.1.1 Basic Flow Requirements

1. Requires at least two EP's for a call to be complete.
2. All EP's must be "On-Net" and all GKs must be "On-Net"
3. An EP needs to be identifiable via either an ID or a phone number.
4. GKs (SP's) must maintain a directory of subscribers, and each subscriber is assigned a unique ID within his domain. Each GK must maintain a directory of other VoIP GKs/GWs.
5. Each GK has a universally unique identifier

A unique call ID is desired to allow the mediation system to identify that IPDRs generated by different GKs and probe are in fact generated for the same call.

2.2.1.2 Basic Flow Usage Attribute List

VoIp Basic Flow Usage Attribute List

Category	Usage Attribute Name	Data Type	Presence	Possible Values	Remarks
When	StartTime	dateTime	R	2000-11-30T22:50:00.000Z	ISO 8601 format. Time when a SC starts using a SE. Answer Time Z = Zulu (Greenwich Mean Time time, GMT)
When	EndTime	dateTime	R	2000-11-30T22:50:00.000Z	ISO 8601 format. Time when a SC stops using a SE. Time when a SC terminates. Hangup time
What	timeZoneOffset	Integer	R	-6	Time offset in hour of local time zone referenced to GMT. Local time zone should reflect calling party time zone for correct billing (as opposed to SE time zone if the two are different).
What	callCompletionCode	string	R	CC:Call completed normally; CAD: abnormal disconnect; UCN: unconnected -network failure; UCI : unconnected-invalid address; CIP: Call in progress	Final call completion code for billing use. CIP indicates event-driven IPDR, which is generated during call/connection progress.
Where	originalDestinationId	string	R	8888@192.168.2.83 214-555-1212	Called-party (designation) number for direct calls; Original dialed number for SIP/800 calls (with called-party number represented by destinationID).

Category	Usage Attribute Name	Data Type	Presence	Possible Values	Remarks
					Equivalent to outpulsed_digits
Who	subscriberId	string	R	"7777@192.171.210.211" "192.171.210.211" "John Doe"	Unique within a service provider network. Tied to a SC or a SE requesting a service.
Who	uniqueCallID	string	R		Unique Call ID to identify that different IPDRs generated by different elements are for the same call.
Who	ipAddress	string	C	199.171.210.211	IP address of the subscriber Conditional based on scenario.
Who	imsiIngress	integer	C	247478674378574	International Mobile Subscriber Identity. Required if calling party is using a cellular phone.
Who	esnIngress	integer	C	A1B2C3D4	Electronic Serial Number which identifies each cellular phone. Required if calling party is using a cellular phone.
What	callProgressState	integer	C	1: Service requested; 2. Address supplied; 3: Address connected; 4. Address not connected 5: Called party answered 6: Conversation in progress 7: call terminated normally 8: call disconnected abnormally	Reported by each SE for call progress state for QoS purpose. Based on callCompletionCode of CIP.
What	disconnetReason	string	C	NormalCallClearing noAnswer busy failure	Reason that call was disconnected based on Call CompletionCode
Where	destinationID	string	C	2145551212 “fred@foo.com”	ID of called party. May be different from

Category	Usage Attribute Name	Data Type	Presence	Possible Values	Remarks
					OriginalDestinationId if translation has occurred such as 800 or SIP redirect
What	thirdPartyID	String	C	2145551212 "fred@foo.com"	Third party ID if PaymentType value is charged_to_3rd_party
Where	ani	string	O	"8888@192.168.2.83"	Automatic Number Identification; Calling party number identification
Where	oLIiiDigit	string	O	000: No special treatment; 006: Hotel; 029: Prison inmate 061: Cellular service type-1 access 062: Cellular service type-2 access	Originating Line Identifier (OLI) per SS7-ISUP signaling (3-digits); Identification Information Digit (iiDigit) for MF-winked signaling (2 digits).
Where	dnis	string	O	2145551212	Dialed Number Identification Service (number sent to an answering service) Should be populated if two stage dialing is performed.
Who	pin	string	O	6294621	
Who	serviceConsumerType	string	O	EU (end user) NE (network element) NK (network partner)	
When	startAccessTime	dateTime	O	2000-11-30T22:50:00.000Z	ISO 8601 format. Time when a SC starts using a NE. Time when a SC initiates a call/connection request. Off-hook Time.
When	endAccessTime	dateTime	O	2000-11-30T22:50:00.000Z	ISO 86-1 format. Time when a SC terminates a call/connection request. Tear-down time
What	callSetupDuration	integer	O	180000	Value in millisecond from Start_access_time to Start_time Can be calculated startTime-startAccessTime.
What	callDuration	integer	O	180000	Value in milliseconds excluding all call setup procedures. Can be calculated endTime-startTime.
What	totalDuration	integer	O	180000	Value in milliseconds from startTime to endTime.

Category	Usage Attribute Name	Data Type	Presence	Possible Values	Remarks
					Can be calculated EndTime-startAccessTime
Whate	TearDownDuration	integer	O	180000	Value in millisecond from endTime to endAccessTime Can be calculated endAccessTime-endTime
What	averageLatency	integer	O	145	Average latency in milliseconds
What	type	string	O	A (Administrative (authentication, authorization)) I (IVR) N (no answer) V (voice) D (data) F (fax) VF (voice and fax) VD (voice and data)	Type of call.
What	payementType	string	O	Toll-free, charge_to_calling party, charge_to_called party, charged_to_3rd_party, prepaid	
What	feature	string	O	R (roaming) H (home)	Indicates if a subscriber is using the service from their home number.
What	codec	string	O	G711Alaw G711Mulaw G723 Low G723 High G726 G727 G728 G729A P (proprietary)	Codec standard (ITU, GSM) used to convert the analog signal (speech) to digital and vice versa.
What	modem	string	O		Required if a modem is involved.
What	supplementaryService	string	O	call forwarding	
What	extendedReasonCode	string	O		
What	disconnectLocation	string	O		
What	proprietaryErrorCode	integer	O		
What	unitsConsum	integer	O		

Category	Usage Attribute Name	Data Type	Presence	Possible Values	Remarks
	ed				
What	inboundByteCount	integer	O	64	Total number of bytes received towards the SE
What	outboundByteCount	integer	O	64	Total number of bytes transmitted from the SE
What	inboundPacketCount	Integer	Optional		
What	outboundPacketCount	Integer	Optional		
What	inboundLostPacketCount	Integer	Optional		
What	outboundLostPacketCount	Integer	Optional		
What	inboundRxmtPacketCount	Integer	Optional		
What	outboundRxmtPacketCount	Integer	Optional		
What	subscribedQoSClasses	integer	O	4= Best 3= High 2= Medium 1= Best Effort	Subscribed VoIP QoS classes as defined in ETSI TIPHON TR-101-329 & DTS-101-512, as a function of call setup time, end-to-end delay, transmission quality and conversation quality
What	callClarityIndex	integer	O	Call Clarity Index, 1 to 5: 5= Excellent 4= Good 3= Fair 2= Poor 1= Bad	Call Clarity Index per ITU-T Rec. P.562 for conversation quality
What	voiceQualityIndex	integer	O	Voice Quality Mean Opinion Score, 1 to 5: 5= Excellent 4= Good 3= Fair 2= Poor 1= Bad	Voice Quality Mean Opinion Score (MOS) per ITU-T Rec. P.862 or equivalent
What	overallTransmissionQualityIndex	integer	O	E-Model Index R: in range of 0 (Bad) to 100 (Excellent)	The E-Model Index per ITU-T Rec. G.107
What	faxPerfomanceMetric	integer	O	Seven-scale figure of merit : I to VII	Figure of Merit for Fax Transmission Performance, per ITU-T Rec. E.458:
What	faxPageTransmittedCount	integer	O		
What	faxPageReceivedCount	integer	O		
What	packetLossP	integer	O	.03	Percentage of packet loss

Category	Usage Attribute Name	Data Type	Presence	Possible Values	Remarks
	ercentage				towards the SE.
What	outOfSequencePackets	integer	O	152	Total number of out of sequence packets received from the SE
What	correctSequencePackets	integer	O	152	Number of in sequence packets received towards the SE.
What	packetDelayVariation	integer	O	Milliseconds	
Where	ipAddressIngressDevice	string	O	199.171.210.211	Required if using a gateway, not present if using DHCP.
Where	ipAddressEgressDevice	string	O	199.171.210.211	Required if using a gateway, not present if using DHCP.
Where	portNumber	string	O	1720	Port number utilized for call
Where	imsiEgress	string	O	247478674378574	International Mobile Subscriber Identity. Required if called party is using a cellular phone.
Where	esnEgress	string	O	A1B2C3D4	Electronic Serial Number which identifies each cellular phone. Required if called party is using a cellular phone.
Where	homeLocationIdIngress	string	O	FF01ABD6	HLRID of the cellular calling party Required if calling party is using a cellular phone.
Where	homeLocationIdEgress	string	O	FF01ABD6	HLRID of the cellular called party Required if called party is using a cellular phone.

2.2.2 Alternative/Specific Flow

In addition to the above-detailed IP-IP scenario, other VoIP service scenarios include:

1. IP to Switched Communications Network (SCN, including wireline PSTN and wireless), as depicted in Figure 2.
2. SCN (wireline) to IP, as depicted in Figure 3, and SCN (wireless) to IP, in Figure 4.
3. IP-SCN-IP, as depicted in Figure 5
4. SCN-IP-SCN, as depicted in Figure 6

Detailed Basic Flow and Usage Attributes for these additional scenarios are to be formulated as further work.

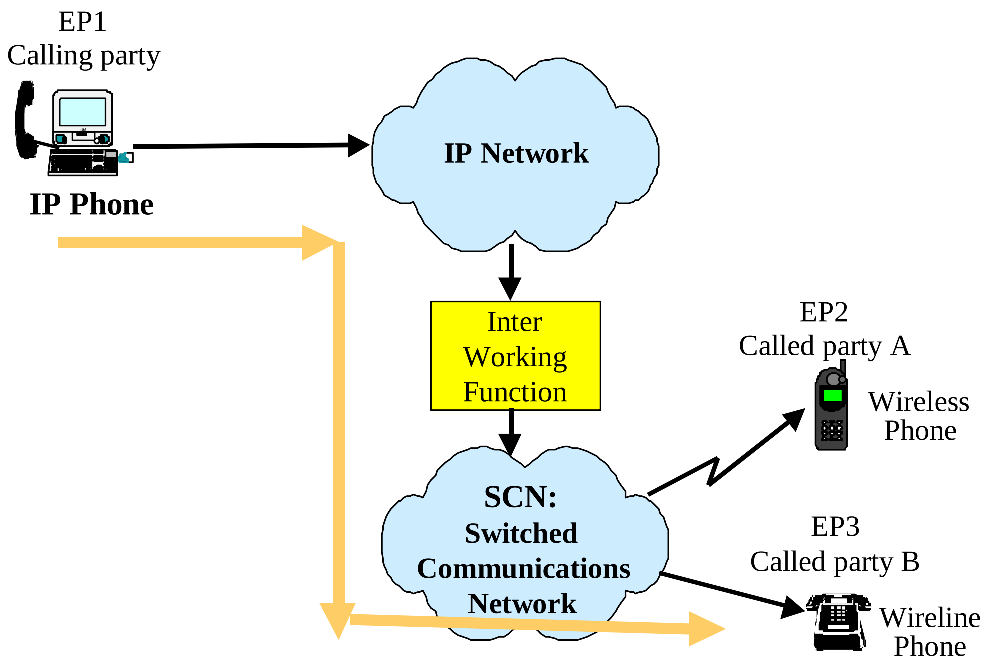

Figure 2: IP to SCN (Switched Communications Network) Service

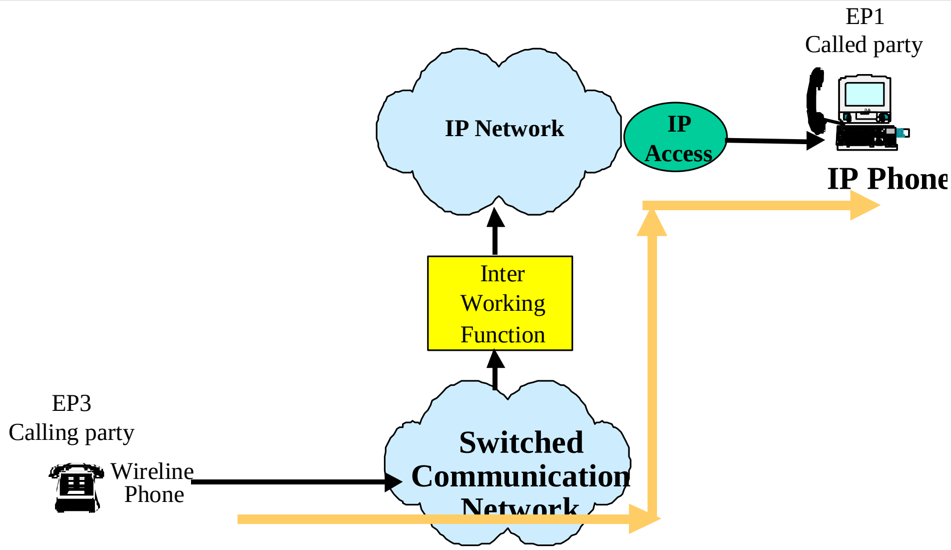

Figure 3: SCN (wireline) to IP Service

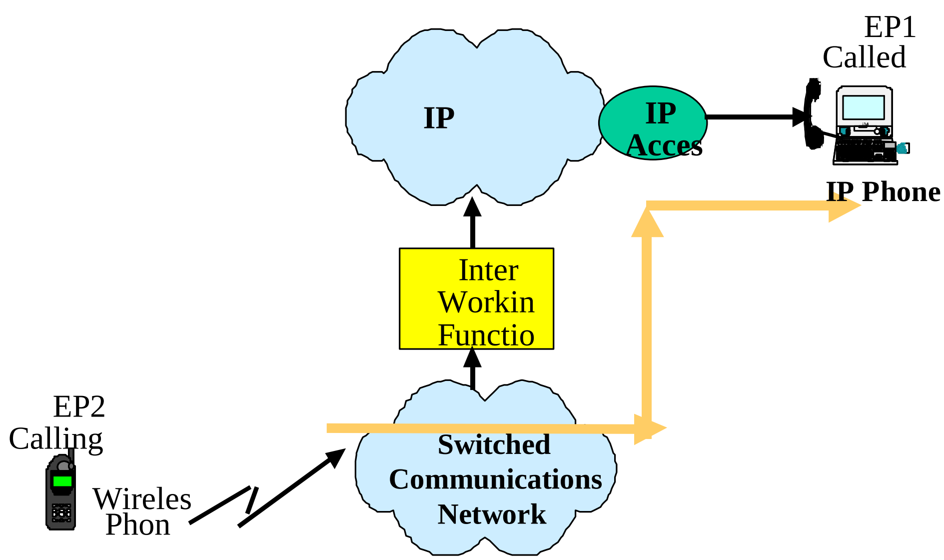

Figure 4: SCN (wireless) to IP Service

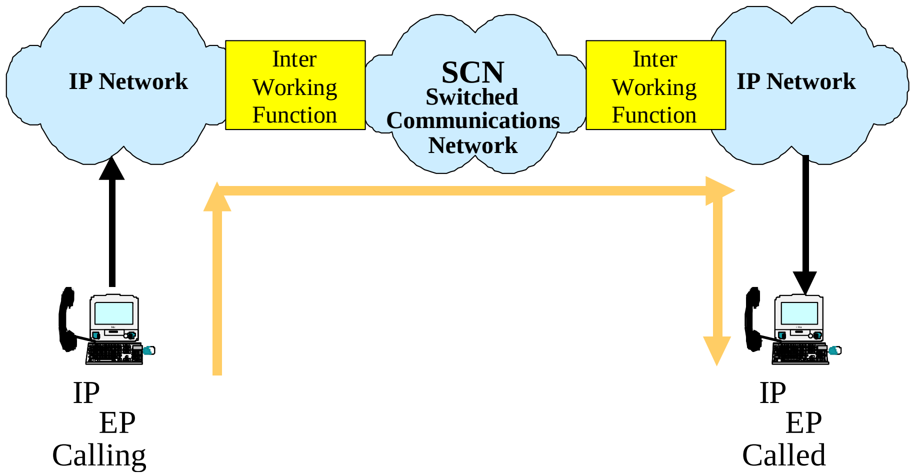

Figure 5: IP - SCN -IP Service

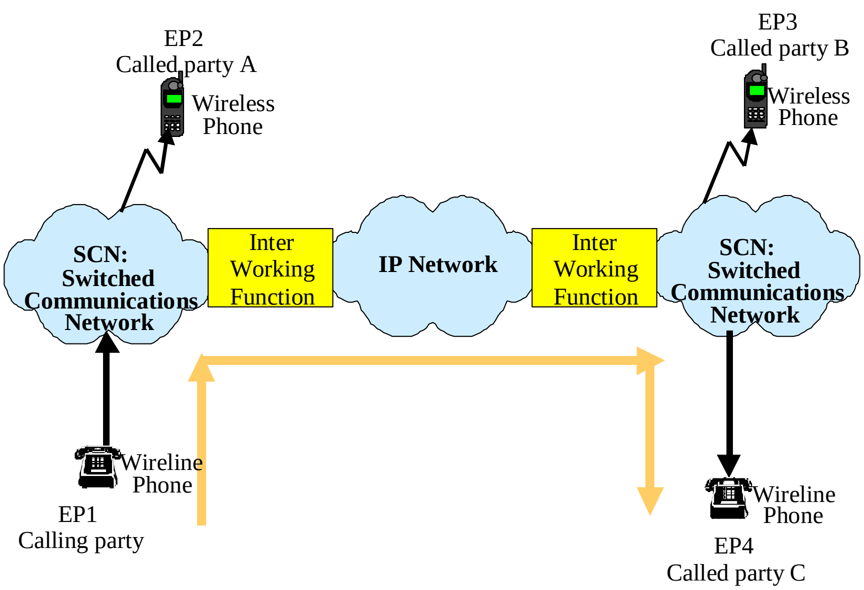

Figure 6: SCN - IP - SCN Service

3.0 Formal Specification

3.1 Schema

```
<?xml version = "1.0" encoding = "UTF-8"?>
<schema xmlns = "http://www.w3.org/2000/10/XMLSchema"
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                                        <element name = "portNumber" type = "string" minOccurs = "0"/>
                                        <element name = "imsiEngress" type = "string" minOccurs = "0"/>
                                        <element name = "esnEgress" type = "string" minOccurs = "0"/>
                                        <element name = "homeLocationIngress" type = "string" minOccurs = "0"/>
                                        <element name = "homeLocationIdEgress" type = "string"
                                                minOccurs = "0"/>
                                </sequence>
                        </extension>
                </complexContent>
        </complexType>
</schema>
```

3.2 Sample Instance Document

3.2.1 VoIP Sample Scenarios

3.2.1.1.1 PSTN to IP

3.2.1.1.1.1 Use Case

1. The pstn subscriber calls the local access number for the gateway.
2. The gateway queries a network element that will verify the subscribers account.

3. The user is prompted to, and, enters a PIN and destination (calling party) phone number.
4. The gateway consults the gatekeeper on ways to route the call.
5. The gatekeeper looks up the destination phone number against a table and sends the gateway the destination IP address.
6. The originating gateway places a call across the IP network to the destination terminal (phone).
7. When the conversation is complete, either party hangs up the phone, and the call is terminated with the cause normal call clearing.

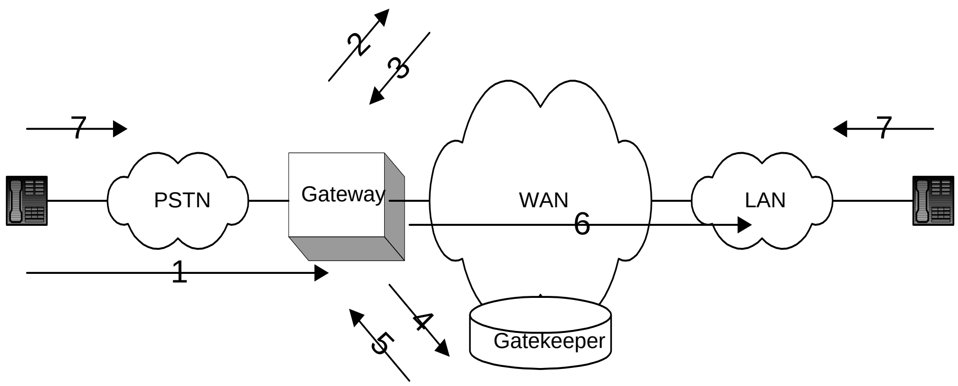

Fig. WW - PSTN to IP

3.2.1.1.1.2 Document Instance

* Editors note: This instance does not contain all required attributes


```
<?xml version="1.0"?>

<!-- Assumptions:
                                   Call is being made from PSTN to IP
                                   Call is terminated (normally) by the called side
                                   Optional and Conditional fields are included based on the type of call made
                                   Fields that did not include specific values (in the ipdr spec) have been
                                   populated with information based on SS7 equivalents -->

<IPDRDoc xmlns="http://www.ipdr.org/namespaces/ipdr"
        xmlns:xsi="http://www.w3.org/2000/10/XMLSchema-instance"
        xsi:schemaLocation="http://www.ipdr.org/namespaces/ipdr VoIP2.5-A.0.xsd"
        docId=”f9c0ca84-1111-0222-90ef-fd73546596bb”
        version="2.5">

        <IPDRRec info="apex.virtualsummit.com"/>
        <IPDR seqNum="1" time="2000-02-01T07:00:00Z" >
                <SS id="ses10" service="rtsp">
                        <SC xsi:type="SC-VoIP-Type">
                                <subscriberId>Vendor Phone-1</subscriberId>
                                <ipAddress>172.1717.10</ipAddress>
                        </SC>
                        <SE xsi:type="SE-VoIP-Type">
                                <hostname>cisco.gateway.234</hostname>
                        </SE>
                </SS>

                <UE xsi:type="UE-VoIP-Type">
                        <serviceConsumerType>EU</serviceConsumerType>
```

```
                <pin>6294621</pin>
                <startAccessTime>2000-11-24T09:59:45Z</startAccessTime>
                <startTime>2000-11-24T10:00:00Z</startTime>
                <endTime>2000-11-24T10:20:00Z</endTime>
                <timeZoneOffset>-5</timeZoneOffset>
                <callDuration>1200</callDuration>
                <type>V</type>
                <codec>G711Alaw</codec>
                <disconnectReason>normalCallClearing</disconnectReason>
                <averageLatency>150</averageLatency>
                <ani>214-680-6400</ani>
                <originalDestinationId>408-830-3711</originalDestinationId>
                <ipAddressEgressDevice>199.171.210.200</ipAddressEgressDevice>
                <portNumber>17779</portNumber>
            </UE>
        </IPDR>
</IPDRDoc>
```


3.2.1.1.2 PSTN to IP, IP user Busy

3.2.1.1.2.1 Use Case

1. The pstn subscriber calls the local access number for the gateway.
2. The gateway queries a network element that will verify the subscribers account.
3. The user is prompted to, and, enters a PIN and destination (calling party) phone number.
4. The gateway consults the gatekeeper on ways to route the call.
5. The gatekeeper looks up the destination phone number against a table and sends the gateway the destination IP address.
6. The originating gateway places a call across the IP network to the destination terminal (phone).
7. The calling party receives a busy signal from the far end, and the call is released with the cause user busy.

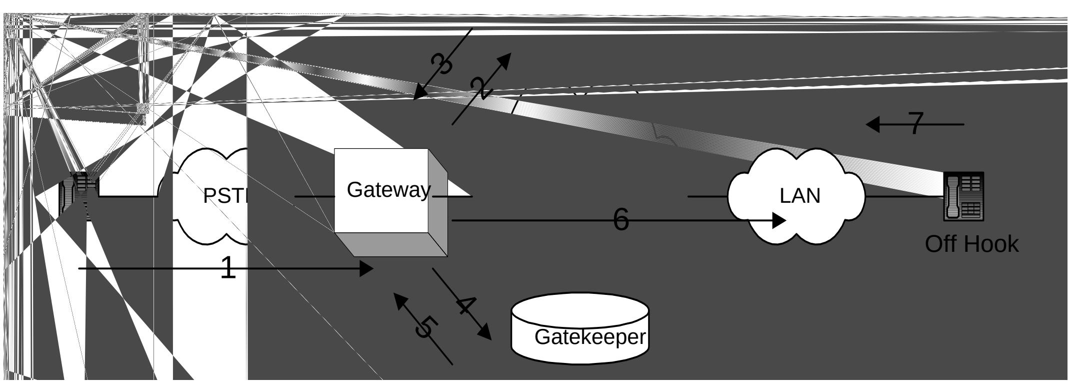

Fig. XX - PSTN to IP, IP party busy

3.2.1.1.2.2 Document Instance

* Editors note: This instance does not contain all required attributes


```
<?xml version="1.0"?>

<!-- Assumptions:
                Call is being made from PSTN to IP, called party is busy
                Call is terminated (normally) by the called side
                Optional and Conditional fields are included based on the type of call made
```

```
                                        Fields that did not include specific values (in the ipdr spec) have been
                                        populated with information based on SS7 equivalents -->

<IPDRDoc xmlns="http://www.ipdr.org/namespaces/ipdr"
        xmlns:xsi="http://www.w3.org/2000/10/XMLSchema-instance"
        xsi:schemaLocation="http://www.ipdr.org/namespaces/ipdr VoIP2.5-A.0.xsd"
        docId="f9c0ca84-2222-90ef-a222-fd73546596bb"
        version="2.5">

        <IPDRRec info="apex.virtualsummit.com"/>
        <IPDR seqNum="1" time="2000-02-01T07:00:00Z">
                <SS id="ses10" service="rtsp">
                        <SC xsi:type="SC-VoIP-Type">
                                <subscriberId>Vendor Phone-1</subscriberId>
                                <ipAddress>172.17.17.10</ipAddress>
                        </SC>
                        <SE xsi:type="SE-VoIP-Type">
                                <hostname>cisco.gateway.234</hostname>
                        </SE>
                </SS>

                <UE xsi:type="UE-VoIP-Type">
                        <serviceConsumerType>EU</serviceConsumerType>
                        <pin>6294621</pin>
                        <startAccessTime>2000-11-25T10:59:20Z</startAccessTime>
                        <startTime>2000-11-25T11:00:35Z</startTime>
                        <endTime>2000-11-25T11:00:40Z</endTime>
                        <timeZoneOffset>-5</timeZoneOffset>
                        <callDuration>5</callDuration>
                        <type>V</type>
                        <codec>G711Alaw</codec>
                        <disconnectReason>userBusy</disconnectReason>
                        <averageLatency>150</averageLatency>
                        <ani>214-680-6400</ani>
                        <originalDestinationId>408-830-3711</originalDestinationId>
                        <ipAddressEgressDevice>199.171.210.200</ipAddressEgressDevice>
                        <portNumber>17779</portNumber>
                </UE>
        </IPDR>
</IPDRDoc>
```


3.2.1.1.3 IP to PSTN

3.2.1.1.3.1 Use Case

1. The IP subscriber dials the destination phone number.
2. The gatekeeper is consulted on ways to route the call.
3. A connection is established between the terminal and gateway.
4. The gateway places a call to the pstn by outpulsing the destination (called) number.
5. When the conversation is complete, either party hangs up the phone, and the call is terminated with the cause normal call clearing.

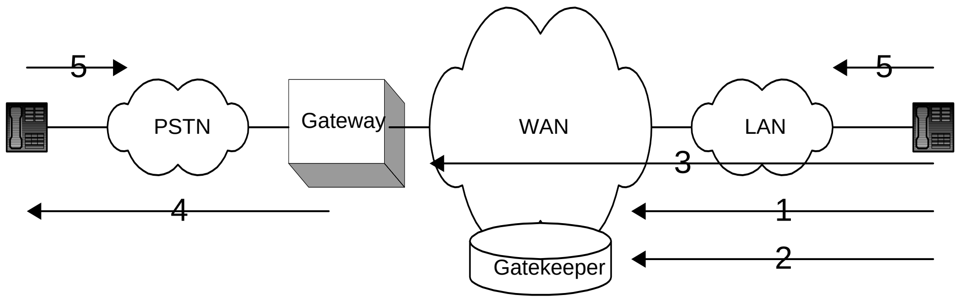

Fig. YY- IP to PSTN

3.2.1.1.3.2 Document Instance

* Editors note: This instance does not contain all required attributes


```
<?xml version="1.0"?>

<!-- Assumptions:
                    Call is being made from Ip to PSTN
                    Call is terminated (normally) by the called side
                    Optional and Conditional fields are included based on the type of call made
                    Fields that did not include specific values (in the ipdr spec) have been
                    populated with information based on SS7 equivalents -->

<IPDRDoc xmlns="http://www.ipdr.org/namespaces/ipdr"
    xmlns:xsi="http://www.w3.org/2000/10/XMLSchema-instance"
    xsi:schemaLocation="http://www.ipdr.org/namespaces/ipdr VoIP2.5-A.0.xsd"
    docId=”f9c0ca84-3333-a222-90ef-fd73546596bb”
    version="2.5">

    <IPDRRec info="apex.virtualsummit.com"/>
    <IPDR seqNum="1" time="2000-02-01T07:00:00Z">
        <SS id="ses10" service="rtsp">
            <SC xsi:type="SC-VoIP-Type">
                <subscriberId>Vendor Phone-1</subscriberId>
                <ipAddress>199.171.210.200</ipAddress>

            </SC>
            <SE xsi:type="SE-VoIP-Type">
                <hostname>cisco.gateway.234</hostname>
            </SE>
        </SS>

        <UE xsi:type="UE-VoIP-Type">
            <serviceConsumerType>EU</serviceConsumerType>
            <pin>8295430</pin>
            <startAccessTime>2000-11-25T11:00:00Z</startAccessTime>
            <startTime>2000-11-25T11:15:15Z</startTime>
            <endTime>2000-11-25T11:35:00Z</endTime>
            <timeZoneOffset>-5</timeZoneOffset>
            <callDuration>1185</callDuration>
            <type>V</type>
            <codec>G711Alaw</codec>
            <disconnectReason>normalCallClearing</disconnectReason>
```

```
                    <averageLatency>145</averageLatency>
                    <ani>408-830-3711</ani>
                    <originalDestinationId>214-680-6400</originalDestinationId>
                    <ipAddressIngressDevice>199.171.210.200</ipAddressIngressDevice>
                    <portNumber>17779</portNumber>
          </UE>
     </IPDR>
```


3.2.1.1.4 Cellular to IP

3.2.1.1.4.1 Use Case

1. The cellular subscriber calls the local access number for the gateway.
2. The gateway queries a network element that will verify the subscribers account.
3. The user is prompted to, and, enters a PIN and destination (calling party) phone number.
4. The gateway consults the gatekeeper on ways to route the call.
5. The gatekeeper looks up the destination phone number against a table and sends the gateway the destination IP address.
6. The originating gateway places a call across the IP network to the destination terminal (phone).
7. When the conversation is complete, either party hangs up the phone, and the call is terminated with the cause normal call clearing.

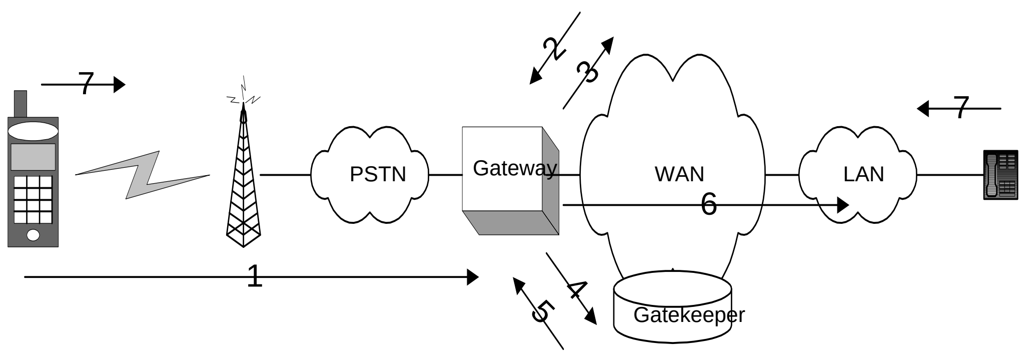

Fig ZZ - Cellular to IP

3.2.1.1.4.2 Document Instance

* Editors note: This instance does not contain all required attributes


```
<?xml version="1.0"?>
<!-- Assumptions:
                         Call is being made from cell phone to IP
                         Call is terminated (normally) by the called side
                         Optional and Conditional fields are included based on the type of call made
                         Fields that did not include specific values (in the ipdr spec) have been
                         populated with information based on SS7 equivalents
                         IMSI/ESN/PIN/HLRID values are fictitious -->
<IPDRDoc xmlns="http://www.ipdr.org/namespaces/ipdr"
     xmlns:xsi="http://www.w3.org/2000/10/XMLSchema-instance"
     xsi:schemaLocation="http://www.ipdr.org/namespaces/ipdr VoIP2.5-A.0.xsd"
     docId="f9c0ca84-4444-90ef-a222-fd73546596bb"
     version="2.5">

     <IPDRRec info="apex.virutalsummit.com"/>
```

```
        <IPDR seqNum="1" time="2000-02-01T07:00:00Z">
                <SS id ="ses10" service="rtsp">
                        <SC xsi:type="SC-VoIP-Type">
                                <subscriberId>Vendor Phone-1</subscriberId>
                                <ipAddress>172.17.17.10</ipAddress>
                        </SC>
                        <SE xsi:type="SE-VoIP-Type">
                                <hostname>cisco.gateway.234</hostname>
                        </SE>
                </SS>

                <UE xsi:type="UE-VoIP-Type">
                        <imsiIngress>247478674378574</imsiIngress>
                        <esnIngress>33375629401</esnIngress>
                        <serviceConsumerType>EU</serviceConsumerType>
                        <pin>6294621</pin>
                        <startAccessTime>2000-11-25T09:45:30Z</startAccessTime>
                        <startTime>2000-11-25T09:45:45Z</startTime>
                        <endTime>2000-11-25T10:00:30</endTime>
                        <timeZoneOffset>-5</timeZoneOffset>
                        <callDuration>885</callDuration>
                        <type>V</type>
                        <feature>H</feature>
                        <codec>G711Alaw</codec>
                        <disconnectReason>normalCallClearing</disconnectReason>
                        <averageLatency>145</averageLatency>
                        <ani>214-924-0258</ani>
                        <originalDestinationId>408-830-3711</originalDestinationId>
                        <ipAddressEgressDevice>199.171.210.211</ipAddressEgressDevice>
                        <portNumber>17779</portNumber>
                        <homeLocationIdIgress>FF01ABD6</homeLocationIdIgress>
                </UE>
        </IPDR>
</IPDRDoc>
```