988
Originating Service Provider Network
SIP Interface Specification

Version 0.2

October 17, 2023

Contents

List of Figures

Document History

Version	Date	Author	Notes
0.1	10/11/2023	John Snapp	Initial Intrado Draft
0.2	10/17/2022	John Snapp	Second Draft - Addition of PSAP ID to X-Header

1.0 Overview

This specification defines how Originating Service Providers (OSPs) pass 988 calls into the Vibrant 988 network.

988 is a 3-digit national number to connect directly with the 988 Suicide and Crisis Lifeline. On July 16, 2020, the FCC adopted rules to establish 988 as the nationwide 3-digit code for people in crisis to connect with suicide prevention and mental health crisis counselors. The Suicide and Crisis Lifeline is a national network of more than 200 crisis centers.

Geolocation services are not currently enabled for 988, as they are for calls to 911. A caller's location information is not transmitted with a 988 call for possible dispatch of emergency services. The Lifeline automatically routes 988 calls based on the area code (Number Planning Area or NPA) and central office code (NXX) to the nearest crisis center. For example, when a wireless caller with the phone number 206-222-0000 (which is assigned to a wire center in Seattle, WA) dials 988, the Lifeline will route the call to a Seattle Lifeline call center regardless of the caller's actual location.

This network-to-network (NNI) interface defines an interface where an OSP can pass a call to 988 with a specific destination code that the 988 network will utilize to route the call to the geographically appropriate crisis center. The destination code will be described in more detail below.

1.1 Purpose and Applicability

The purpose of this specification is to define connectivity options, signaling, and parameter exchange between an IP-based originating network and the 988 network. Orginating networks may be wireless, wireline, or VoIP Service Providers (VSPs) that are deploying the IP-based networks required to carry 988 traffic.

This specification includes:

- Interconnection architecture
- SIP call/session control signalling
- Signaling and media transport

The following are not defined in this document:

- Destination code generation within OSP networks
- Translations within OSP networks

1.2 Assumptions

1. Each originating network provider must use a set of call control, call routing, and bearer functional entities to connect to the 988 network.
2. SIP is used for call control signaling.
3. RTP is used for voice transport.
4. G.711 uLaw encoding at 20ms packetization is recommended for best voice quality.
5. The OSP network will anchor media.
6. Only IP transport is used. TDM, ISUP, ISDN, etc., forms of signaling are not supported.

2.0 Getting Ready for IP Ingress

2.1 OSP to 988 Network

This section is repeated from the OSP to 988 network using Basic SIP interfaces. If the OSP already has IP connectivity to the 988 network, no further connectivity is required. The OSP may want to take the opportunity to review bandwidth requirements and make adjustments as warranted.

2.2 IP Ingress to 988 network from OSP

Below are some high-level views for how SIP IP traffic will transit from OSP to 988 network.

Figure 1: Direct IP from OSP to 988 network

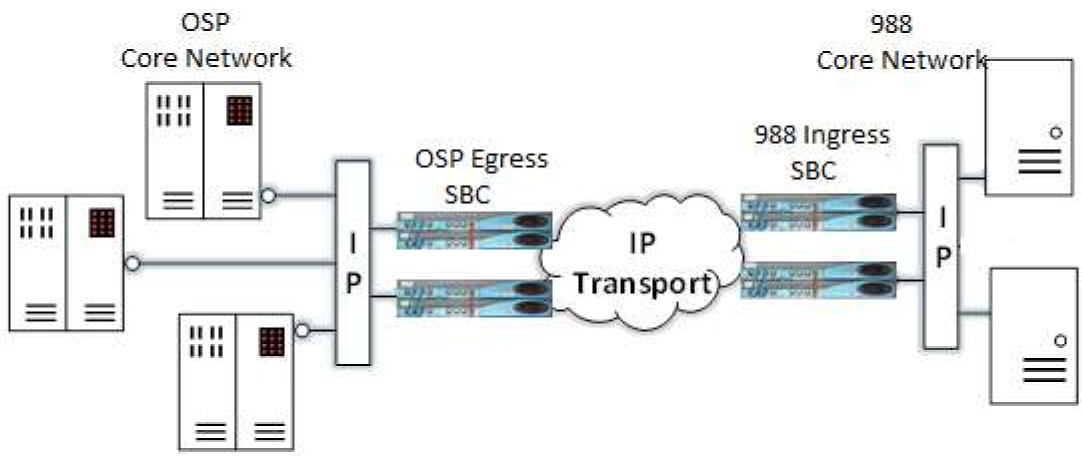


Figure 2: Direct IP from OSP to 988 network

In the arrangement depicted above, the OSP has the same two IP egress points (Session Border Controllers, also known as SBCs). Typically, egress SBCs are operated in High Availability (HA) pairs and are geographically dispersed. The OSP egress Session Border Controllers (SBCs) have IP connectivity to the 988 Service Provider's ingress SBCs.

2.3 OSP to 988 network: Transport

The standard connection would be redundant transport across the public internet. The OSP must have static public IP addresses. VPN and dedicated MPLS circuits will be considered on a case by case basis.

2.3.1 Bandwidth Considerations

As with SS7 traffic, the OSP will traffic engineer their 988 traffic for IP ingress to the 988 network. Each End Office, VoIP platform or 5G platform, will be traffic engineered to meet appropriate Poisson thresholds (Poisson calculation to the appropriate blocking factor). In the legacy TDM world this would result in a DS0 count. In IP, this typically results in a concurrent call path (CCP) count.

To determine the bandwidth required, take the DS0 or CCP count and multiply it by 100 kilobits/sec. Why 100kbps? Typically providers use the G.711 codec for voice. This codec typically operates at around 87 kbps. 100 kbps provides some buffer and allows for additional monitoring traffic. The OSP may want to add additional bandwidth for future growth (perhaps this is accounted for via the peaking factor in the Poisson calculation). IP bandwidth is typically very affordable, so this should not add a significant amount of new costs. The appropriately sized IP transport is then used between the OSP and 988 network.

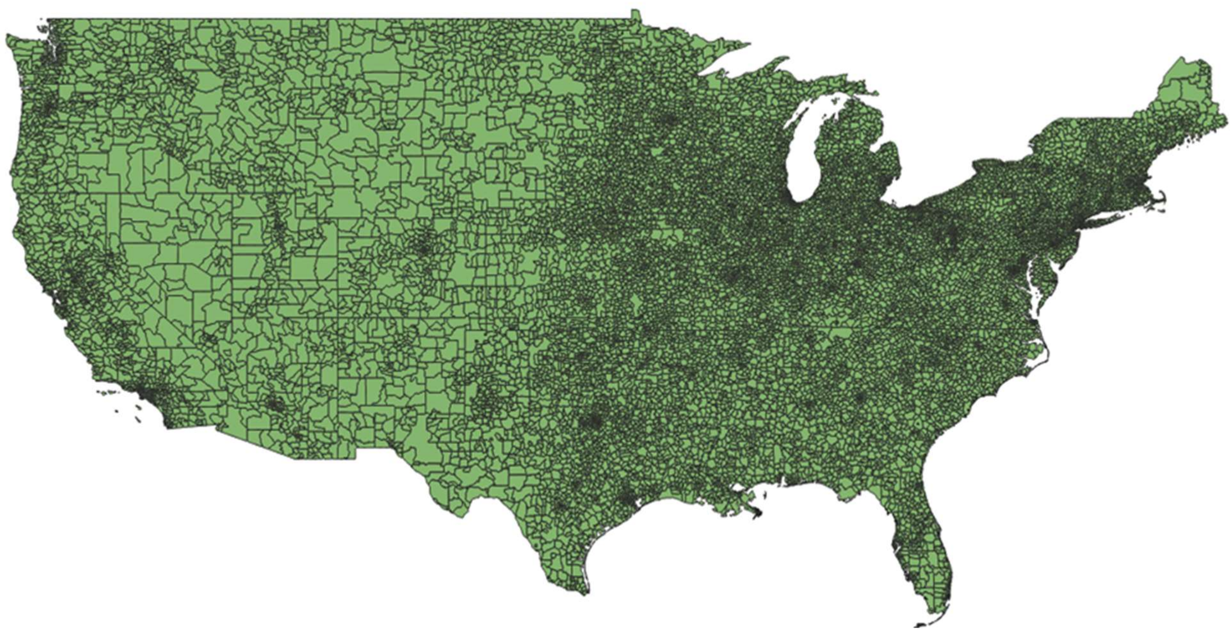
2.3.2 IP Trunks

As with TDM trunks, an IP trunk contains a collection of calls. Unlike TDM, in SIP there is not a requirement to organize the calls into any kind of context. Calls from all over the country could be sent to a single IP trunk, as long as the call was appropriate to the 988 network.

3.0 Destination Code

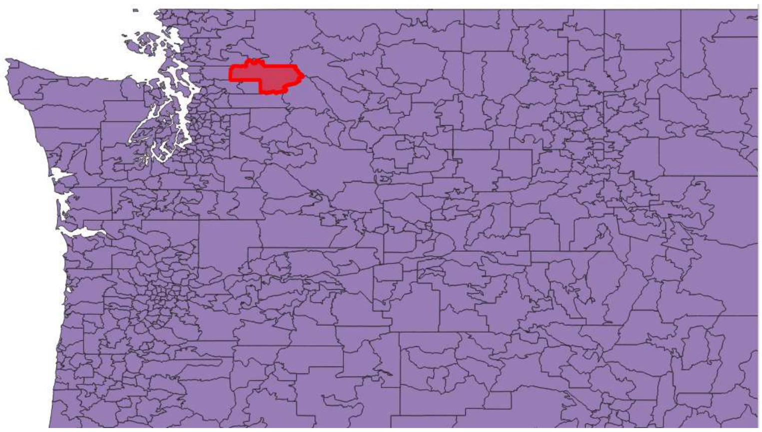
A destination code is used to convey the location area of the caller that is compatiable with the 988 network existing routing methodology. The destination code will allow the call to be routed to the 988 crisis center that is geographically relevant to where the caller is without infringing upon the caller's expectation of location privacy. It provides this functionality by providing a geographically relevant number based upon the geographic wire center boundary they are calling from. Wire center boundaries are the basic unit of telecommunications geography and were traditionaly the area that was served by a wireline central office. Across the united states there are over 20,000 wirecenters that have been defined.

Figure 3: Map of US Wire Center Boundaries



When new telephone number ranges or exchanges are allocated, they are associated with an individual wire center. They can be assigned in blocks of 10,000 or 1,000 numbers. The exchange is typically refered to as NPA-NXX where the NPA is the area code and the NXX is the office code. For example the number range of 360-436-0000 through 9999 was assigned to the wire center of Darrington, WA in 1994. The actual wire center CLLI code is DRTNWAXX. These numbers were originally assigned to wireline phones within the geographic wire center boundary of DRTNWAXX. Every phone number range (10,000 number range or 1,000 number range) within the United States is assigned to a single wire center, but a single wirecenter may have many different number ranges assigned to it. The LERG or local exchange routing guide defines the relationship of number ranges to wire centers.

Figure 4: Map of Darrington Wire Center Boundary



Destination codes utilize these wire center boundaries and and the number ranges assigned to them to convey the rough geographic area of the caller and are utilized by the 988 network to route the 988 call to the appropriate crisis center for the geographic area.

The OSP must construct a destination code by determining the wire center boundary that the 988 caller is calling from. One method of doing this would be to determine the wire center that contains the cell site that is being used by the 988 caller. Once the wire center is determined, a 10-digit destination is chosen from any of the number ranges assigned to that wire center. It does not matter if this number has been actually assigned to a customer or has been ported out. It will only be used to route the call within the 988 network and will not be delivered to the 988 call taker.

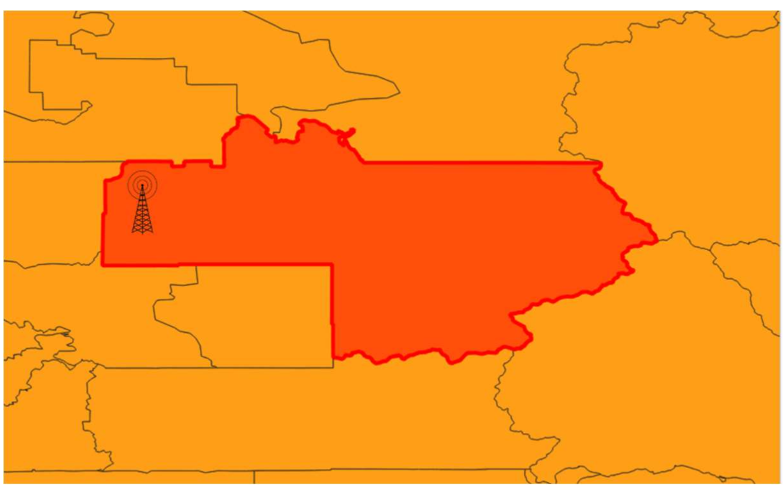


Figure 5: Map of Darrington Wire Center Boundary with cell within

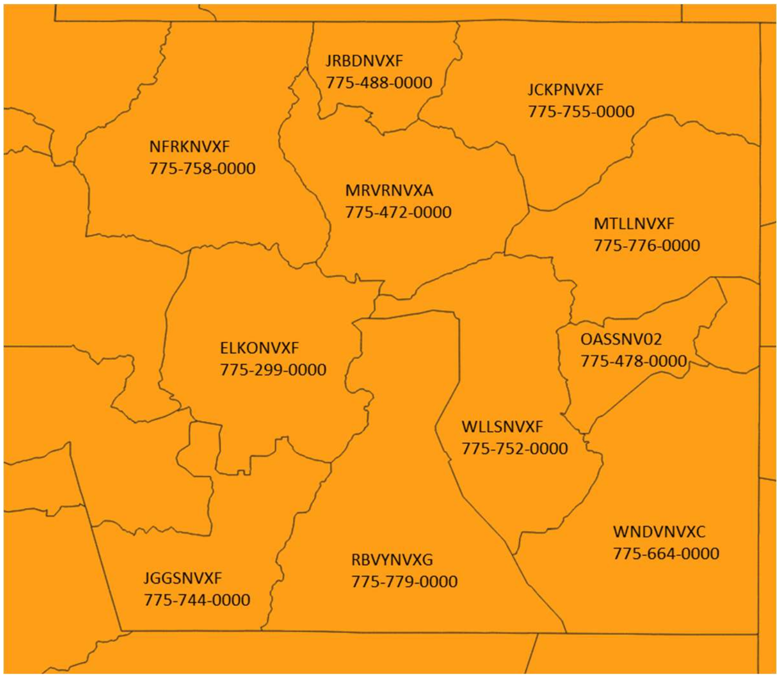


Figure 6: Example possible destination codes and WC Boundaries in NV

For example, if a 988 call was placed from a cell tower that was within the Darrington, WA wire center, as shown above, any NPA-NXX-XXXX that is assigned in the LERG to the Darrington, WA wire center can be used as a destination code. Once such exchange that is assigned to Darrington is NPA=360, NXX=436, STARTLINE=0000, ENDLINE=9999. From that range, we could pick 360436000 as our destination code.

The destination code will be conveyed to the 988 network by utilizing a special X-Header field in the SIP invite between the OSP and the 988 network. The X-Header is formatted as follows:

X-988: 999bbbbbbbbbb

Where X-988 is the name of the X-Header

999 indicates that the digits to follow are a 10-digit destination code

bbbbbbbbbb is the 10-digit destination code

For the example above, the destination code would be:

X-988:9993604360000

4.0 PSAP ID

When a 988 call is passed to a Crisis Center, around 3% of those calls need PSAP involvement (Police, Fire or Ambulance). When this occurs, the Crisis Center obtains location information from the caller and then determines which PSAP covers that area. While Wire Center derived destination codes are appropriate for georouting to Crisis Centers, they do not align well with PSAP boundaries.

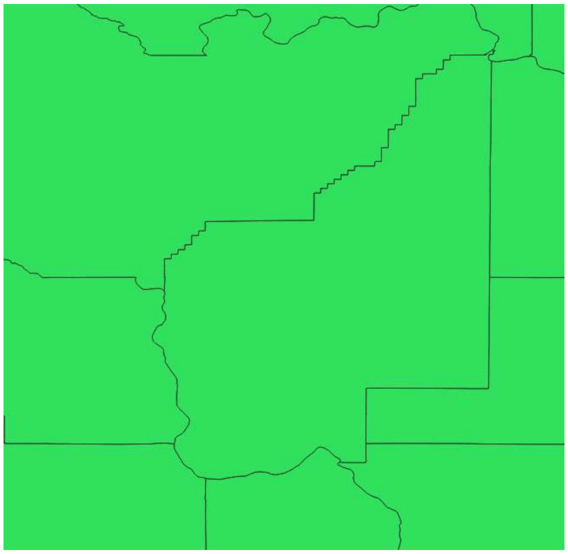


Figure 7: PSAP Boundary for Moses Lake WA

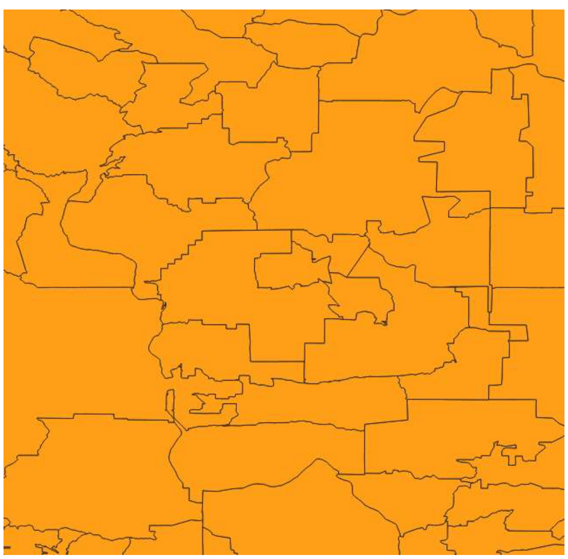


Figure 8: Wire center Boundary for same area

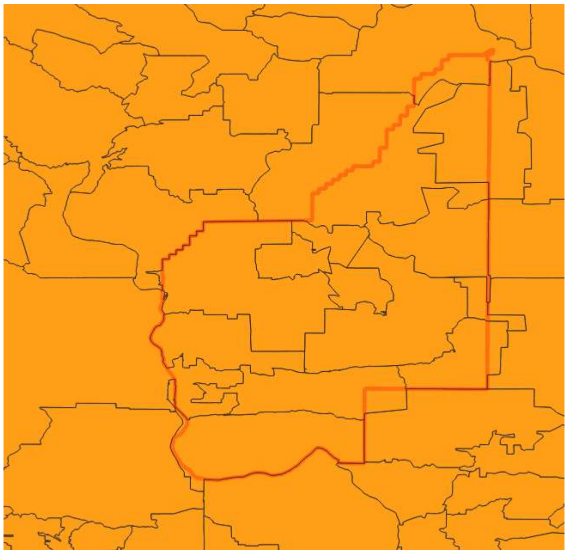


Figure 9: PSAP boundary overlayed on rate center boundary

Figure 6 above shows a typical PSAP boundary. This boundary is for Moses Lake, WA. Figure 7 shows the Wire centers covering the same area. As you can see in Figure 8, the wire centers often do not follow the same boundaries as PSAPs.

By passing a PSAP ID (see below) along with the Destination code, the call can be delivered to the geographically appropriate Crisis Center along with the Crisis Center having the proper PSAP for the caller if public safety involvement is required.

To pass the PSAP ID, a similar process is used to determine the PSAP as was used to determine wire center for the Destination Code. To construct the PSAP ID, the OSP takes the location of the cell ID and determines the PSAP that the call falls within. This is a similar process to what is done to determine the Phase 1 routing of 911 calls.

The PSAP ID will be defined as the FCC PSAP ID. This is currently a 4-digit code but 5 digits will be reserved. The format of the SIP header will be as follows:

X-988: 998PPPPPDDDDDDDDDD

Where:

X-988:	- SIP Header
998	- Identifies this is a 988 call with PSAP ID and Destination Code
PPPPP	- 5-Digit FCC PSAP ID
DDDDDDDDDD	- 10-Digit Destination Code

5.0 SIP Interface Options

5.1 SIP

The SIP trunks to the 988 network will be the primary route for 988 calls, but because of the critical nature of the 988 calls, failure routing must be implemented in addition to main routing. The two options are discussed below.

5.1.1 Main Routing

The OSP will route the call, over the dedicated routes to 988. The R-URI and TO: headers should contain the called party destination of 8002738255 and not 988. This is to comply with the current FCC requirements that 988 calls be translated by the OSP to 8002738255.

The OSP should include the ANI of the calling phone in the P-Asserted-Identify field.

The OSP should include the destination code in a X-Header field defined as:

X-988: xxxy…y

Where **xxx** defines the type of encoding to follow

xxx = 999 – Wire center based destination code.

In this format, defined by the encoding format of 999, the encoding format is followed by the 10-digit destination code. This destination code is constructed as defined in section 3.0 above.

X-988: 999NPANXXX000

Where the NPANXXX000 is the specific destination code of the calling device. The destination could should only be used for routing and should not be passed on to the terminating terminal.

xxx = 998 – PSAP ID with Wire center based destination code

In this format, defined by the encoding format of 998, the encoding information is followed by the 5-digit FCC PSAP ID, as defined in section 4.0 above, followed by the 10-digit destination code. The PSAP ID may be passed on to the terminating terminal. The destination code should only be used for routing and should not be passed on to the terminating terminal.

5.1.2 Failure Routing

It is the goal that all calls are routed to the 988 network utilizing the main routing methodology but failures in the cell site determination, destination code determination, SIP trunks to the 988 network and other failures may occur that make main routing not possible. In this case, the 988 calls should be routed, over the PSTN, to 800-273-8255 without including the X-HEADER parameter.

5.1.3 Heartbeat Mechanism

The heartbeat mechanism is implemented in the SBC. SIP Options are configured to be sent between each SBC on each side. Using this mechanism, the OSP doesn't have to program SIP Options into the interface; it is handled via configuration in the SBC. Using the SBC will also remove routes that are not responding. This will allow for fast route advance in the SBC or the return of a 503 to the OSP, so that the

OSP may route advance. The SIP Options usage in the SBC is also used to verify network-to-network connectivity on turn-up.

5.2 988 SIP Examples

In this SIP example, a 988 call is being sent from an OSP to the 988 network. An example SIP message is shown below. The specific headers of interest are denoted. The highlighted headers will have an implementation discussion below.

```
Summary:
                            Request: Invite
                        Request URI: sip:8002738255@75.76.77.167;user=phone
                            Call-ID: SDo599801-7e55a9cf30ca6e78598e5fe2bbcbc57c-
v300g00010
                                 To: 8002738255<sip:
8002738255@75.76.77.167:5060;user=phone>
                               From: "John Doe" <sip:+13035000499;verstat=TN-
Validation-Passed@64.65.66.117:5204;user=phone>;tag=SDo599801-648759c1-
648a226b3794d6d7-Soo-lucentNGFS-005820
                               CSeq: 1 INVITE
                            Contact: <sip:lucentNGFS-005820;tgrp=MTA03;trunk-
context=orig.att.com@64.65.66.117:5204;transport=udp>;+g.3gpp.icsi-ref="urn%3Aurn-
7%3A3gpp-service.ims.icsi.mmtel";+g.3gpp.mid-call;+g.3gpp.srvcc-
alerting;+g.3gpp.ps2cs-srvcc-orig-pre-alerting;video;text;orig-tas=a
              P-Access-Network-Info: 3GPP-E-UTRAN;utran-cell-id-
3gpp=311180ff7cf46e200;local-time-zone="UTC-07:00";daylight-saving-time="01";network-
provided
                P-Asserted-Identity: "John Doe" <tel:+13035000499; oli=62;verstat=TN-
Validation-Passed>
                P-Asserted-Identity: "John Doe" <sip:+13035000499;oli=62; verstat=TN-
Validation-Passed@64.65.67.117;user=phone>
                             SDP IP: 64.65.66.117
                           SDP Port: 50842

Detail Data:

---- SIP ----
INVITE sip: 8002738255@75.76.77.167;user=phone SIP/2.0
Via: SIP/2.0/UDP 64.65.66.117:5204;branch=z9hG4bK9otqc2302glhqf3a48c0.2
Call-ID: SDo599801-7e55a9cf30ca6e78598e5fe2bbcbc57c-v300g00010
To: 8002738255<sip: 8002738255@75.76.77.167:5060;user=phone>
From: "John Doe" <sip:+13035000499;verstat=TN-Validation-
Passed@64.65.66.117:5204;user=phone>;tag=SDo599801-648759c1-648a226b3794d6d7-Soo-
lucentNGFS-005820
CSeq: 1 INVITE
Accept: application/sdp
Accept: application/3gpp-ims+xml
Accept-Contact: *;+g.3gpp.icsi-ref="urn%3Aurn-7%3A3gpp-service.ims.icsi.mmtel"
Allow: INVITE, BYE, REGISTER, ACK, OPTIONS, CANCEL, SUBSCRIBE, NOTIFY, PRACK, INFO,
REFER, UPDATE
Contact: <sip:lucentNGFS-005820;tgrp=MTA03;trunk-
context=orig.ocn.com@64.65.66.117:5204;transport=udp>;+g.3gpp.icsi-ref="urn%3Aurn-
7%3A3gpp-service.ims.icsi.mmtel";+g.3gpp.mid-call;+g.3gpp.srvcc-
alerting;+g.3gpp.ps2cs-srvcc-orig-pre-alerting;video;text;orig-tas=att
Content-Type: application/sdp
Max-Forwards: 64
P-Asserted-Identity: "John DOe" <tel:+13035000499; verstat=TN-Validation-Passed>
P-Asserted-Identity: "John DOe" <sip:+13035000499;oli=62;verstat=TN-Validation-
```

```
Passed@64.66.67.117;user=phone>
P-Early-Media: supported
User-Agent: SM-G960U-G960USQU7DTA8 Samsung IMS 6.0
Content-Length: 695
P-Com.Nokia.B2BUA-Involved: yes
P-NokiaSiemens.OriginatingServiceData: reg-id=WTM1686773441049973
X-ALU-Trace:
cfedport=[2001:1890:1001:2a32::22:14]:5060;remoteport=[2001:1890:1001:2a32::21:8]:5092
;transport=UDP
P-Attestation-Indicator: A
P-Origination-Id: 2FB71233-D177-4DAA-9B34-E343EA35473F
Supported: 100rel, replaces, resource-priority
P-Served-User: <tel:+3035000499>;sescase=orig;regstate=reg
P-NokiaSiemens.Default-IMPU: <tel:+13035000499>
P-com.Siemens.Access-Information: GGSN
X-AS-URI: <sip:ebc1.wa1il.uvp.itn.att.net;forced-breakout=true;destination=SIP>
P-com.Siemens.Corr-ID: c72f4ce6ea52a2f63bfec010ff60f278,LU-1686774379932497-1515@stas-
tafe.fsgrp-016.dpa2actsf5001v.la1il.uvp.itn.att.net
X-IPV4-ROUTE: <sip:12.13.14.215:5060;lr>
X-Call-ID: LU-1686774379932497-1515@stas-tafe.fsgrp-
016.dpa2actsf5001v.la1il.uvp.itn.att.net
X-988: 9993604360000

v=0
o=LucentFS5000 267183073 267183074 IN IP4 64.58.53.117
s=-
c=IN IP4 64.65.66.117
t=0 0
m=audio 50842 RTP/AVP 116 107 97 115 0 101 111 100
b=AS:80
b=RS:612
b=RR:1837
a=rtpmap:116 AMR-WB/16000
a=fmtp:116 mode-set=0,1,2; mode-change-capability=2;max-red=0
a=rtpmap:107 AMR-WB/16000
a=fmtp:107 mode-set=0,1,2; octet-align=1;mode-change-capability=2;max-red=0
a=rtpmap:97 AMR/8000
a=fmtp:97 mode-change-capability=2;max-red=0
a=rtpmap:115 AMR/8000
a=fmtp:115 octet-align=1;mode-change-capability=2;max-red=0
a=rtpmap:101 AMR-WB/16000
a=rtpmap:111 telephone-event/16000
a=fmtp:111 0-15
a=rtpmap:100 telephone-event/8000
a=fmtp:100 0-15
a=sendrecv
a=maxptime:80
a=ptime:20
```

Invite:

The Invite should route the call to a similar destination of `INVITE sip: 8002738250` It is important that the call route to the destination of 8002738255 to comply with the current FCC Report and Order FCC 20-100. The TO: header will also typically contain the same SIP destination.

From:
P-Asserted-Identity:

These headers convey the TN(s) associated with call. The only required header, from the perspective of this specification, is From. In calls from a simple mobile line, there is only one (1) TN. The OSP may (probably should) populate the P-Asserted-Identity and the From. The OSP may also just populate the From (and not have a P-A-I). Which headers get populated by the OSP is a function of the capabilities of their SIP software. From a SIP perspective, the precedence of headers is P-Asserted-Identity and then the From header. From the perspective of this spec, P-Asserted-Identity should always contain the true call back number. If the P-A-I is omitted, then the From should contain the true callback number.

The recommended codec is G.711 (PCMU).

There are the other standard SIP headers (Via, To, CSeq, etc.) that are required for a canonical SIP message. Those are SIP requirements, not OSP ingress requirements. The 988 network also realizes that the OSP's SIP message software may have other headers and tags inserted at various parts of the SIP message. In general, the 988 network will attempt to ignore headers/tags our SIP software does not require/understand.

X-988: This header contains the destination code. If the destination code is made up of a 10-digit number that is derived from a wire center NPA-NXX then the X-Header should start with 999 (indicating this is a wire center destination code) followed by the 10-digit destination code. -This example shows a destination code of 3604360000. Prefixes other than 999 are reserved for future use.

6.0 SIP Error Codes

Following are the expected set of error codes the OSP can expect to be sent in the event of a processing error. If a SIP Error Code is received by the OSP, then some kind of incident is in progress. The OSP should invoke a notification process associated with an outage.

408 – Timeout: In this case, there is probably a network failure that did not deliver the OSP Invite to the 988 network. The OSP could try other pathways to deliver the call. If all attempts result in a 408 SIP Error Code, then the OSP is isolated from the 988 network. The OSP would then use their failure routing.

503 – Service Unavailable: In this case, the 988 network has received the OSP SIP Invite. Some kind of processing error has occurred, and the call could not be delivered to any destination. The OSP should try other pathways to the 988 network. If all of these attempts fail, then the OSP would then use their failure routing.

All other SIP Error codes are possible (but not probable). The OSP would attempt alternate pathways to the 988 network. If all Pathways result in error, then the OSP would then use their failure routing.

7.0 Appendix A: Network Interconnection Notes

IP traffic between the originating network and the 988 network should use dedicated network connections.

To allow for redundancy, the OSP may choose to deploy any number of origination hosts to connect to the appropriate 988 network destinations. In addition, each SIP trunk group will be provisioned with a unique value for the maximum number of concurrent call paths that will be supported.

There are a number of requirements for the network configuration between an originating service provider and the 988 network, primarily driven by the high-availability goals for critical services. The following is a list of the basic Network-to-Network (NNI) networking requirements.

- IPv6 at the NNI will be supported as customer demand develops.
- IPv4 is supported based upon bilateral agreements.
- Each SIP trunk group to the 988 network is associated with a single originating service provider. A single OSP is required to have two geographically diverse Points of Interconnection (POIs) into the 988 network. Each POI will have the capability to fail over to the other POI during error conditions or when calls are rejected.
- The OSP network will load balance between POIs.

8.0 Appendix B: Definitions, Abbreviations, and Acronyms

8.1 Definitions

SIP Trunk Group – Similar to a traditional TDM trunk group but carries SIP traffic. SIP trunk groups reside between two IP endpoints and ports that pass SIP messages.

8.2 Abbreviations and Acronyms

Term	Definition
ANI	Automatic Number Identification
BCF	Border Control Function
CCP	Concurrent Call Paths
CLLI	Common Language Location Identification
CLR	Circuit Layout Record
DLR	Design Layout Record
DNS	Domain Name System
DTMF	Dual Tone Multi Frequency
FQDN	Fully Qualified Domain Name
HELD	HTTP Enabled Location Delivery
HA	High Availability
HTTP	Hyper Text Transfer Protocol
HTTPS	Hyper Text Transfer Protocol Secure
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
LERG	Local Exchange Routing Guide
NENA	National Emergency Number Association
OSP	Originating Service Provider
PIF	Protocol Interwork Function
PCMU	Pulse Code Modulation µ-Law

Term	Definition
PSAP	Public Safety Answering Point
RFC	Request for Comments
RTP	Real Time Protocol
RTT	Real Time Text
SBC	Session Border Controller
SIP	Session Initiation Protocol
SRTP	Secure Real-time Transport Protocol
SS7	Signaling System 7
SSL	Secure Sockets Layer
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
TN	Telephone Number
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
URN	Uniform Resource Name
UTC	Coordinated Universal Time (aka Greenwich Mean Time (GMT))
VoIP	Voice over Internet Protocol
XML	eXtensible Markup Language