ATIS-0300106

ATIS Standard on -

Intercarrier Call Completion/Call Termination Handbook

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Intercarrier Call Completion/Call Termination Handbook

Alliance for Telecommunications Industry Solutions

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Abstract

This handbook describes some of the problems being encountered by rural telephone service provider (SP) customers in receiving long distance calls. It discusses some of the industry standards and practices relevant to ensuring call completion, particularly signaling, routing, and trouble handling. This handbook attempts to relate these standards and practices to the call completion problems reported and offers some best practices for ensuring call completion. This handbook provides a resource to SPs to address issues as they are encountered related to long distance call completion/call termination.

Foreword

The Alliance for Telecommunication Industry Solutions (ATIS) serves the public through improved understanding between carriers, customers, and manufacturers. The Next Generation Interconnection Interoperability Forum (NGIIF) addresses next-generation network interconnection and interoperability issues associated with emerging technologies. Specifically, it develops operational procedures that involve the network aspects of architecture, disaster preparedness, installation, maintenance, management, reliability, routing, security, and testing between network operators. In addition, the NGIIF addresses issues that impact the interconnection of existing and next generation networks and facilitate the transition to emerging technologies.

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 an 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, Next Generation Interconnection Interoperability Forum (NGIIF), 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, Next Generation Interconnection Interoperability Forum (NGIIF), which was responsible for its development, had the following leadership:

Karen Riepenkroger, NGIIF Co-Chair, Sprint

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1 Scope, Purpose, & Application

1.1 Scope

Call completion/call termination in today's Public Switched Telephone Network (PSTN)[1] depends on coordination between different service provider (SP) entities, each playing their part in setting up a workable connection between calling and called parties. As the PSTN has evolved through the Bell System divestiture, the Telecom Act of 1996, and the introduction of Internet Protocol (IP)-based technologies, the number and diversity of these entities has grown. In this context, problems with call completion may arise.

This handbook:

- Is a living document, which will be updated as applicable
- Describes some of the problems being encountered
- Discusses various industry standards and practices relevant to ensuring call completion, particularly signaling, routing, and trouble handling
- Attempts to relate these standards and practices to the call completion problems reported
- Offers best practices for ensuring call completion, especially in the management of intermediate SPs

1.2 Purpose

This handbook provides a resource to SPs to address issues as they are encountered related to intercarrier LD call completion/call termination.

1.3 Application

1.3.1 Problems Reported

Customers of some telecommunications SPs (particularly those in rural areas) have experienced difficulties with the receipt of LD calls via their phone service, including problems that generally fall into the following categories:

- Call completion failure
  - Failure scenarios reported included:
    - The Calling Party hears ringing but the Called Party hears nothing (no ringing)
    - The Called Party's phone rings, but the Called Party hears nothing when the call is answered, i.e., "dead air"
    - The Calling Party hears local busy tone (when the line was not busy)
    - The Calling Party hears fast or network busy, or hears a network failure announcement including inappropriate "number not in service"

- Very long post dial delay

- Poor transmission quality
  - Both voice and fax

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[1] PSTN is used here to refer to the set of networks used to complete calls using E.164 number addressing.

- Misidentification of Calling Party

1.3.2 Call Completion Components

In the *Report and Order* (*R&O*) and *Further Notice of Proposed Rulemaking* (*FNPRM*) in FCC 13-135 and WC Docket No.13-39, adopted October 28, 2013 and released November 8, 2013 ("RCC Order"), the FCC defined the term "answered call"[2] to mean:

> "A call that was answered by or on behalf of the called party (including calls completed to devices, services or parties that answer the call, such as an interactive voice response, answering service, voicemail or call-forwarding system), causing the network to register that the terminating party is prepared to receive information from the calling user."

Also, as a result of the Final Rules issued in the RCC *R&O*, the term "call attempt"[3] means "a call that results in transmission by the covered SP toward an incumbent local exchange carrier (LEC) of the initial call setup message, regardless of the voice call signaling and transmission technology used."[4]

With respect to call completion, a call attempt can be signaled either as Answered, Busy, Ring No Answer, or Unassigned Number.[5]

2 References

The following standards contain provisions which, through reference in this text, constitute provisions of this Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

ATIS-1000002, *Number Portability Switching Systems*.[6]

ATIS-1000113, S*ignaling System No. 7 (SS7) – Integrated Services Digital Network (ISDN) User Part*.[6]

ATIS-1000607, *Integrated Services Digital Network (ISDN) – Layer 3 Signaling Specification for Circuit Switched Bearer Service for Digital Subscriber Signaling System Number 1 (DSS1)*.[6]

ATIS-1000679, *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control or ISDN User Part*.[6]

ATIS-0300009, *NGIIF Reference Document Part I- Installation and Maintenance Responsibilities for Special Access Services, WATS Access Lines, and Switched Access Services Feature Group "A"*.[6]

ATIS-0300010, *NGIIF Reference Document Part II- Installation and Maintenance Responsibilities for Switched Access Services Feature Groups "B," "C," and "D"*.[6]

ATIS-0300011, *NGIIF Reference Document Part II- Installation and Maintenance Responsibilities for Switched Access Services Feature Groups "B," "C," and "D"*.[6]

ATIS-0300012, *NGIIF Reference Document Part III- Attachment A- MTP Compatibility Tests*.[6]

ATIS-0300013, *NGIIF Reference Document Part III- Attachment B- ISUP Compatibility Tests*.[6]

ATIS-0300014, *NGIIF Reference Document Part III- Attachment C- SCCP Protocol Class 0 Compatibility Tests*.[6]

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[2] FCC 13-135, paragraph 72 and FCC 13-135 footnote 114.

[3] 47 CFR § 64.2101 (b).

[4] Appendix A to FCC 13-135, 47 CFR §64.2103, *Retention of Call Attempt Records*.

[5] Appendix A to FCC 13-135, 47 CFR § 64.2105, *Reporting Requirements*.

[6] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS) at < https://www.atis.org/docstore/ >.

ATIS-0300015, *NGIIF Reference Document Part III- Attachment D- Test Severity Analysis Criteria.*[6]

ATIS-0300016, *NGIIF Reference Document Part III- Attachment E- SS7 Network Gateway Screening.*[6]

ATIS-0300017, *NGIIF Reference Document Part III- Attachment F- SS7 ISUP Tests for ISDN Network Interconnection*[6]

ATIS-0300018, *NGIIF Reference Document Part III- Attachment G- SS7 Link Diversity Validation Guidelines.*[6]

ATIS-0300019, *NGIIF Reference Document Part III-Attachment H SS7 Cause Codes and Tones and Announcements.*[6]

ATIS-0300020, *NGIIF Reference Document Part III- Attachment I- SS7 Security Base Guidelines.*[6]

ATIS-0300021, *NGIIF Reference Document Part III- Attachment J- SS7 Software Validation.*[6]

ATIS-0300022, *NGIIF Reference Document Part III- Attachment K- Dual STP Failure Prevention Procedures.*[6]

ATIS-0300023, *NGIIF Reference Document Part IV- Installation and Maintenance Responsibilities for X.75 Gateway Services.*[6]

ATIS-0300024, *NGIIF Reference Document Part V- Test Line Guidelines.*[6]

ATIS-0300030, *NGIIF Reference Document Part IX- Installation, Testing, and Maintenance Responsibilities for Facilities.*[6]

ATIS-0300032, *Next Generation Interconnection Interoperability (NGIIF) Reference Document: Part X, Interconnection Between LECS Operations Handbook – Local Interconnection Service Arrangement.*[6]

ATIS-0300035, *NGIIF Reference Document Part XII- Toll Free Industry Test Plan.*[6]

ATIS-0300046, *Recommended Notification Procedures to Industry for Changes in Access Network Architecture.*[6]

ATIS-0300082, Guidelines for Reporting Local Number Portability Troubles in a Multiple Service Provider Environment.[6]

ATIS-0300105, *Next Generation Interconnection Interoperability Forum (NGIIF) Auto Dialers Reference Document.*[6]

ATIS-0300119, *Thousands-Block (NPA-NXX-X) & Central Office Code (NPA-NXX) Administration Guidelines.*[6]

ATIS-0300209, *Operations, Administration, Maintenance and Provisioning (OAM&P) – Network Tones and Announcements.*[6]

IETF RFC 3261, *SIP: Session Initiation Protocol.*[7]

IETF RFC 3325, *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks.*[7]

3GPP TS 24.229, *IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3.*[8]

GR-905-CORE, *Common Channel Signaling Network Interface Specification (CCSNIS) Supporting Network Interconnection, Message Transfer Part (MTP), And Integrated Services Digital Network User Part (ISDNUP).*[9]

GR-317-CORE, *LSSGR: Switching System Generic Requirements For Call Control Using The Integrated Services Digital Network User Part (ISDNUP).*[9]

GR-394-CORE, *LSSGR: Switching System Generic Requirements For Interexchange Carrier Interconnection (ici) Using The Integrated Services Digital Network User Part (ISDNUP).*[9]

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[7] This document is available from the Internet Engineering Task Force (IETF) at < http://www.ietf.org >.

[8] This document is available from the Third Generation Partnership Project (3GPP) at < http://www.3gpp.org/specs/specs.htm >.

[9] This document is available from the Telcordia Technologies at < http://telecom-info.telcordia.com >.

ITU-T E.721, *Network grade of service parameters and target values for circuit-switched services in the evolving ISDN.*[10]

FCC Report and Order and FNPRM, In the Matter of Rural Call Completion (WC Docket No.13-39) (FCC 13-135).[11]

Report and Order and Further Notice of Proposed Rulemaking, Connect America Fund (WC Docket No. 10-90), A National Broadband Plan for Our Future , (GN Docket No. 09-51), Establishing Just and Reasonable Rates for Local Exchange Carriers (WC Docket No. 07-135), High-Cost Universal Service Support (WC Docket No. 05-337), Developing an Unified Intercarrier Compensation Regime (CC Docket No. 01-92), Federal-State Joint Board on Universal Service (CC Docket No. 96-45), Lifeline and Link-Up (WC Docket No. 03-109), Universal Service Reform Mobility Fund (WT Docket No. 10-208) (FCC 11-161).[11]

Declaratory Ruling, In the Matter of Developing an Unified Intercarrier Compensation Regime (CC Docket No. 01-92) and Establishing Just and Reasonable Rates for Local Exchange Carriers, (WC Docket No. 07-135) (DA 12-154).[11]

First Report and Order, In the Matter of Implementation of the Local Competition Provisions in the Telecommunications Act of 1996 (CC Docket 96-98) and Interconnection between LECs and Commercial Mobile Radio Service (CMRS) Providers, (CC Docket No. 95-185) (FCC 96-325).[11]

Report and Order, In the Matter of Rules and Regulations Implementing the Truth in Caller ID Act of 2009, (WC Docket No. 11-39) (FCC 11-100).[11]

Perlman, Radia, *Interconnections: Bridges, Routers, Switches, and Internetworking Protocols.* Addison-Wesley, 2000.

3 Definitions, Acronyms, & Abbreviations

3.1 Definitions

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

3.2 Acronyms & Abbreviations

3GPP	3rd Generation Partnership Project
ACM	Address Complete Message
ANI	Automatic Number Identification
ANSI	American National Standards Institute
AOCN	Administrative Operating Company Number
ASC	Access Service Customer
ASP	Access Service Provider
ASR	Answer Seizure Ratio (also called Call Answer Rate)
ATIS	Alliance for Telecommunications Industry Solutions
BICC	Bearer-Independent Call Control
BIRRDS	Business Integrated Routing and Rating Database System
BT	Busy Tone
CDR	Call Detail Records
CLEC	Competitive Local Exchange Carrier
CMRS	Commercial Mobile Radio Service

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[10] This document is available from the International Telecommunications Union. < http://www.itu.int/ >

[11] This document is available from the FCC at < https://www.fcc.gov/ >.

CN	Charge Number
CPC	Calling Party's Category
CPE	Customer Provided Equipment (also Customer Premises Equipment)
CPN	Calling Party Number
CPNI	Customer Proprietary Network Information
DMoQ	Direct Measures of Quality
DSS1	Digital Subscriber Signaling System Number 1
EAS	Extended Area Service
ENS	Emergency Notification System
ETS	Emergency Telecommunications Service
FCC	Federal Communications Commission
FG	Feature Group
I-IWU	Incoming Interworking Unit
IAM	Initial Address Message
ICC	Intercarrier Compensation
ICN	Interconnecting Networks
ID	Identification
IETF	Internet Engineering Task Force
INC	Industry Numbering Committee, ATIS Committee
INT	Intercept
IP	Internet Protocol
IPM	Impulses per Minute
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
ITU	International Telecommunication Union
ITU-T	International Telecommunication Union Telecommunication Standardization Sector
IXC	Interexchange Carrier
kbps	kilobit per second
kHz	kilohertz
LAN	Local Area Network
LATA	Local Access and Transport Area
LD	Long Distance
LEC	Local Exchange Carrier
LNP	Local Number Portability
LRN	Location Routing Number
MF	Multi-Frequency
MOU	Minutes of Use
MTP	Message Transfer Part
NANP	North American Numbering Plan
NANPA	North American Numbering Plan Administration
NCA	No Circuit Announcement
NER	Network Effectiveness Ratio
NGIIF	Next Generation Interconnection Interoperability Forum, ATIS Committee

NGN	Next Generation Network
NPA	Numbering Plan Area
NPAC	Number Portability Administration Center
NXX	Central Office Code – 3 digits following the NPA in a World Zone 1 TN
OAM&P	Operations, Administration, Maintenance and Provisioning
OLI	Originating Line Information
PBX	Private Branch eXchange
POI	Point of Interface
POS	Packet Over SONET
POT	Point of Termination
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
REL	Release Message
RCC	Rural Call Completion
ROTL	Remote Office Test Lines
RI	Redirection Information
RO	Reorder Tone
ROA	Reorder Announcement
SAS	Switched Access Services
SCPS	Signal Control Points
SDP	Session Description Protocol
SIM	Subscriber Identity Module
SIP	Session Initiated Protocol
SP	Service Provider
SPCD	Service Provider Contact Directory
SS7	Signaling System No. 7
TBCOCAG	Thousands-Block (NPA-NXX-X) & Central Office Code (NPA-NXX) Administration Guidelines
TDD	Telecommunications Devices for the Deaf
TDM	Time Division Multiplexing
TN	Telephone Number
UDI	Unrestricted Digital Information with Tone and Announcement
USF	Universal Service Fund
VCA	Vacant Code Announcement
VoIP	Voice over Internet Protocol
WATS	Wide Area Telephone Service

4 General Standards and/or Guidelines

General standards and/or guidelines should be employed by SPs to help maintain proper network function and to detect and respond to RCC/call termination issues.

SPs should implement quality processes to ensure basic functions of network maintenance, modernization, and repair do not introduce failures into the network, and have procedures for all activity that touches the network. Network alarming should be in place to promptly alert SPs to service-affecting events.

4.1 General Network Management Practices

Given billions of call events annually, even low levels of call completion failures related to process gaps or technical problems could overwhelm investigative resources and detract from efforts to identify and address catastrophic/systemic problems. SPs should continue to implement quality processes to ensure that basic functions of network maintenance, modernization, and repair do not introduce failures into the network. In that respect:

- SP should have written procedures for all activity that touches the network

- Devices, software, and configurations should be validated in lab environments before being introduced into production

- Implementation procedures should be documented and should include verified back-out procedures to ensure ability to revert to last known good operating environment in the event the implementation does not go as planned

- Network alarming should be in place to promptly alert SPs to service-affecting events

- Network metrics should be monitored to ensure performance within intended operating parameters

4.2 Metrics for Monitoring Performance

Network metrics should be monitored to ensure performance within intended operating parameters. Some SPs monitor call completion rates using a variety of measurements, such as Answer Seizure Ratio (ASR) and Network Effectiveness Ratio (NER). Performance of these metrics over various periods of time may be useful to help spot trends, persistent issues, and sudden deviations from baseline performance, any of which might merit further attention depending on the circumstances.

Metrics may be less useful if the terminating network cause codes are incorrect in accordance with GR-905, *Common Channel Signaling Network Interface Specification (CCSNIS) Supporting Network Interconnection, Message Transfer Part (MTP), And Integrated Services Digital Network User Part (ISDNUP)*. For example, in response to an Initial Address messages (IAM), calls to unallocated TNs should always receive an immediate cause code of 1 in either the Address Complete message (ACM), or the Release (REL) message. If the SP chooses to respond with a cause code of 1 in an ACM message, the SP will also be expected to play the recorded announcement indicating the number called is a non-working TN. In this scenario, the call path will stay connected to the SPs switch until the originating party hangs up, or until the SPs announcement timer expires whereby the disconnect will be initiated by the SPs switch. If the SP chooses to respond to the IAM with a cause code of 1 in a REL message, the call path between the IXC and the SP will be torn down immediately, thus saving network resources, and the IXC or the originating SP will play the recorded announcement. When the correct cause code is not provided by a SPs EO, call attempts to unallocated numbers may be incorrectly categorized as call failures, thus negatively affecting metrics calculations. In addition, calling parties that are able to modify their calling lists based on cause code information may continue to place calls to the same unallocated TN in the absence of receipt of an appropriate cause code.

4.2.1 Answer Seizure Ratio (ASR)

ASR (also known as Call Answer Rate) is calculated by taking the number of successfully answered calls and dividing by the total number of calls attempted. In some implementations, SPs attempt to adjust the formula for calls to unallocated numbers. The formula for determining ASR admittedly includes unanswered calls that are in fact successfully completing on the terminating end (i.e. ring no answer, user busy or fail with the terminating LEC). However, this data point may be less susceptible to variations in data reporting or to differences in the quality or accuracy of signaling; the Called Party either answered the call or it did not.

ASR % = (Total number of answered calls / total number of attempted calls) x 100

4.2.2 Network Effective Ratio (NER)

NER expresses the relationship between the number of seizures and the sum of the number of seizures resulting in either an answer message, or a user busy, or a ring no answer, or, in the case of Integrated Services Digital Network (ISDN), a terminal rejection/unavailability. Unlike ASR, NER attempts to exclude the effects of customer

behavior and terminal behavior. Some SPs use NER instead of ASR to better identify network behavior that may merit further investigation.

NER % = (answers + user busy + ring no answer + terminal rejects) / number of total call attempts (seizures)

5 Applicable Standards/Guidelines for Certain Root Causes

This section identifies some of the existing applicable standards and/or guidelines for certain root causes which may have relevance to issues related to RCC/call termination.

5.1 Signaling

5.1.1 Identification of Calling Party

The origin of a call can be identified in signaling through several parameters. In Signaling System No. 7 (SS7) ISUP, the Calling Party Number (CPN) parameter is carried in the Initial Address Message (IAM) and contains the Telephone Number (TN) of the originating end user (see ATIS-1000113.2015-Chapter 2, Clause 2.16, *Calling Party Number*, and ATIS-1000113.2015-Chapter 3, Clause 3.7, *Calling Party Number*).

CPN is the parameter that determines what the Called Party sees as the caller identification (ID), the TN directly, and caller name, based on database look-up of the number in the CPN. In general, CPN is propagated from the originating switch to the terminating switch as long as signaling is end-to-end SS7. If there is inband, i.e., Multi-Frequency signaling (MF) in the call path, CPN will not be received at the terminating switch.

In Session Initiated Protocol signaling (SIP), the phone number of the calling end user is populated in P-Asserted-Identity header (RFC 3325)[12]. ATIS-1000679, *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control or ISDN User Part,* defines interworking between SS7 ISUP and SIP, including a mapping between the SS7 CPN parameter and the SIP P-Asserted-Identity header[13].

The NGIIF recommends that the CPN field should be populated, by the originating network, with a valid 10-digit North American Numbering Plan (NANP) subscriber line number or directory number. More information can be found in Clause 8, *Regulatory Environment*.

5.1.2 Missing Caller ID

The problem of missing caller ID (i.e., no number available) as seen by the Called Party will result when CPN is not delivered to the terminating exchange. This could result if: 1) no CPN was populated at origination; 2) there was inband signaling on the path; 3) SIP-SS7 interworking was not handled properly; or 4) CPN was removed by an entity in the call path. Missing CPN may interfere with terminating the call (e.g., if anonymous call rejection is engaged). Additional information can be found in Clause 8, *Regulatory Environment*.

5.1.3 Incorrect Caller ID

The issue of incorrect caller ID, as seen by the Called Party, may result when a CPN, other than that normally associated with the Calling Party, is delivered to the terminating end office. This may be the result of an entity in the call path explicitly manipulating the CPN, or of a call being terminated and then re-originated in the process of routing to the terminating end office. (In the latter case, CPN may be a number associated with the network element that re-originates the call. Note that this differs from call forwarding, which retains the original CPN information in the Calling Party Address parameter in the IAM.) Changes in the CPN delivered may also interfere with terminating the call. It should be noted, while the Called Party may view it as incorrect, the caller ID may be different than the CPN due to a legitimate change to the TN based on the FCC's allowed practices. Additional information can be found in Clause 8, *Regulatory Environment*.

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[12] If the P-Asserted-Identity header is not present, CPN may sometimes be populated based on the From header. This may occur when interworking does not support P-Asserted Identity, but is not the preferred situation.

[13] Note that ISUP parameters may also be encapsulated in SIP messages and thus passed to subsequent SS7 elements, although this approach does not appear to be widely implemented.

5.1.4   Identification of the Chargeable Party

SS7 IAM may also include identification of the chargeable party for the call. When the number to be charged differs from the CPN, it is carried in the (IAM) in the Charge Number (CN) parameter (see ATIS-1000113.2015-Chapter 2, Clause 2.25A, *Charge Number*, and ATIS-1000113.2015-Chapter 3, Clause 3.10, *Charge Number*). Standards provide that where the Charge Number is the same as the CPN, the originating exchange can signal just the CPN and a valid Originating Line Information (OLI) parameter (ATIS-1000113.2015, Clause 2.1.9.3A, *Charge Information*). In SIP, the mechanism for population of the comparable information is not yet standardized. Therefore, an ISUP-SIP mapping has not been standardized in ATIS-1000679. Work under development in the Internet Engineering Task Force (IETF) for a P-Charge-Info header [draft-york-sipping-p-charge-info-12 (2011-09-15)] may form a basis for this in the future. Note also that a SIP mechanism corresponding to the SS7 OLI parameter has not been finalized. See the expired IETF work in draft-patel-dispatch-cpc-oli-parameter-03. Note that while the Calling Party's Category (CPC) portion of the IETF work is incorporated into the definition found in in Annex J of 3GPP TS 24.229, *IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3*, the OLI portion is not.

Where signaling from an originating exchange is inband (MF), a billing number (ANI) can be signaled (e.g., from a LEC to an IXC). ANI can be mapped into CN but is *not* mapped into CPN.

5.1.5   Cause Codes, Tones, & Announcements

Cause codes, tones, and announcements play an important role in call completion. They are key to the identification, and thus resolution of network problems, however, their misuse may exacerbate problems. SPs have reported instances in which a busy tone or number-not-in-service announcement has been delivered to callers, when in fact the number was in service and was not engaged. Such signals, when erroneously applied, not only mislead the caller but may mask call completion problems from detection by the caller's LD SP. This Clause provides guidance for the proper use of cause codes, tones, and announcements.

5.1.5.1 NGIIF SS7 Cause Code & Tones & Announcements

The purpose of this section is to provide information related to the application of cause codes with the associated treatment, and the appropriate verbiage to be played to the customer where an announcement is required. Complete documentation is available in ATIS-0300019, *NGIIF Reference Document Part III-Attachment H SS7 Cause Codes and Tones and Announcements.*

5.1.5.1.1   General Information

ATIS-0300019 describes the tones and announcements that are used to inform customers and network operators of various conditions that are encountered on dialed calls. Tones and announcements are also used for service analysis of conditions that result in failure to complete dialed calls. Analysis data is used to evaluate administrative, engineering, and maintenance efforts to improve service. Tones are used primarily to identify the condition of called lines and network blockage of failure conditions.

For some network blockage or failure conditions, announcements are used to provide customers or network operators with additional information and to suggest action that should be taken.

5.1.5.1.2   Cause Code Standards

There are three standards by which cause codes are utilized within the telecommunications industry with one being held in reserve; they are as follows:

00 = CCITT[14]

01 = OTHER INTERNATIONAL

10 = ANSI

11 = RESERVED

The Coding Standards are used in the subfields of the Cause Indicators Parameter Field:

Table 5.1 - Cause Indicators Parameter Field

8	7	6	5	4	3	2	1
Ext	Coding Standard		Spare	Location			
Ext	Recommendation						
Ext	Cause Value						

In some instances, an alpha character "N" is utilized in the Cause Code matrix to indicate the Cause Code as a national standard.

5.1.5.1.3 Mapping Matrix

The cause code mapping matrix located in ATIS-0300019 identifies the action and direction of the cause codes as they traverse the network.

5.1.5.1.4 Cause Code Classes & Cause Code Default

Any cause code not being recognized will be routed to a default code in the appropriate cause code class. Cause codes should only be mapped to the applicable default code for that class.

Table 5.2 - Cause Code Classes and Default Codes

CAUSE CODE CLASS	CAUSE CODE GROUPING DESCRIPTION	CAUSE CODE GROUPS	CAUSE CODE DEFAULT
001	Normal Event	1-30	31
010	Resource Unavailable	32-46	47
011	Service Option not Available	48-62	63
100	Service Option not Implemented	64-78	79
101	Invalid Message	80-94	95
110	Protocol Error	96-110	111
111	Interworking	112-126	127

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[14] Also known as the International Telecommunication Union Telecommunication Standardization Sector (ITU-T).

5.1.5.1.5   Cause Code Treatments

This Clause provides the definition of the different announcements that can be applied to the appropriate cause codes.

ATIS-0300019 details the circumstances under which different cause codes should be provided and the corresponding tones or announcements to be provided. Adhering to these procedures will address problems such as those reported by the   SPs; for example, incorrect provision of busy tone or number-not-in-service announcements which may mislead callers about the status of the Called Party, and hinder the SP maintaining the retail LD relationship with caller from detecting call completion problems.

5.1.5.1.5.1   **No Circuit Announcement (NCA)**

This announcement is played when there are no circuits available for the call to be completed. An example of such a recording is as follows:

> *"All circuits are busy now. Please try your call again later".*

5.1.5.1.5.2   **Reorder Announcement (ROA)**

This announcement is played when a call did not traverse the network to completion for a myriad of reasons. An example of such a recording is as follows:

> *"We're sorry, your call did not go through. Please try your call again later".*

> > NOTE: A reorder tone (also known as "fast busy") may sometimes be applied in lieu of a reorder announcement.

5.1.5.1.5.3   **Intercept (INT)**

This announcement is played when a customer dials a number that has been disconnected or is no longer in service. An example of such an announcement is:

> *"We're sorry, you have reached a number that has been disconnected or is no longer in service. If you feel you have reached this recording in error, please check the number and try your call again".*

5.1.5.1.5.4   **Vacant Code Announcement (VCA)**

This announcement is played when a customer misdials a TN in some manner, including when the dialed number appears to be a proper TN, but is an unassigned (vacant) number. "Misdialing" (entering incorrect digits than intended) a TN that has been assigned should result in call completion but would be considered dialing "the wrong number". An example of an announcement in regard to dialing an invalid or unassigned TN is:

> *"We're sorry; your call cannot be completed as dialed. Please check the number and dial again".*

5.1.5.1.5.5   **Ineffective Other**

There should be announcements with the appropriate verbiage for the following situations. Specific wording is determined by the SP.

- Prefix or access code dialing irregularity

- Improper initial coin deposit

- Screened line access denial

- Dialing irregularity

At a minimum, the announcements should include the following information:

- The call cannot be completed as dialed

- Instructions for correct dialing procedures

- The customer should try the call again (except for Customer Calling Feature Calling)

In addition to announcements, there are instances when the application of a tone is the applicable treatment for a call. The following clauses describe tones that are normally applied as treatments.

5.1.5.1.5.6 **Busy Tone (BT)**
Busy Tone is applied to an originating customer's line when the Called Party is engaged in another call or the phone is at an off-hook condition. The tone applied to the originating customer's line will be at a rate of 60 Impulses per Minutes (IPM).

5.1.5.1.5.7 **Reorder Tone (RO)**
Reorder tone (also known as "fast busy") is applied to the originating customer's line when the call cannot be completed, which may be due to insufficient facilities. In such instances, a tone will be applied at a rate of 120 IPM. This tone may be applied in lieu of an announcement.

5.1.5.1.6   **Mapping Matrix**
The following are examples of cause codes received by each network or CPE and the cause codes sent by each network or CPE. The action that generated the cause code is listed to indicate the network or CPE where the event occurred (YY and XX represent different cause codes):

- ➔ Indicates direction of cause code.

- ← Indicates direction of cause code.

Arrow facing towards cause code indicates cause code being generated.

Arrow facing away from cause code indicates cause code being passed.

At the end office, treat the call per cause code XX for analog or Non-ISDN call.

Table 5.3 - Examples of Cause Codes Received and Sent by ach Network or CPE

XX	(message)	Cause Code Generated by or Received at Network of CPE
XX←	(action/message)	Network or CPE generated cause code XX
➔XX	(action/message)	Network or CPE generated cause code XX
➔XX←	(action/message)	Network or CPE generated cause code XX in both directions
←XX	(action/message)	Cause code XX being passed to next network or CPE ⬜
XX➔	(action/message)	Cause code XX being passed to next network or CPE⬜
←XX➔	(action/message)	Cause code XX being passed in both directions
←YY←XX	(message)	Network or CPE receives cause code XX from (message) XX➔YY➔ previous network or CPE and maps cause code XX to cause code YY

5.1.5.2 **Other Applicable Standards Addressing Tones & Announcements**

5.1.5.2.1   **ATIS-0300209,** *Operations, Administration, Maintenance and Provisioning (OAM&P) – Network Tones and Announcements*

ATIS-0300209 addresses tones and announcements associated with ineffective call attempts. ATIS-0300209 addresses the following tones:

- Busy Tone.

- Reorder Tone.

- Special Information Tones.

Announcements addressed include:

- Reorder

- No Circuit

- Vacant Code

- Intercept

- Ineffective Other

ATIS-0300209 also addresses the mapping of Cause Indicator Values specified for the ISDN User Part (ISUP) and Digital Subscriber Signaling System Number 1 (DSS1) in ATIS-1000113 and ATIS-1000607, respectively, and the tones and announcements identified in this standard (ATIS-0300209). This mapping is for use in call processing involving two or more interconnecting networks (ICNs) when SS7 is used for call control. This standard considers the provision of these tones and announcements by originating, intermediate, and terminating ICNs.

5.1.5.2.2   ATIS-1000113, *Signaling System No. 7 (SS7) – Integrated Services Digital Network (ISDN) User Part*

Integrated Services Digital Network (ISDN) User Part defines the protocol that supports the signaling functions required to provide voice and non-voice services in an Integrated Services Digital Network. The messages and signals are defined in ATIS-1000113.2015-Chapter 2 and their format and content are contained in ATIS-1000113.2015-Chapter 3. ATIS-1000113.2015-Chapter 4 describes the basic signaling procedures for the set-up and clear-down of national and international ISDN and non-ISDN connections. This chapter also includes generic procedures for supplementary services. Service-specific procedures for supplementary services are contained in separate American National Standards Institute (ANSI) documents (see Clause 2.2 of ATIS-1000113.2015-Chapter 1).

5.1.5.2.2.1   Address Signaling

In general, the call set-up procedure described is standard for both voice and non-voice connections using enbloc address signaling for calls between ISDN terminals and non-ISDN terminals.

5.1.5.2.2.2   Basic Procedures

The basic call control procedure is divided into three phases: call set-up, data/conversation, and call cleardown. Messages on the signaling link are used to establish and terminate the different phases of a call. Standard inband supervisory tones or recorded announcements or both are returned to the caller on speech and 3.1 kiloHertz (kHz) connections to provide information on call progress. Calls originating from ISDN terminals may be supplied with more detailed call progress information by means of additional messages in the access protocol supported by a range of messages in the network.

5.1.5.2.2.3   Tones & Announcements, Basic Call Control, & Signaling Procedures

More information about basic call control and signaling procedures can be found in ATIS-1000113.2015, Chapters 2 and 3, and Chapter 4, Clause 2, *Basic Call Control & Signaling Procedures*. Specifically, Chapter 4, Clause 2 addresses successful call set-up and unsuccessful call set-up. With respect to unsuccessful call set-up, tones and

announcements are addressed for speech, 3.1 kHz audio, and 64 kilobit per second (kbps) Unrestricted Digital Information with Tone and Announcement (UDI-TA).

5.1.6 Interconnection Parameters & Looping

Passing certain SS7 parameters are crucial to the prevention of call loops in which a call cycles back and forth between networks without ever reaching its destination (this is further discussed in the Clause 5.3, *Routing*). Loop detection and control is important in such circumstances because, in addition to causing call failure, loops consume network capacity and also diminish the likelihood of completion of calls that do not involve looping.

5.1.6.1 Hop Counter

Interconnecting parties should exchange the initial value of the Hop Counter at the time of negotiation for interconnection and as changes are anticipated/made in signaling between the two networks.

If no Hop Counter is received with the SS7 incoming IAM, then if the Hop Counter capability is active, a non-forwarding transit exchange should include the Hop Counter parameter in the outgoing IAM. For technologies where the Hop Counter cannot be set on a per call type basis and for non-Emergency Telecommunications Service (ETS) calls, the network operator should set the initial count value within a range between 15 and 20. In addition, if the initial count value can be set by the network operator on a per-call type basis, then the initial count value for ETS calls should be the maximum value allowed in the exchange.

The value received should be accepted as a valid value and the recipient switch should not reinitialize the Hop Counter. For SIP-I, the Incoming Interworking Unit (I-IWU) acting as an independent exchange should perform the normal Bearer-Independent Call Control (BICC)/ISUP Hop Counter procedure using the Hop Counter taken from the encapsulated IAM.

ATIS-1000679, *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control or ISDN User Part*, Clauses 6.1.3.8 and 7.1.4, describe the interworking of the SS7 hop counter and the analogous SIP max forwards parameter.

Where multiple protocols are involved in a single call process due to Hop Counter and max forwards having no interworking between SS7 and SIP, Hop Counter information cannot be used.

5.1.6.2 Call Forwarding/Call Looping Issues

The NGIIF has looked at customer call forwarding/call looping issues and the related network parameters [such as the Hop Counters, max forwards in an IP network, redirection information (RI), history information] which are available in either the Time Division Multiplexing (TDM) or IP networks. The current state of the industry's transition to All-IP will not support a full resolution of problems induced by customer call forwarding using these capabilities; therefore, the NGIIF offers the following recommendations:

- Network SP should review the use of simultaneous call forwarding restrictions that limit the number of times calls can be forwarded through a single line.

- SPs should review their services to determine whether they easily allow configurations that can generate call looping. For example, it might be desirable to design restrictions so that customers cannot simultaneously forward their mobile calls to their land line at that same time their land line is forwarding calls to their mobile line.

While call forwarding can sometimes result in looping, it seems more likely that where looping is involved in call completions problems, it may be the result of either translations errors or the involvement of multiple call completion intermediate SP who are unaware of which other intermediate SP's networks the call may have already traversed.

5.1.7 SS7 Inter Network Trunk Signaling Testing

The Message Transfer Part (MTP), ISUP, and SCCP compatibility tests found in the NGIIF Reference Documentation (ATIS-0300011 to ATIS-0300022, *NGIIF Reference Document Part III- Attachments A through K*),

are recommended to verify the compatibility of networks during interconnection. These tests are intended to be used as a recommended set of minimum tests of the SS7 protocol.

It is assumed that the two interconnecting networks may have some additional tests they may wish to perform during interconnection. These tests should be part of the bilateral agreements developed for SS7 network interconnections.

Network compatibility testing verifies the correct interworking of SS7. The tests are written for the interconnection of the two networks.

The full test suite of all recommended tests should be run between the two interconnecting companies for any interconnection configuration that was not previously tested. Both the manufacturer model and software load of the interconnecting signaling network elements define the interconnection configuration. Subsequent interconnections, using configurations previously tested by the two interconnecting companies, may be tested at their discretion.

The tests in the NGIIF Reference documentation have been divided into Intrusive Tests and Non-Intrusive Tests and defined as follows:

- *Intrusive Tests:* The interconnecting circuit shall be interrupted, with the testing unit inserted into the circuit and acting as an emulator to the signaling point under test.

- *Non-Intrusive Tests:* The test shall be able to observe traffic traversing the link(s) between the two (2) signaling points, in a monitor mode.

5.1.8 Call Set-up Delay

Call set up is part of the TDM SS7 and IP SIP protocols in the PSTN for the connection of the Calling Number to the Called Number across the PSTN.

Long delayed calls, especially without feedback that the call is proceeding, may lead customers to abandon their call attempts and/or report call failure. When a caller hears nothing, it is sometimes referred to by callers as experiencing "dead air".

For SS7, the standards for the timing between SS7 messages are specified in ATIS–1000113-Chapter 4, Table 3, page 4-123. This table is for Timers. In that table, T11 shows the ACM timing as 15 - 20 seconds in response to the IAM. Along with the T7 IAM timing to ACM of 20-30 seconds, this allows for 35 to 50 seconds for call set up, up to the Release Complete Message.

For SIP, timers are defined in IETF RFC 3261, *SIP: Session Initiation Protocol,* and are generally an estimate of roundtrip transmission time. The RFC defaults to 500ms. The resulting timer values can thus be in the same ranges as the SS7 timers described above.

The following text, from 3GPP TS 24.229, Section 7.7, *SIP Timers*, provides additional guidance for mobile applications:

The timers defined in RFC 3261 [26] need modification in some cases to accommodate the delays introduced by the air interface processing and transmission delays. Table 7.7.1 shows recommended values for IM CN subsystem.

Table 7.7.1 lists in the first column, titled "SIP Timer" the timer names as defined in RFC 3261 [26].

The second column, titled "value to be applied between IM CN subsystem elements" lists the values recommended for network elements – e.g., P-CSCF, S-CSCF, MGCF, when communicating with each other i.e., when no air interface leg is included. These values are identical to those recommended by RFC 3261 [26].

The third column, titled "value to be applied at the UE" lists the values recommended for the UE, when in normal operation the UE generates requests or responses containing a P-Access-Network-Info header field which included a value of "3GPP-GERAN","3GPP-UTRAN-FDD", "3GPP-UTRAN-TDD", "3GPP-E-UTRAN-FDD", "3GPP-E-UTRAN-TDD", "3GPP2-1X", "3GPP2-1X-HRPD", "3GPP2-UMB", "IEEE-802.11", "IEEE-802.11a", "IEEE-802.11b", or "IEEE-802.11g", or "IEEE-802.11n". These are modified when compared to RFC 3261 [26] to accommodate the air interface delays. In all other cases, the UE should use the values specified in RFC 3261 [26] as indicated in the second column of Table 7.7.1.

The fourth column, titled "value to be applied at the P-CSCF toward a UE" lists the values recommended for the P-CSCF when an air interface leg is traversed, and which are used on all SIP transactions on a specific security association where the security association was established using a REGISTER request containing a P-Access-Network-Info header field provided by the UE which included a value of "3GPP-GERAN","3GPP-UTRAN-FDD", "3GPP-UTRAN-TDD", "3GPP-E-UTRAN-FDD", "3GPP-E-UTRAN-TDD", "3GPP2-1X", "3GPP2-1X-HRPD", "3GPP2-UMB", "IEEE-802.11", "IEEE-802.11a" or "IEEE-802.11b", or "IEEE-802.11g", or "IEEE-802.11n". These are modified when compared to RFC 3261 [26]. In all other cases, the P-CSCF should use the values specified in RFC 3261 [26] as indicated in the second column of Table 7.7.1.

The final column reflects the timer meaning as defined in RFC 3261 [26].

Table 5.4 - SIP timers[15]

SIP Timer	Value to be applied between IM CN subsystem elements	Value to be applied at the UE	Value to be applied at the P-CSCF toward a UE	Meaning
T1	500ms default (see NOTE)	2s default	2s default	RTT estimate
T2	4s (see NOTE)	16s	16s	The maximum retransmit interval for non-INVITE requests and INVITE responses
T4	5s (see NOTE)	17s	17s	Maximum duration a message will remain in the network
Timer A	initially T1	initially T1	initially T1	INVITE request retransmit interval, for UDP only
Timer B	64*T1	64*T1	64*T1	INVITE transaction timeout timer
Timer C	> 3min	> 3 min	> 3 min	proxy INVITE transaction timeout
Timer D	> 32s for UDP	>128s	>128s	Wait time for response retransmits
	0s for TCP/SCTP	0s for TCP/SCTP	0s for TCP/SCTP	
Timer E	initially T1	initially T1	initially T1	non-INVITE request retransmit interval, UDP only
Timer F	64*T1	64*T1	64*T1	non-INVITE transaction timeout timer
Timer G	initially T1	initially T1	initially T1	INVITE response retransmit interval
Timer H	64*T1	64*T1	64*T1	Wait time for ACK receipt.
Timer I	T4 for UDP	T4 for UDP	T4 for UDP	Wait time for ACK retransmits
	0s for TCP/SCTP	0s for TCP/SCTP	0s for TCP/SCTP	
Timer J	64*T1 for UDP	64*T1 for UDP	64*T1 for UDP	Wait time for non-INVITE request retransmits
	0s for TCP/SCTP	0s for TCP/SCTP	0s for TCP/SCTP	
Timer K	T4 for UDP	T4 for UDP	T4 for UDP	Wait time for response retransmits
	0s for TCP/SCTP	0s for TCP/SCTP	0s for TCP/SCTP	
NOTE: As a network option, SIP T1 Timer's value can be extended, along with the necessary modifications of T2 and T4 Timers' values, to take into account the specificities of the supported services when the MRFC and the controlling AS are under the control of the same network operator and the controlling AS knows, based on local configuration, that the MRFC implements a longer value of SIP T1 Timer.				

Timers for SS7/SIP interworking are defined in Clause 8 of ATIS-1000679, *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control or ISDN User Part.*

Under normal circumstances with SS7 or SIP signaling, post dial delay is likely to be almost an order of magnitude shorter than the timer values discussed above and the timers serve mostly to kill the odd call that has gone awry. Where Intermediate SPs fail to release in a timely manner a call they cannot complete, attempt many routes, or queue calls for completion, long timer values could result in excessive post dial delay for eventually successful calls. NGIIF does not recommend that existing signaling timers be changed. Instead, SPs should expeditiously release calls that they cannot complete. SPs should not queue calls for an extended period or cycle calls through further intermediate SPs.

Even without failure to release in a timely fashion, complicated routing arrangements can result in undesirable post dial delays. Consider a group of intermediate SPs, each of which has some potential routes to a given termination. Those routes may include making use of another SP in the set whose routes may in turn include some of the same routes used by the initial intermediate SP as well as additional members of the set of intermediate SPs. This situation is analogous to that in which a group of Local Area Network (LAN) bridges is interconnected and the "spanning tree"[16] algorithm must be employed to enable only certain paths to avoid looping. In the absence of information about other SPs' routes, looping may result, as well as fruitless reattempts of routes already tried by other SPs. The potentially large number of routing attempts alone may introduce unacceptable post dial delay.

5.1.8.1 Delayed Ringing or Ringing Without Call Set-up

Callers expect to hear, during call processing, that their call is progressing, and that when it has been set up, end to end, they will hear tone (ring back), indicating that the call set up has progressed to the point that it is ringing at the called end. While delayed ringing due to call post dial delays is a problem as discussed above, ring back should not be presented until the terminating switch has received and processed an IAM and is responding to the IAM with an ACM.

Delayed ringing may be due to call set up delay (refer to Clause 5.1.5).

In SIP signaling, ring back should depend on receipt of a 18X Ringing message from the far end.

When ring back is presented to the caller, in the absence of receipt of the proper SS7 or SIP message, the caller may infer that the phone they are calling is ringing when in fact it is not. Refer to Clause 5.1.2.1, *NGIIF SS7 Cause Code & Tones & Announcements*, and IETF RFC 3261 for more information.

5.1.8.2 Dead Air

A Called Party may answer a call and experience "dead air". This situation can sometimes result from the use of predictive auto dialers, but in the context of normally initiated calls is likely to have other causes in the setup of the media path.

5.2 Transmission Quality

Transmission quality issues typically reported are identified as static, noise on line, choppy voice, echo, loss, etc. Transmission issues can occur in many places within the network such as customer CPE, customer inside wiring, local loop/cable pair, central office facilities, bad trunk groups.

When trying to resolve transmission degradations, trouble isolation practices should be employed to identify the origination point of trouble. Trunk maintenance procedures in many cases will identify "bad" trunks, taking the trunk out of service for testing and repair. Transmission issues may occur on an intermittent basis, making them more difficult to identify. Trouble reports and the trouble resolution procedures are another way to identify and resolve transmission quality issues within the network.

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[15] Table 7.7.1 From 3GPP TS 24.229, Section 7.7, *SIP Timers*

[16] Perlman, Radia, *Interconnections: Bridges, Routers, Switches, and Internetworking Protocols*.

Important parameters for transmission quality are included in the best practices for managing intermediate SPs.

5.2.1 Fax

TDM circuits using the G.711 codec support facsimile (i.e., fax) as well as voice transmission. Voice over Internet Protocol (VoIP) can support fax if properly engineered, but VoIP connections engineered to support voice will not necessarily support fax. Fax machines generally convert scanned data into voiceband analog signals for transmission. Non-waveform network codecs (e.g., G.729, G.723.1, etc.) used to convert these analog signals for digital transmission – either packet or circuit – will not support fax. If a packet or circuit-switched connection carries audio via a non-waveform codec, all faxes will fail. VoIP gateways provisioned for voice codecs like G.729 must renegotiate to G.711 fax pass-through or T.38 fax relay if they detect fax answer. Fax calls may also be more sensitive to IP packet loss than voice calls.

G.726 is a waveform codec that operates at lower bitrates than G.711. Some VoIP SPs provision a line for the G.726 codec if they know that the line will be used for some combination of fax, modem, and voice. This typically works, although transmission is slower than normal.

5.2.2 Voiceband Data

Voiceband data/modems would follow similar guidelines as for fax. G.711 will be needed for higher speed modems (>14.4 kbps). G.726 should work for slower speed modems.

5.3 Routing

Multiple entities will be involved in routing for all calls except those that originate and terminate within the network of a single SP. At a minimum, there will be an originating and a terminating SP and, generally, there will be an IXC or SPs involved on LD calls. Coordination in routing is thus a prerequisite to successful call completion.

Each entity has particular responsibilities. Originating SPs should ensure facilities are properly provisioned and interconnection is in place prior to routing calls. They also must maintain accurate routing table translations.

The terminating SP that serves the called number must populate the necessary information into industry databases [e.g., LERG Routing Guide and Numbering Portability Administrative Center (NPAC)] so that originating and intermediate entities can determine the serving switch and its homing arrangements.

The entity providing LD service to the caller must make use of current industry data and knowledge of its own network connectivity to build its routing tables. Where the LD SP makes use of subcontracting entities, it needs to agree with those entities on their coverage of the area that calls are being routed.

This section discusses industry data sources, procedures for their population and use, and other issues related to routing. When a routing change is made for any reason, SPs should test the new route to mitigate call completion issues. Additional information can be found in Clause 7, *Trouble Reporting & Contact Directories*.

5.3.1 Numbering Plan Area/ Central Office Code Routing (NPA/NXX)

NPA/NXX routing uses the analytics of digits dialed, jurisdictional decisions, and subsequent route selection to direct voice traffic to properly installed points of interconnection. This is primarily accomplished through the automated table-based selection of a route or routes and applies to the selection of routes by switching systems, as well as the planning of routes for a properly functioning network. Similarly, table-based routing applies to other aspects of call completion such as call setup via the signaling network. Routing is one element of the overall interconnections that work together concurrently in call completion.

For parties to properly establish and maintain NPA/NXX routing, and to troubleshoot NPA/NXX routing issues, expertise in the following aspects of telephony data, equipment, processes, etc., is essential:

- In-depth understanding of numbering resources which includes ordering, billing, and notification processes (Telcordia Technologies, Inc. dba iconectiv®) Business Integrated Routing and Rating Database System (BIRRDS) products – e.g., LERG™ Routing Guide, TPM™ Data Source).

- o The LERG Routing Guide is referenced in various ATIS guidelines (e.g., ATIS-0300119, Thousands-Block (NPA-NXX-X) & Central Office Code (NPA-NXX) Administration Guidelines (TBCOCAG)) and is considered an integral part of the routing data exchange among SPs.

  - o The LERG Routing Guide contains local routing information obtained from BIRRDS, reflects the current network configuration and scheduled changes within the PSTN, and provides limited routing information pertinent to other technologies that include wireless and VoIP.

  - o Timely receipt of LERG Routing Guide notifications (daily/monthly/quarterly) is important to the maintenance and integrity of routing tables.

  - o NPA/NXX and thousands-block data entry into BIRRDS products is important to mitigate call completion issues and assist in troubleshooting.

- Knowledge and understanding of the operation of the PSTN:

  - o Switches

  - o Interconnection

- Knowledge and understanding of VoIP and SIP interaction with TDM and SS7 interaction in the PSTN.

- Knowledge and understanding of Extended Area Service (EAS) or franchise and Local Access and Transport Area (LATA)/Major Trading Area (MTA) boundaries, Exchanges/Rate Centers.

- Knowledge and understanding of handling of interLATA EAS – Local Number Portability (LNP) queries when interLATA Location Routing Number (LRN) is returned.

- Knowledge and understanding of the various tandem functions within a given network architecture (i.e., access, local, intraLATA, interLATA, intermediate, operator services, 911).

- Knowledge and understanding of network interconnection homing hierarchies.

5.3.2 Interconnection Agreements

For calls to originate and terminate within the PSTN, numerous companies must interface physically, thus "interconnecting" with each other. Contractual agreements must be established.

Individual local exchange tandem SP companies may have different rules as to how/what traffic traverses their network based on the interconnection agreements between the local exchange tandem SP and the interconnected SP.

> NOTE: An IXC SP follows FCC tariff rules.

In addition to agreements developed between companies that physically interconnect with each other, further agreements for billing and call termination purposes may be needed among other SPs in the call path to complete a local or toll call.

5.3.3 Homing Arrangements

Homing arrangement(s) is the relationship between switching system(s) in a routing scenario. An SP's subtending switch serving a portion of an incumbent LEC's franchise territory may home on the incumbent LEC's tandem.[17] A competitive tandem in the same territory may have a different serving arrangement from the incumbent LEC.

There are three tandem homing jurisdictions, which are Inter-state, Intra-state/Intra-LATA, and Intra-state/Inter-LATA. Due to regulatory constraints, some SPs may be prohibited from establishing all three tandem homing jurisdictions. There are also various types of tandem functions (e.g., local, toll).

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[17] In some cases, an SP may choose to home on a competitive tandem rather than an incumbent LEC tandem.

Once a valid effective date of the homing arrangement is determined, the NPA/NXX, valid switch, and supporting homing arrangement information must be entered in a timely manner into the iconectiv BIRRDS database for notification to other SPs via the LERG Routing Guide.

Homing arrangements entered into BIRRDS must be valid and denote connectivity between the two switching entities for the function(s) indicated. Hence, when a switching entity indicates that it subtends or homes on a given tandem, it becomes a confirmation that there is interconnection between the two entities. On a terminating basis, the homing tandem is considered the last choice for completing traffic destined for the switching entity.

Incorrect homing arrangements in BIRRDS may result in blocking calls destined for a switching entity. For example, if the BIRRDS data entry for a switching entity indicates that the switch homes on a particular local tandem when in fact it does not, only the local tandem SP will knowhow to correctly route calls and only those calls that originate from its own subscribers. Other companies, however, will route the calls to the local tandem in accordance with LERG Routing Guide entries. If there is no connectivity between the local tandem and the terminating switching entity, the local tandem may block the calls. Likewise, there may not be interconnection between the local tandem and a toll tandem owned by the same SP. Once the calls reach the local tandem there is nowhere for the local tandem to terminate the traffic, and it will be blocked.

5.3.4 Routing Implementation

- *General*:
  - o There are industry recommended minimum BIRRDS data entry time intervals for network activity that should be followed to minimize problems associated with call completion/call termination. The intervals are noted in ATIS-0300046, *Recommended Notification Procedures to Industry for Changes in Access Network Architecture* (there should be an understanding that interconnection arrangements and facilities need to be in place prior to activation of an NPA/NXX and other network changes involving interconnecting companies):
    - ▪ There are different minimum time intervals for various network changes that include activity associated with new or discontinued NPA/NXXs, tandem homing arrangements, office capability changes associated with a new or changed rate center, destination code changes, etc.
    - ▪ Not following the minimum time intervals may result in call completion failure.
    - ▪ Expedites for establishing a new NPA/NXX or modification/disconnect of an NPA/NXX may run the risk of inadequate time for other SPs impacted by the activity to reflect the activity in their networks.
- *External* (Industry Notification):
  - o Switching information (End Office, Tandem homing arrangements) published in the LERG Routing Guide:
    - ▪ The LERG Routing Guide is the appropriate source for parties terminating calls to rural companies to use to set up their routing. SPs intending to route non-local traffic to a rural SP should direct their traffic as shown in the LERG Routing Guide. Parties should therefore keep their routing information updated in the LERG Routing Guide.
  - o Parties are encouraged to ensure that their NPA/NXX and NPA/NXX-X assignments are entered into and maintained in BIRRDS for publication in the LERG Routing Guide.
  - o Some companies may not subscribe to the LERG Routing Guide and may use alternate sources for obtaining data; however, these companies must have current data populated in their switches.
  - o If the SP is not its own Administrative Operating Company Number (AOCN), it will need to contract with a third-party vendor to update NPA/NXX, NPA/NXX-X, and switch data into BIRRDS. Data entry should occur within seven calendar days of assignment by per the ATIS Industry Numbering Committee (INC) TBCOCAG.
  - o Parties are encouraged to ensure that their NPA/NXX or NPA/NXX-X serving switch record be populated to reflect the local exchange tandem homing arrangement(s) in BIRRDS/the LERG Routing Guide. However, there is no industry requirement that this information be populated.

- *Internal* (Building Routing Instructions):

  o Obtain interconnection layouts for each office for which routing is being addressed.

  o Apply applicable routing as outlined above in this Clause, which includes routing to end offices, wireless offices, Point of Interface (POI)s, remotes, and tandems.

  o Pass routing information to routing translations personnel for implementation in the appropriate switching infrastructure.

  o Some originating companies, supported by appropriate interconnection agreements, may elect to implement alternate routes other than those published in the LERG Routing Guide.

5.3.5 Considerations in the Code Routing Process

- Secure a reliable source for embedded base and code activity (e.g., NPAs, NPA/NXXs, NXXs) related to adds, deletes, or modifies (e.g., the LERG Routing Guide).

- Identify codes (e.g., NPAs, NPA/NXXs, NXXs) that require routing/translations activity as a result of adds, deletes, or modifies.

- With appropriate trunking in place, identify/create the primary routing path, including overflow routes (in today's environment, these actions are typically automated processes).

- Considerations an SP should take into account when selecting a primary route:

  o Routing must be based on dialed digits or LRN

  o Routing arrangements between originating office(s) and terminating office(s)

  o IntraLATA versus interLATA, if aplicable

  o Local calls should be routed as local calls

  o Toll calls should be routed as toll calls

  o Optional calling plans

  o Signaling required on the terminating end (e.g., seven digits versus 10 or 11 digits)

  o Determine dialing patterns on number of digits that can be dialed on a given call

  o Determine when dialed digits require digits to be deleted and/or prefixed

- Additional Routing Considerations:

  o It is important to understand the LERG Routing Guide is the guide for *local* exchange routing and its core function is to indicate the terminating switch associated with an NPA/NXX and the tandem homing arrangements for that switch. The LERG Routing Guide does not provide routing information for IXCs. IXCs must establish routes between originating offices and terminating local networks as defined in the LERG Routing Guide.

  o Not all originating SPs have direct connectivity to the terminating end offices to which given NPA/NXX codes are assigned or to the access tandems to which the terminating switch is homed in the LERG Routing Guide. Additionally, originating SPs may opt to route through other SPs due to various network conditions (for example, network congestion) to reach the terminating end office.

  o Originating SPs, utilizing intermediaries, should internally maintain in their respective routing tables current NPA/NXX reachability information provided to them from each of their interconnected intermediaries denoting the details of each intermediary's coverage area and other aspects relevant to route selection. Reference Clause 6, *Intermediate SPs*, of this Handbook for additional information.

5.3.5.1 Example #1 – Routing Ported Traffic – Intra LATA

Example #1 depicts how LNP is handled in an intraLATA network. This example is for illustrative purposes only and is not intended to represent all possible call paths, SPs, network components, technologies, etc. There can be variations as to which switch launches the LNP query dependent upon where a call originates.

A Code Holder is the SP assigned an NPA/NXX code (NXX-A) record in the LERG Routing Guide. The NXX-A Code Holder is responsible for default routing functions (e.g., vacant code treatment) associated with its own numbering resources and any unassigned block(s) in the pooled NPA/NXX code. More information on Code Holder responsibilities can be found in ATIS-0300119, *Thousands-Block (NPA-NXX-X) & Central Office Code (NPA-NXX) Administration Guidelines*.

An LRN is a unique 10-digit number assigned by a Code Holder. The Code Holder assigns LRN(s) to select switches in its own network that require an LRN(s). The Code Holder assigns an LRN(s) to each of the selected switches using an NXX-A record and associated thousands-blocks served by each of its individual terminating switches. Each switch is assigned an LRN(s) that uniquely identifies the homing arrangement(s) of the terminating switch of the ported number.

Using LNP processes for this intraLATA example, the dialed number (NPA-NXX$_1$-1234) is determined to be "ported" via a database dip that occurs in the call setup. The dialed number, NPA-NXX$_1$-1234 is "mapped" in the LNP database to the new SP's LRN of NPA-NXX$_2$-9999. The NPA-NXX$_2$-9999 LRN is processed through the call setup as if it was the called number and routed accordingly to/toward the new SP. The actual dialed number (NPA-NXX$_1$-1234) is stored in the Ported Number GAP of the IAM message being sent. At a point prior to completing the call, the stored dialed number (NPA-NXX$_1$-1234) in turn replaces the LRN and the call completes to the dialed number via the new SP's network.
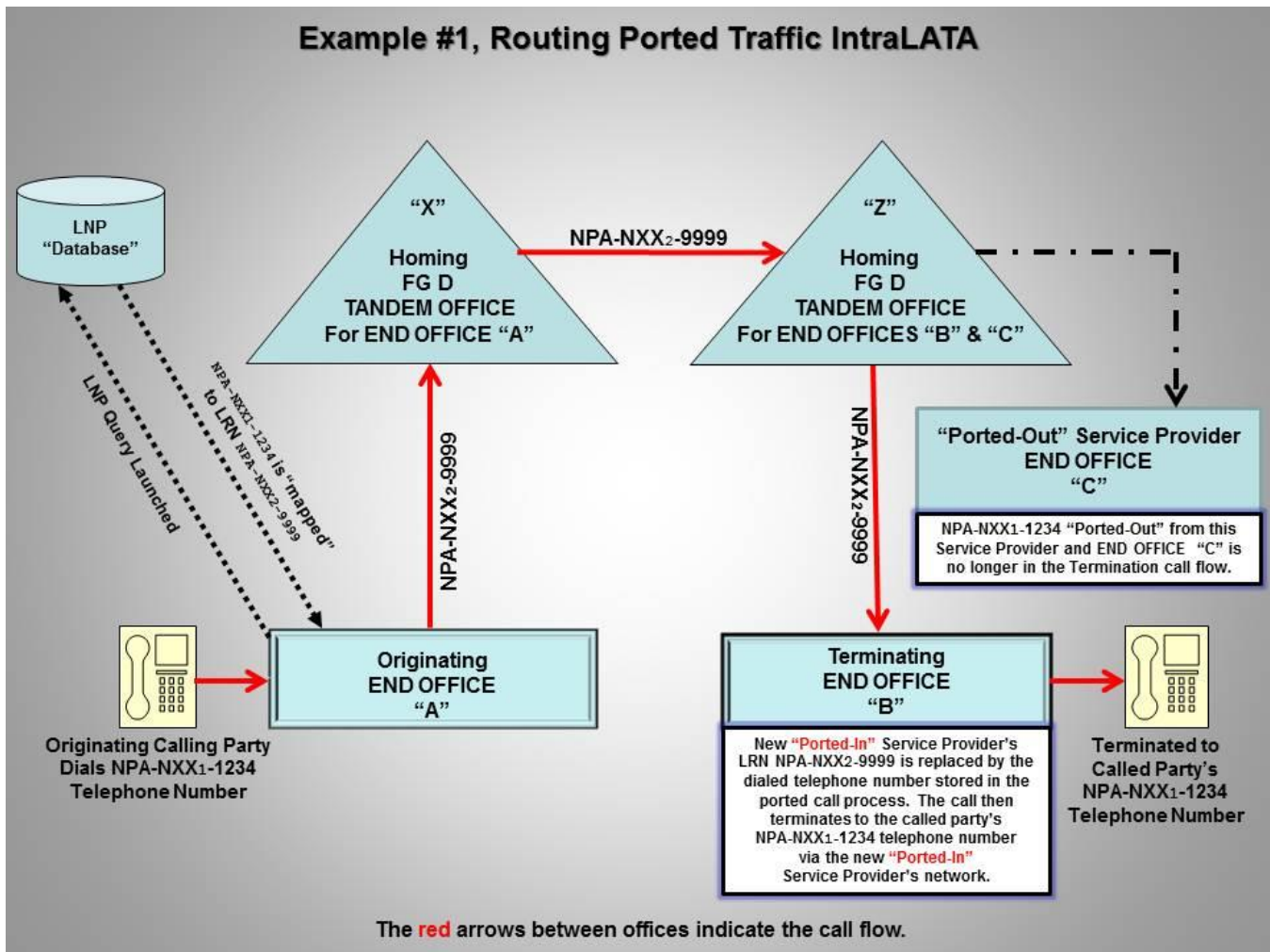
Figure 5.1 - Example #1: Routing Ported Traffic IntraLATA

5.3.6 Potential Call Failure Points

A call failure may occur anywhere in the call path, beginning with the originating Calling Party dialing a TN, all the way to the call completion point. A call may traverse multiple SPs, multiple switching entities, multiple LATAs, multiple technologies, etc. It has become commonplace for a call that originates on a TDM network, for instance, to terminate on an IP network and vice versa. Further, there are SPs who enter into contractual arrangements with other SPs, sometimes referred to as intermediaries, to "carry" their traffic. The intermediary SPs may in turn, have contractual arrangements with other intermediary SPs. Multiple technologies, SPs, etc., create a cascade effect in the network and each can generate points of vulnerability.

SPs who are not constrained by LATA boundaries may not require the services of an IXC in the call flow process. Wireless SPs, when feasible, may elect to carry their originating traffic over their own backbone network to the point of completion for both intraLATA and interLATA calls.

5.3.6.1 Examples #2-#5 –InterLATA Traffic

It should be noted that an LNP query is performed for ported and pooled areas, to determine whether or not a TN is ported or is affected by pooling.

Examples #2 and #3 show generic interLATA call flows for wireline and wireless originating calls, respectively, to a non-ported number.

Example #4 shows a more complicated routing scenario in which the IXC does not perform an LNP query on a call to a ported number.

Example #5 shows a case where the Calling Party's IXC makes use of multiple intermediate SPs.

These examples are for illustrative purposes only and are not intended to represent all possible call paths, SPs, network components, technologies, etc. Each of these examples depicts a "possible" call path.
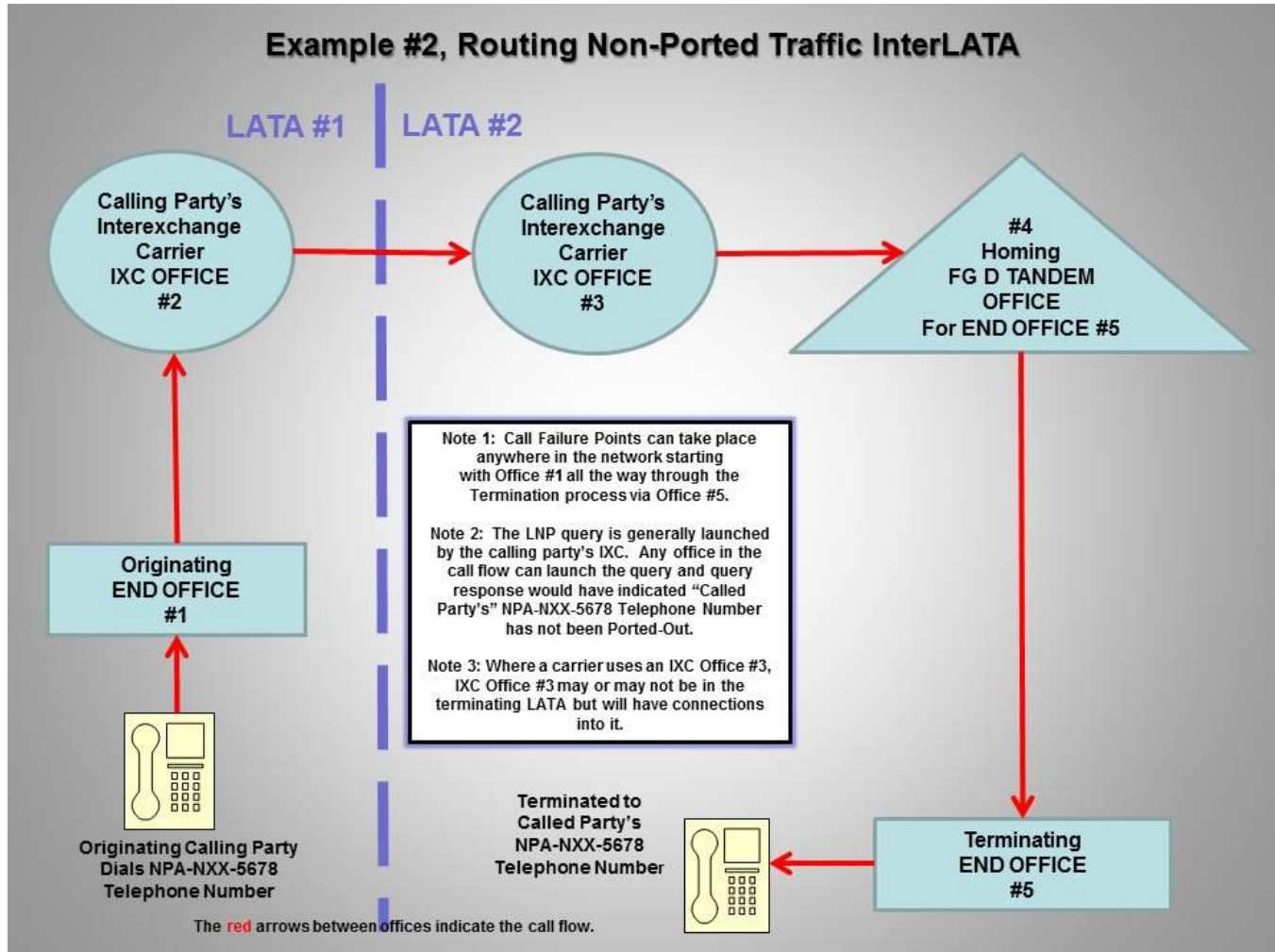
Figure 5.2 - Example #2: Routing Non-Ported InterLATA Traffic
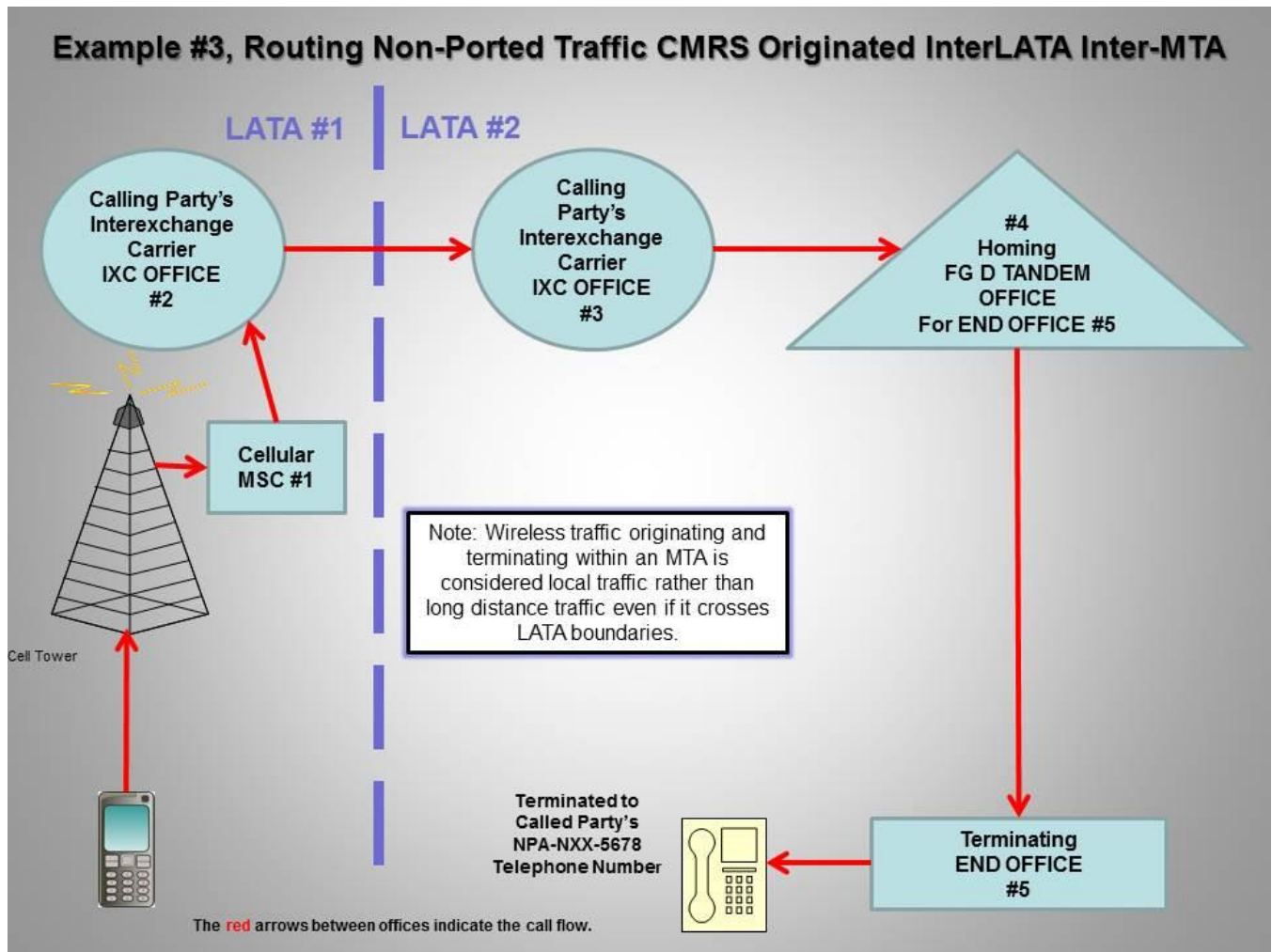
Figure 5.3 - Example #3: Routing Non-Ported Traffic CMRS Originated InterLATA Inter-MTA
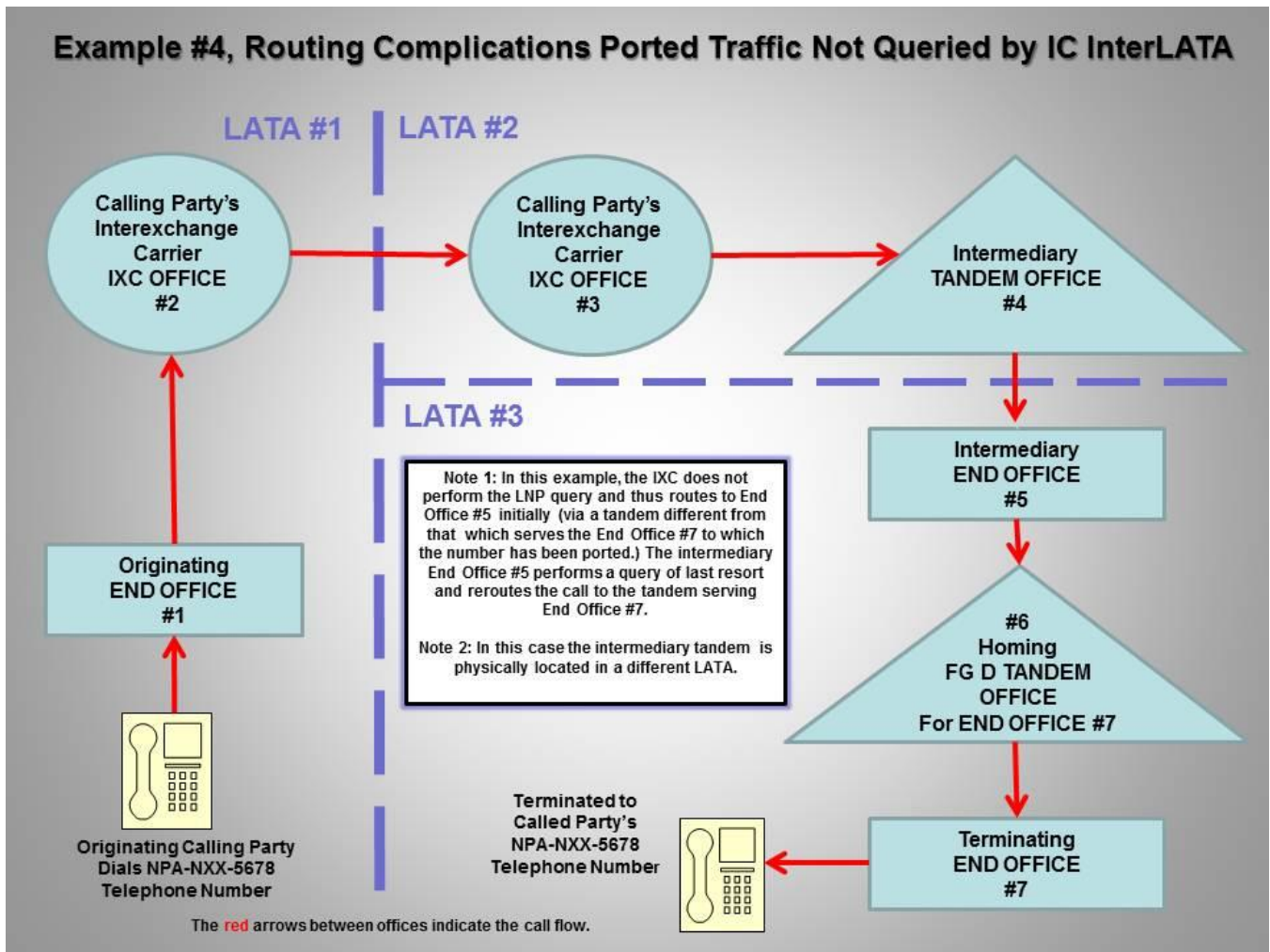
Figure 5.4 - Example #4: Routing Complications, Ported Traffic Not Queried by IXC
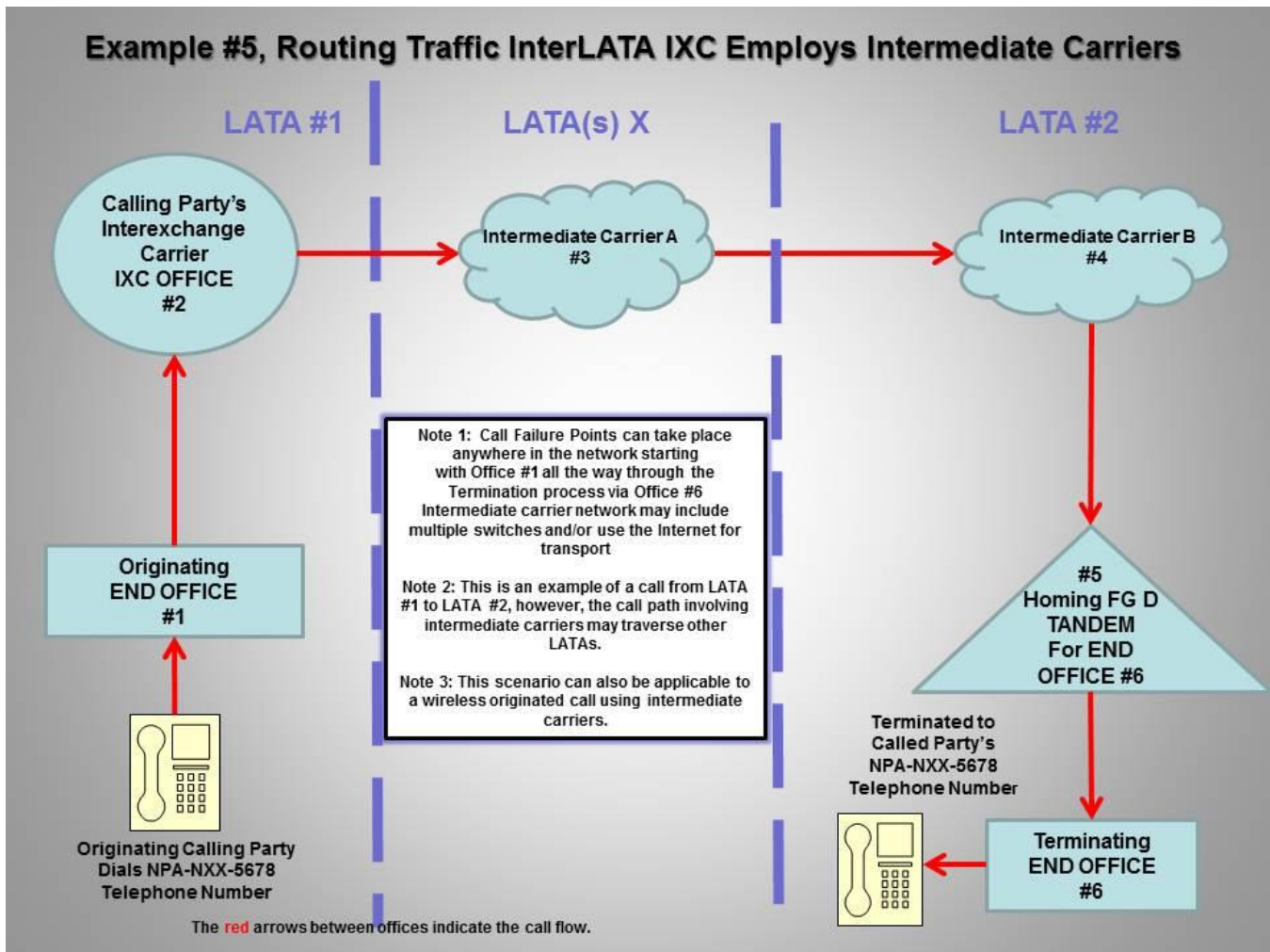
Figure 5.5 - Example #5: Routing Traffic InterLATA, IXC Employs Intermediate Carriers

5.3.6.2 Translations

Software translations direct the routing of calls across the network (i.e., business/residence lines, trunks, switches, etc.). Software translations are input via automated or manual processes. Errors can be introduced into the network when translations are input incorrectly, resulting in calls being misrouted, call failures, etc.

5.3.6.3 Toll Free

Completion of toll-free calls may be impacted by the call completion issues discussed in this handbook when a toll-free number translates to a Plain Old Telephone Service (POTS) number.

In addition, toll free numbers involve the SMS/800 database and associated Service Control Points (SCPs). Toll free numbers are managed through the RespOrgs for updates to the SMS/800 database which in turn updates the associated SCPs. The routing of toll-free calls is based on such factors as time of day, geographic attributes, etc. Therefore, reported problems in reaching a dialed toll free number may not be a POTS routing issue per se, but rather due to characteristics in how that number is translated in the SMS/800 database and SCPs.

5.3.7 Looping

The diagram below is a basic example of call looping. Call looping may occur at any point in the call flow process after the call leaves the originating SP's network.

Long Distance – Call Termination

- Looping Diagram:
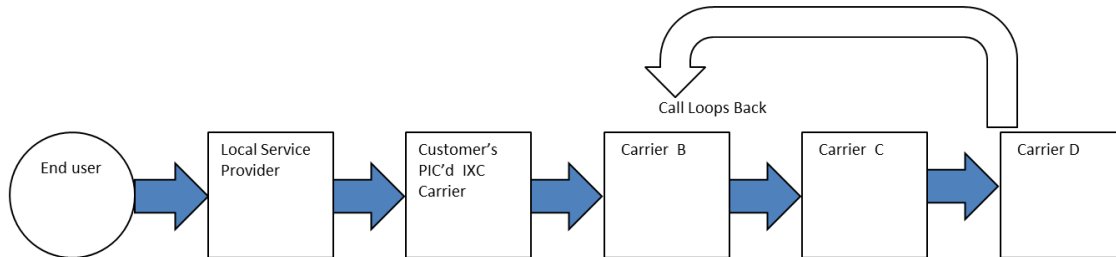
Figure 5.6 - Basic Example of Long Distance (LD) Call Looping

5.3.8 LNP Implications

Call completion/call termination issues may occur in some ported TN scenarios. In particular, when a native TN ports away from the native SP then the TN is ported back to the native SP. In this case, the port should be completed via the "Port To Original" option in the NPAC. The "Port to Original" option allows for a coordinated snap back of the TN to the native SP so routing may be completed via the LERG Routing Guide, in lieu of an LRN. Where a native TN is ported back to the native switch via an LRN, instead of the "Port To Original" option, it is possible for the N-1 SP to not recognize the LRN as valid and thus to create a looping or call failure scenario. This prevents the call from properly terminating at the native SP's switch. Because failure to properly complete a call ported back to the Code Holder via an LRN represents a violation of LNP processing standards (ATIS-1000002, *Number Portability Switching Systems*), SPs are encouraged to use the "Port To Original" option. More information can be found in Clause 8.6, *LNP* of this document.

5.4 Network Congestion

Network congestion results when traffic demand (customer call attempts) exceeds the capacity of the network, unless the network has already been degraded by some other physical situation. One standard in use in the industry is the managing of the P.01 Grade of Service may help reduce congestion-related call completion issues.

When very high calling volume occurs, a direct engineered route may overflow to a designed alternate routing path(s). However, if the calling pattern is severe and sustained (and left unchecked by either automatic or manual network management action), the designed robustness of the network then contributes to the spread of congestion throughout the network. This condition often results in lost calls.

A common cause of network congestion is mass calling generated by calling situations such as telemarketing, political campaigns, or Emergency Notification System (ENS) messages. Mass calling can initiate a high volume of traffic on a network over a relatively short duration. During this period of time, normal network traffic patterns are disrupted and may result in network congestion. These situations may impact calls destined to all customers including those located in rural areas.

The following information applies to wireless, wireline and Next Generation Networks (NGN). Network congestion may be due to one or more of the following overload factors detailed in sections below.

5.4.1 Network Element Degradation

The performance of a network element may be negatively impacted due to a component failure caused by a hardware or software trouble. The loss of an element component may limit the call handling capacity of the network and contribute to network congestion.

- Switching systems

- Facility systems

- Signaling systems

- Wireless systems

- NGN systems (e.g., VoIP)

- Routing assignments

5.4.2 Mass Calling

Customer complaints or trouble reports may be received when mass calling events, holiday call overloads, or peak day traffic patterns occur, preventing a call completion or termination. The customer generating the trouble report may not be aware of congestion events occurring in other areas of the country or in networks between point A and point B.

Networks are normally designed to accommodate average business day customer calling patterns. Peak day or holiday customer calling may result in network congestion; however, due to the regionalization of the traffic patterns, this type of mass calling is generally well-handled by network management techniques.

Mass calling scenarios generated by auto dialer devices may result in a focused overload on the network and may contribute to network congestion.

5.4.2.1 Auto Dialers

5.4.2.1.1   "Dead Air" or "Abandoned Call" Situations

Consumers experiencing "dead air" or an "abandoned call" may not be aware that they may be receiving a call generated by a type of auto dialer. When an auto dialer connects an answered call to a live agent, it is often called a predictive or power dialer, and uses real-time analysis to determine the optimal time to dial more numbers.

If someone answers but no agent is available within two seconds of the person's greeting, under FCC regulations, the call is abandoned, and the dialer is required to play a recorded message. The FCC requires predictive dialers to abandon less than 3% of answered calls.

A "silent call" is a call generated by a predictive dialer that does not have an agent immediately available to handle the call. In this instance, the call may be terminated by the Calling Party, and the Called Party receives silence, i.e., dead air, or a tone from the SP indicating the call has been dropped.

In the United States, the Federal Trade Commission (FTC) uses the term "abandoned call" instead of "silent call" in its regulations applying to telemarketing. Abandoned calls in non-FTC contexts may refer to a caller who decides not to wait for an answer before hanging up.

Although there are FTC and FCC regulations and/or requirements; there is no means to ascertain if these requirements are followed by generators of auto dialed type calls, or even if consumers are aware that they are receiving auto dialer type calls.

Additional information on auto dialers is available in ATIS-0300105, *Next Generation Interconnection Interoperability Forum (NGIIF) Auto Dialers Reference Document.*

5.4.2.1.2   Failure to Receive Calls from Emergency Notification, Public Service, Political, or Other Type of Automated Announcement Systems

Consumers may generate complaints or trouble reports when they are aware that an automated announcement was sent out in their area which they did not receive.

Emergency notification, public service, political, or other types of automated announcement system users may send out announcement calls at a rate greater than available network capacity in a given time period and subsequently not all calls will complete. There are additional factors that go into the determination of ENS call completion including, but not limited to: time of day, holidays, trunk capacity, host or remote switching configurations, traffic patterns, length of announcement, if the ENS call was originated locally or from another area traversing between multiple networks, completeness or accuracy of the calling system TN database, reaction of calling systems to answering machines, voice mail, no answer or busy line conditions, and consumers who may utilize call blocking or call selection type features that prevent receiving calls from unknown calling systems.

Terminating SPs may wish to identify entities that engage in this type of calling activity on a regular basis in their service territory, such as local school districts. The SP may be able to work with the calling entity so that the calling activity is structured in a manner that reduces the likelihood for network congestion, for example, by spacing calls out to a greater degree; calling at different times, or rotating through called NPA-NXXs in a way that better distributes the calling load across the terminating SP's network.

5.4.3 Fraud

Call completion may be impacted by fraudulent activity in the network. Individuals or entities may purchase wireless service and use the associated subscriber identity module (SIM) card in a device (SIM box) through which they offer to terminate LD traffic by re-originating it as a wireless call. The concentration of wireless calls originating within a cell site area may congest the wireless network, resulting in poor call completion rates for the traffic offered to the vendors using SIM boxes. As this usage violates wireless terms of service, SPs will terminate the service, but not necessarily before congestion-induced blocking occurs. Even after termination, calls may continue to route to the entity that had been set up in the SIM box and engender call completion problems.

Wireline fraud schemes also exist and may impact call completion when call volume takes out routes that SPs may have counted on.

Terminating SPs and wireless SPs may want to explore ways of detecting such potential fraud schemes, for example, by looking at calling patterns from their retail customers that might be indicative of fraud.

5.4.4 Force Majeure & Disasters

The performance of a network may be negatively impacted by a disaster. The following events not only cause physical damage to a network, but compound the situation by generating excessive customer calling attempts:

- Weather
- Earthquake
- Volcanic eruption
- Solar activity
- Fire, flooding, etc.
- Terrorism

5.4.5 Inadvertent Human-Related Issues

Inadvertent human-related issues have the opportunity to cause physical damage to a network and may exacerbate the situation by generating excessive customer calling attempts leading to network congestion. Examples of these issues include accidents and human error (e.g., planning or forecasting miscalculations).

5.4.6 Traffic Pumping/Access Stimulation

Traffic pumping, also known as access stimulation, occurs when a LEC with high switched access rates enters into an arrangement with a SP of high call volume operations such as chat lines, adult entertainment, and free conference services. The arrangement inflates or stimulates the access minutes terminated to the LEC, and the LEC then shares a portion of the increased access revenues resulting from the increased demand with the

underlying SP of services, or offers some other benefit to the SP (¶656 of the *Universal Service Fund (USF)/Intercarrier Compensation (ICC) Reform Report and Order*).Increases in traffic related to access stimulation could cause network congestion if trunk groups are not properly sized and may result in call completion/call termination issues.

5.5 Call Analytics

Call analytics generally uses algorithms to develop a reputation score for a call. These algorithms involve a variety of sources and may identify historical call traffic patterns for many phone numbers and user specific details, such as the calling and called parties' numbers, and SIP header signals. Analytics providers may receive consumer spam reports and complaints directly or reports sent to the FCC and FTC. The analytics may rely upon industry registry information, number validity/allocation status, and business registry databases. Alternatively, analytics may be as simple as an evidentiary determination to block a number that has been determined to be associated with the source of originating fraudulent calls. Of note, National Security/Emergency Preparedness (NS/EP) priority calls should not be subject to call analytics.

5.5.1 Use of Call Analytics

Call analytics are generally used to mitigate illegal and unwanted calls destined to terminating subscribers. The call analytics are usually applied at the terminating end of the call. Call analytics may be used to determine if a call should be flagged as "unwanted", and if so, then if it should be labeled as fraudulent or nuisance. Analytics also may provide more granular call classification information, such as sub-category of fraudulent or nuisance which could then be labeled e.g., "Spam Likely" or "Robocall Likely". The establishment of these categories may be subject to interpretation and can result in a false negative or false positive result. The results of the analytics can cause a call to be accepted and answered or the call could go unanswered based on the displayed information.

The information provided by analytics providers can also lead to a decision whether or not to block or screen a particular phone call, although the analytics provider may not make that decision directly, dependent on their terms of service with their customers. SPs or user preferences would generally control which calls will be blocked versus allowed to continue and screened.

Reputation scoring generally involves metadata associated with the initiation of calls that traverse networks in real time. The only way to conclusively determine the intent of a call is by evaluating the content of the call, which can only be accomplished upon the establishment of the call to the intended end user. Analytics are used to anticipate the likelihood of illegal or unwanted intent before the call has been established. Analytic providers strive to minimize false positives and false negatives by maintaining a high percentage of accurate reputation scoring.

NGIIF recommends that SPs who employ call analytics have an efficient process to identify and quickly resolve false positive and false negative analytic results. While false positives will likely be identified by the customer of the originating SP, false negatives will generally be identified by the customer of the terminating SP or analytics engine provider. Thus, customers of the originating SP will likely inquire with the originating SP but may also inquire with terminating SP to resolve false positives. False negatives are generally only identifiable by the customer of the terminating provider who will inquire for resolution with the terminating provider or call analytics engine provider. Both intake approaches should be considered.

5.5.2 Application of Call Analytics

Call analytics are typically applied on a call-by-call basis and designed to either be controlled by the SP (network-wide) or by the terminating subscriber at the service level if the use of analytics is a subscription service versus an add on to basic service. Additionally, there may be analytics applied by the SP specific to the originating subscriber service level or class of subscriber service.

5.5.2.1 Subscriber Service Level Analytics

Subscriber Service Level Analytics are typically configured to provide the subscriber with control of the relative application of the reputation ranking of incoming calls that may result in labeling or blocking. These services can be offered by the SP, by analytics engine providers or by a SP via its contractual arrangement with an analytics engine

provider. These analytic services may be provided through a handset based/CPE application, through the SP's network service or through redirection of calls or signaling through the analytic engine provider's platform.

5.5.2.2 Service Provider Network Level Analytics

Service Provider Network Level Analytics are applied network-wide so that all or most subscribers experience the same result from the application of analytics. Examples include Do-Not-Originate blocks, network-wide blocking of invalid, unallocated, or unassigned numbers[18] as well as blocking of calling linked to campaigns that have been specifically identified as originating illegal calls.

5.5.2.3 Blacklist

Blacklists restrict the privilege of a telephone number such that calls originating with the backlisted number are typically blocked. Do-Not-Originate is a negative reputation associated with a telephone number by the SP network resulting in SP blocking and is a type of 'Blacklist'. Blacklists may also be incorporated into Customer Provided Equipment (CPE) or provided as a service feature. There are two benefits of backlisting telephone numbers. Blacklisting numbers will block calls that are associated with known sources of illegal or unwanted calls and may include authorization of the party with the right to use a number to block the unauthorized spoofing of the number. Also, blacklists protect the users of inward-only services such as Toll Free or Direct Inward Dialing services. The blacklisted numbers also have the potential to be ported or reassigned. Thus, these blacklisted telephone numbers should be monitored for discontinued use and could eventually be removed from the blacklist or Do-Not Originate list.

Blacklists may be applied at either networkwide or CPE application level. For example, a blacklist could rely upon a Do-Not-Call list or applied across an SP's network or multiple SPs' networks.

5.5.2.4 Whitelist

Whitelists allow a telephone number the privilege to bypass call analytics or blocking due to the accepted level of trust associated with the originating telephone number. The term 'Whitelist' is indicative of a permanent positive reputation for a telephone number. However, whitelisted telephone numbers may be spoofed, ported, or used for a new purpose. For this reason, the term is also referred to as 'registration'. Registration allows a legitimate call originator to provide details of their identity and the nature of their calls directly to carriers, analytics engine companies, or caller identity aggregators. This registration of a number when vetted can then be taken into consideration by analytics algorithms, and in most cases have a negative reputation removed. Whitelisting should not be permanent. The registry and its registered telephone numbers should be monitored for breeches, abusive dialing practices, or suspected nefarious activity and could be assigned a negative reputation or blacklisted as a result.

Whitelists may be applied at either networkwide or CPE application level. For example, a whitelist could rely upon the information within a mobile device's contact list or applied across an SP's network or multiple SPs' networks.

5.5.2.5 Call Labeling/Flagging

Call labeling or flagging relies on a generalized label applied on a call-by-call basis and placed on calls that have been determined to have call characteristics consistent with the label or applied flag. These labels are then displayed on the CPE device. In some instances, a label or flag may be associated with a specific caller to more specifically and accurately label calls originating from a specific number. For example, a specific debt collector's calls that were originally labeled as "ROBOCALL" could be re-labeled to "DEBT COLLECTOR".

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[18] FCC 17-151 permitted SPs to block calls in well defined circumstances.

5.5.3 False Negative

False negative is a result that indicates that a condition does not exist when the condition in fact does exist. In the context of call analytics, false negative calls lack the affirmative identification of an unwanted call when the call should be labeled/flagged as unwanted. This may happen where call analytics mischaracterize a call with a positive reputation score, when the call should be categorized with a negative reputation score. For example, an illegal call is initiated where analytics are applied, but the call is not blocked when it should have been blocked.

5.5.4 False Positive

False positive is a "false alarm." False positive is a result that indicates a condition exists when it does not. In the context of call analytics, false positive calls are affirmatively identified as unwanted, labeled/flagged or blocked and yet should not be identified as unwanted and labeled/flagged or blocked. False positive analytics mischaracterize a call with a negative reputation score when the call should be categorized with a positive reputation score. For example, a legal call is initiated by a pharmacy and when analytics are applied, the call is blocked when the call should not have been blocked.

6 Intermediate SPs

Some SPs have suggested that reported call completion/call termination problems involve the use of intermediate SPs. Intermediate SPs have been used in the industry for many years, including by rural SPs. This section contains best practices for management of intermediate SPs.

6.1 Intermediate SPs' Registration

The Commission announced that the Office of Management and Budget (OMB) approved, for a period of three years, the collection of information associated with rules requiring intermediate SPs to register with the Commission. All intermediate SPs are required to register with the Commission before offering to transmit covered voice communications adopted in the Commission's Rural Completion, Third Report and Order. The registration requirements for the intermediate SPs are in 47 CFR Part 64.2115.

On April 15, 2019, the Commission announced, in the Federal Register, the OMB approval of the rules for the Rural Call Completion registry-related requirements for intermediate SPs and covered SPs and the effective date of these rules.[19] The following rule provisions took effect on the following dates:

- Any intermediate SP "that offers or holds itself out as offering the capability to transmit covered voice communications from one destination to another and that charges any rate to any other entity (including an affiliated entity) for the transmission"[20] must register with the Commission.[21] Intermediate SPs must register on or before May 15, 2019.

- Any covered SP will have until August 13, 2019, 90 days from the effective date of the registry rules, to ensure that all intermediate SPs involved in the transmission of its covered voice communications are registered with the Commission[22].

All registrations must be submitted through the Commission's website at https://www.fcc.gov/ipr. The instructions for submitting a registration are at https://www.fcc.gov/files/ipr-instructionspdf. Intermediate SPs must submit any change to their registration information to the Commission within 10 business days of the change.[23]

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[19] Rural Call Completion, Final Rule, Announcement of Effective Date, 84 Fed. Reg. 15124 (April 15, 2019).

[20] 47 U.S.C. § 262(a).

[21] *Third RCC Order*, 33, FCC Rcd at 8407, para, 17; 47 CFR § 64.2115. In the March 2019 *Fourth RCC Order,* the Commission clarified that the registry requirements in section 64, do not apply to non-U.S. intermediate providers on calls terminating outside of the United States. *See Rural Call Completion,* Fourth Report and Order, FCC 19-23, para. 27 n.84 (2019) (*Fourth RCC Order).*

[22] *Third RCC Order*, 33, FCC Rcd at 8416, para, 42; 47 CFR § 64.2117.

[23] *Third RCC Order*, 33, FCC Rcd at 8408, para, 20; 47 CFR § 64.2115.

A list of registered intermediate SPs, published as a .csv file, may be downloaded at any time at https://www.fcc.gov/ipr-ext.

6.2   Contractual Arrangements

The responsibilities of intermediate SPs related to call completion can be defined in an agreement between an SP and the intermediate SP with which it contracts. This contractual arrangement may determine and specify procedures for testing, updating, implementing, and maintaining the geographic area(s) and/or NPA/NXXs for which a given intermediate SP has the responsibility for completing calls to the appropriate terminating SP. Terms and conditions in the contract can also define acceptable service levels so the SP can ensure the intermediate SP complies with performance expectations and hold the intermediate SP accountable. Should the intermediate SP not be performing at the agreed upon service level, appropriate action can be taken as defined in the contract or as otherwise agreed to by the parties. Such actions could include, but are not limited to, temporarily or permanently removing the intermediate SP from the routing path.

6.3 Manage the Number & Identity of Intermediate SPs

As the number of SPs handling a call increases, there is the potential for lengthier call setup delay and other impairments. Troubleshooting may also prove more difficult. Some SPs have found it useful to limit intermediate SPs to include no more than one additional SP (not including the terminating SP) in the call path. Some SPs have also found it useful to insist on transparency with respect to who is handling their traffic. Intermediate SPs may be required through contractual arrangements to disclose the identity of any additional underlying SPs that they use either in advance or as part of call completion troubleshooting and investigation. If SPs are aware of which downstream SPs are involved in handling their traffic, they can perform due diligence and possibly better manage call completion issues.

6.4 Management of Direct & Indirect Looping

There may be cases where a SP purchases wholesale service from an intermediate SP that is hand the call back to the initial SP. This behavior may result in looping as well as adding delay and other impairments in the call setup. Effective network management, such as frequent review of routing tables, can help identify this issue and all SPs involved should then work to eliminate the issue.

6.5 Crank-Back on Failure to Find a Route

If an intermediate SP cannot find a route for termination, it should not drop the call but rather release the call back-crank back- to the original IXC in a manner that allows the IXC to attempt to complete the call over its own facilities.

6.6 Maintain Sufficient Direct Termination Capacity

In conjunction with crank-back, if the first intermediate SP in a call path maintains sufficient termination facilities, it can complete its own traffic when an intermediate SP cannot complete the call.

6.7 Do Not Terminate & Re-Originate Calls

Intermediate SPs should not process calls so as to terminate and re-originate them. Doing so may affect both the signaling information delivered to the called network/party and the likelihood of successful completion. Additionally, if termination/re-origination results in sending an answer indication back to the original IXC before the final Called Party answers, the Calling Party may receive a ringing indication before the terminating SP has signaled that the Called Party is being alerted to an incoming call.

6.8 Direct Measures of Quality

IXCs and Covered SPs need to establish Direct Measures of Quality (DMoQs) for their vendors to meet and need to require vendors to report on these metrics. IXCs and Covered SPs also need to monitor these DMoQs directly. The following table provides some metrics that have been found useful.

Table 6.1 - Examples of Direct Measures of Quality Metrics

Call Completion	Voice Quality	FAX	Voiceband Data
Call Completion Rate	One-way voice path delay	Echo Cancellation	Support of Low Baud Rate Modems, i.e., Telecommunications Devices for the Deaf (TDD) and Packet over SONET (POS)
Call Cut-Off Rate	Echo Cancellation	Packet Loss	V.90 modem performance
Post Dial Delay	Mean Opinion Score	Completion Rate	V.34 modem performance
Post Answer Delay	Loss	Error-Free Pages	Echo Cancellation
	Idle Channel Noise	Percentage of pages sent at top speed for completed transmissions	Signal to C-Notched Noise Ratio
	Signal to C-Notched Noise Ratio		Phase Jitter
	Crosstalk		Envelop Delay Distortion
	Clipping		Signal to Total Distortion
			Intermodulation Distortion
	Signal to Total Distortion		Frequency Shift
			Phase Hits
			Dropouts
			Impulse Noise

6.9 Do Not Manipulate Signaling

Intermediate SPs should not manipulate signaling information, especially the CPN, so as to obscure proper jurisdiction for settlements., Also, intermediate SPs must pass the signaling information from downstream SPs on the terminating side, unaltered, to originating SPs in the call path indicating the terminating SP is alerting the called party. Additional information can be found in Clause 8, *Regulatory Environment*.

6.10 Inheritance of Restrictions

Where an intermediate SP makes use of an additional intermediate SP to reach the terminating SP, the first intermediate SP contracting with the IXC should in turn manage the additional intermediate SP to a standard no less than that of the originating IXC.

6.11 Intercarrier Process Requirements

Maintenance responsibilities for the service, including contact points and escalation lists, should be defined in advance. Expectations for repair times, status reporting intervals, and trouble ticket handling procedures should also be agreed to as part of the contacting process.

6.12 Require Acceptance Testing

Before offering live traffic to an intermediate SP, an IXC and/or covered SP should conduct acceptance testing with the intermediate SP to ensure compliance with call processing requirements and DMoQs.

7   Trouble Reporting & Contact Directories

This section provides information on trouble reporting and the ATIS NGIIF Service Provider Contact Directory (SPCD) which includes SP contacts for rural call completion issues. Among other things, this section lays out trouble reporting responsibilities of SPs that provide originating or terminating access for LD calls and their local service customers.

Timely resolution of troubles related to call completion depends on identifying the SP(s) that handle a given call. Originating callers' LD traffic may not be handled by their local SP or its affiliate; where callers are PIC'd to an independent entity, the originating local SP may be restricted from providing that information to a terminating LEC reporting a completion issue. The terminating LEC may need the wireline originating caller to confirm their IXC SP by dialing 700-555-4141 from the calling TN or identifying that they used any other means to complete the call (e.g., dial-around, calling card, or some other means). Once the SP with responsibility for the call has been identified, the reporting and sectionalization processes can be invoked to achieve a resolution.

It is imperative that a case of trouble be reported at the time or near the time when the trouble call occurs. It is recommended that all SPs accept a trouble report at least up to 72 hours from the time of the call in question. Delays in reporting a case of trouble degrades the ability of the SP to identify, isolate, and investigate the cause of the trouble reported. Untimely trouble reporting can result in confusion and delays due to network interconnections and routing tables that have since changed, retention periods for trouble logs that have passed, etc., making the ability to duplicate the specific trouble situation difficult if not impossible. These are just a few examples of why it is important to place a trouble report with the appropriate SP after the problem occurs to enable that SP to successfully investigate the case of trouble. Collaboration is important. Also key is promptly identifying and involving all parties in the call path to investigate and resolve the issue.

The NGIIF has multiple documents addressing trouble detection, reporting, management, and more, for different aspects of telecommunications. Such documents include:

- ATIS-0300009, *NGIIF Reference Document Part I- Installation and Maintenance Responsibilities for Special Access Services, WATS Access Lines, and Switched Access Services Feature Group "A".*

- ATIS-0300010, *NGIIF Reference Document Part II- Installation and Maintenance Responsibilities for Switched Access Services Feature Groups "B," "C," and "D".*

- ATIS-0300011, *NGIIF Reference Document Part III - Installation and Maintenance Responsibilities for SS7 Links and Trunks.*

- ATIS-0300030, *NGIIF Reference Document Part IX- Installation, Testing, and Maintenance Responsibilities for Facilities.*

- ATIS-0300032, *Next Generation Interconnection Interoperability (NGIIF) Reference Document: Part X, Interconnection Between LECS Operations Handbook – Local Interconnection Service Arrangement.*

- ATIS-0300035, *NGIIF Reference Document Part XII- Toll Free Industry Test Plan.*

- ATIS-0300082, *Guidelines for Reporting Local Number Portability Troubles in a Multiple Service Provider Environment.*

The following series of documents cover aspects of SS7:

- ATIS-0300011, *NGIIF Reference Document Part III - Installation and Maintenance Responsibilities for SS7 Links and Trunks.*

- ATIS-0300012, *NGIIF Reference Document Part III- Attachment A- MTP Compatibility Tests.*

- ATIS-0300013, *NGIIF Reference Document Part III- Attachment B- ISUP Compatibility Tests.*

- ATIS-0300014, *NGIIF Reference Document Part III- Attachment C- SCCP Protocol Class 0 Compatibility Tests*.

- ATIS-0300015, *NGIIF Reference Document Part III- Attachment D- Test Severity Analysis Criteria*.

- ATIS-0300016, *NGIIF Reference Document Part III- Attachment E- SS7 Network Gateway Screening*.

- ATIS-0300017, *NGIIF Reference Document Part III- Attachment F- SS7 ISUP Tests for ISDN Network Interconnection*.

- ATIS-0300018, *NGIIF Reference Document Part III- Attachment G- SS7 Link Diversity Validation Guidelines*.

- ATIS-0300020, *NGIIF Reference Document Part III- Attachment I- SS7 Security Base Guidelines*.

- ATIS-0300021, *NGIIF Reference Document Part III- Attachment J- SS7 Software Validation*.

- ATIS-0300022, *NGIIF Reference Document Part III- Attachment K- Dual STP Failure Prevention Procedures*.

- ATIS-0300023, *NGIIF Reference Document Part IV- Installation and Maintenance Responsibilities for X.75 Gateway Services*.

There are common aspects of trouble detection, reporting, and management which have been captured in Clause 6.1.

7.1 Trouble Reporting

It is recommended that non-circuit-specific troubles be reported immediately in order to facilitate the rapid restoral of service.

Existing trouble handling procedures for interexchange calls focus on the case where a trouble is reported by calling customers to their SP and where there is a direct connection between the IXC and the terminating local SP. In many of the scenarios of concern in this document, the party reporting trouble is a Called Party who has failed to receive a call. If the Calling Party can be induced to report the trouble to their IXC[24], normal procedures can be used to resolve the issue. Where this is not possible, the Called Party's local SP should contact the SP they believe to be the Calling Party's serving IXC.[25] Except by report of the Calling Party directly or, as reported by the Calling Party to the Called Party, the terminating SP may not be able to identify the responsible IXC directly. Instead, it may determine the caller's local SP and contact it. Where the local SP is also the Calling Party's LD SP, the trouble can be addressed by the LD entity on behalf of the Calling Party.

If the report is made by a separate carrier-to-carrier channel, which some IXCs have set up in response to the RCC situation, and the caller's local SP also happens to be the caller's IXC as well, the IXC may be able to address the problem, although CPNI restrictions may prevent the IXC from working the trouble without first contacting their customer and obtaining permission. Customer Proprietary Network Information (CPNI) restrictions may also prevent sharing full details of resolution with the Called Party's SP. Where the caller's local SP is not the caller's IXC, CPNI restrictions will prevent the caller's local SP from revealing the PIC'd IXC's identity. It is possible that changes/clarifications to CPNI rules would facilitate the helpful sharing of information between SPs related to trouble resolution.

SPs are responsible for the acceptance of trouble reports from their end user. The SP accepting and responsible for the case of trouble should first test to determine if the trouble is in their network. If the trouble is found in its network, the responsible SP will clear the trouble and no referral to other SPs is necessary. If the trouble is sectionalized by the responsible SP towards another SP, then the trouble report will be referred to that receiving SP. The receiving SP will clear the trouble or will work cooperatively with any other SPs to sectionalize the trouble where necessary.

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[24] To confirm the identity of their IXC, the Calling Party should dial 1-700-555-4141 for wireline originations.
[25] Carriers should contact the appropriate IXC via the NGIIF's SPCD.

The following information should be exchanged when handing off or referring the trouble:

- Trouble report number or equivalent
- Contact TN
- Contact ID (i.e., name or initials)
- Time and date report was received from the responsible SP
- Responsible SP testing information (If requested by any receiving SP(s)
- Circuit ID [41-Character Common Language® Message Trunk Circuit Codes (CLCI™ MSG Code)]
- Non-circuit specific (Circuit ID may not be appropriate)
- Trouble reported
- Other information that may be of assistance (e.g., history, subsequent reports)
- Dispatch authorization

7.1.1 Applicability

Clause 7.1 provides guidelines for trouble reporting; however, it does not replace or supersede any tariffs, contracts, or other legally binding documents. In case of conflict between this document and any legally binding document, such other document will prevail.

7.1.2 Responsibilities

SPs working a trouble report have the following baseline responsibilities when investigating their trouble report with other SPs.

7.1.2.1 SP Generating Trouble Reports

- Provide trained personnel
- Advise the relevant SPs when there is a potential service-affecting network failure
- Provide contact information for trouble reporting
- Maintain complete and accurate installation and repair records
- Provide access to test lines where appropriate
- Accept trouble reports from its end users
- Accept trouble reports from other SPs
- Ensure the test equipment used is compatible with the other relevant SP's test equipment
- Assume control functions for maintenance of its trunk(s)
- Consult with other relevant SPs before requesting any changes, except under emergency conditions
- Sectionalize and clear any trouble in its own network
- Test cooperatively with other relevant SPs to identify and clear a trouble, when the trouble has been sectionalized to a network
- Keep its end user advised of the status of all trouble report(s)
- Perform cooperative analysis to determine if a trouble pattern exists
- Refer troubles to other SPs or analytics engine providers using the trouble reporting procedures
- Dispatch its own maintenance forces

- Perform verification tests to ensure that trouble has been cleared

- Participate cooperatively with other SPs or analytics providers to further isolate and clear the trouble when trouble exists and cannot be sectionalized to a particular SP portion

- Where it is technically feasible, send signaling for all internetwork calls to a 10-digit TN using 10-digits for the Called Party number, regardless of how the call is dialed

7.1.2.2 SP Receiving Trouble Reports

- Provide trained personnel

- Advise the relevant SPs when there is a potential service affecting network failure

- Provide contact information for trouble reporting

- Maintain complete and accurate installation and repair records

- Consult with other relevant SPs before requesting any changes, except under emergency conditions

- Provide access to test lines where appropriate

- Notify the receiving SP of any changes affecting the service requested, including the service due date

- Accept trouble reports from SPs generating trouble reports

- Sectionalize and clear any trouble in its own network

- Test cooperatively with other relevant SPs to identify and clear a trouble when the trouble has been sectionalized to another SP's network

- Perform cooperative analysis to determine if a trouble pattern exists

- Refer troubles to other relevant SPs or analytic engine providers using the trouble reporting procedures

- Dispatch its own maintenance forces

- Perform verification tests to ensure that trouble has been cleared

- Participate cooperatively with other SPs to further isolate and clear the trouble when trouble exists and cannot be sectionalized to a particular SP portion or involve the application of analytics

- Provide status reports to the SP who generated the trouble report

7.1.2.3 CPNI

CPNI must be protected as SPs work cooperatively to resolve trouble reports related to call completion/call termination. Refer to Clause 8.5 for further explanation of CPNI.

7.1.2.4 Sectionalization

Sectionalization is a joint responsibility of the SPs, with control for sectionalization under the direction of the SP that generated the trouble report. It is anticipated that sectionalization may involve cooperative testing; both entities are expected to participate in this activity when requested.

7.1.2.5 Non-Trunk Specific Troubles

Non-trunk specific troubles are those that are not directly attributable to a given trunk. Non-trunk specific troubles generally fall into the following categories:

- Reorder;

- No ring;

- Wrong number or misdirected;

- Transmission impairment;

- Cut-off;

- No answer supervision;

- Analytics false positive or false negatives; and

- Other.

When the non-trunk specific trouble has been detected and sectionalized, the trouble report will be referred to the appropriate SP's trouble reporting center or equivalent.

7.1.2.6 Trouble Report Clearing Information

When the trouble has been cleared by either SP, the trouble report will be closed out with the originating SP and generic status information will be updated bearing in mind CPNI rules.

7.2 End-To-End/Intercarrier Testing

Circuit networks comprising SP services may experience trouble conditions that cannot be isolated by each SP testing and maintaining its own services. Although the call delivery provided by each SP may show in each SP's respective network as performing properly, trouble may be identifiable on an end-to-end test, i.e., from origination to termination of the call. In such cases, the SP generating the trouble report may require coordinated intercarrier testing.

7.2.1 Use of Test Lines for Call Completion Trouble Resolution

In cases where the trouble has been reported by the Called Party rather than the Calling Party, one way in which terminating SPs may be able to expedite trouble resolution is to provide a test line number for the destination end office in their trouble report. As discussed in Clause 7.1.2.3, CPNI issues can complicate working troubles in the Called-Party-complaint scenario. An SP can call a test line without involving its customer. Moreover, a test line call will eliminate any issues that may be specific to the Called Party's access and CPE arrangements.

As a best practice, per the following NGIIF Guidelines, SPs should publish test numbers associated with specific NPA/NXXs in the LERG Routing Guide, so that originating SPs can test call quality proactively and test when any customer or SP refers a call quality issue to the originating SP. Without such capability, the originating SP can only test its portion of the network and must rely upon the third party IXC to test its portion of the network that may be involved in the call flow.

- From NGIIF Guideline ATIS-0300024*, NGIIF Reference Document Part V- Test Line Guidelines*: All telecommunications companies (wireline, cable, IP, wireless etc.) applying to the NANPA administrator and receiving a new NPA/NXX or Thousands block(s) are expected to follow testing procedures as Code Holder. More information can be found in the full Guideline.

- From NGIIF Guideline ATIS-0300024: The SP opening a new NPA/NXX shall establish a working test number for call through purposes. The test number shall be established for a minimum period of 180 days. More information can be found in the full Guideline.

7.2.2 Types of End-to-End/Intercarrier Testing

Best practices may include end-to-end/intercarrier testing. This testing for call completion can include manual testing, automated testing, or a combination of both manual and automated to resolve the call completion issue(s).

7.2.2.1 Manual Testing

Manual test calls may be required to properly isolate and investigate the issue, and could include any or all of the following steps, depending on the scope and nature of information observed and identified during trouble reporting:

- Coordinated Intercarrier testing may include two or more of the SPs involved in the end-to-end call path
  - Originating SP
  - IXC, if different from the originating SP
  - Tandem SP
  - Intermediate SP(s)[26]
  - Terminating SP (i.e., Rural LEC)
- Using applicable signaling call trace methods/equipment
  - To monitor the exchange of signaling information in real time
  - To confirm signaling messages are coming from all downstream SP(s)
- As a result of manual testing, SPs may find it necessary to confirm the following:
  - Routing arrangements
  - Trunk groups have appropriate level of capacity
  - Network translations are properly configured

Any issues found during manual testing should be promptly corrected and the call path retested until all parties involved conclude no trouble is found and the call(s) completes properly.

7.2.2.2 Automated Testing

Best practices for automated testing call completion to Rural OCNs include utilizing existing data sets and call review processes as well as internal testing resources.

Call Detail Records (CDRs), routing tables, and daily traffic information provide a baseline on which SP can build automated algorithms, and internal analysis tools, to determine OCNs and routes which could have cause for investigation. This can be done by applying specific system coding, designed for selecting components of call paths where network performance could be impacted due to network traffic, and which may have effects on NER, and ASR The SP's internal analysis tool can then filter trouble tickets into the SP's normal trouble ticketing system for normal trouble ticket processes to be applied against. Technicians then have the CDR details, routing information, timing of the issues and information to troubleshoot and test, in order to determine the cause of the issue. Specific testing can then be done using established milliwatt test line numbers to call route destinations.

It is expected that if testing is done daily, over a month, several OCNs will be tested, including (as programmed up front) testing OCN areas with negative spikes. This approach to daily, routine, automated analysis, and testing provides a reliable and controlled process, and delivers targeted, statistically correct, fixed testing, using established processes, in a near real time approach, to address the network performance.

7.3   Call Setup Time Trouble Reporting & Sectionalization

ATIS-0300010, *NGIIF Reference Document Part II- Installation and Maintenance Responsibilities for Switched Access Services Feature Groups "B," "C," and "D",* defines installation and maintenance responsibilities of LECs providing switched access service [Access Service Providers (ASPs)] and IXCs obtaining switched access service from them [Access Service Customers (ASCs)]. The ASP and ASC terms used in this section are the same as in

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[26] As defined in 47 C.F.R. § 64.1600(f).

ATIS-0300010. The document thus focuses on situations in which IXCs and LECS are directly connected. However, Intermediate SP may also be customers of ASPs. Further, in many problem cases, calls may not even reach the terminating ASP, yet it is the ASP's customers who are initiating the trouble reports. ATIS-0300010 nonetheless clarifies the IXC's responsibilities in and procedures for resolving end-to-end troubles such as overlong post dial delay.

- ASC: The ASC has the overall installation and maintenance responsibility for the total service to its end user. It is responsible for the overall coordination of installation and testing of its services.

- ASP: The ASP is responsible for ensuring that the Switched Access Services (SAS) furnished to an ASC are installed and function properly. In addition, the ASP should work cooperatively with the ASC in the acceptance testing of the SAS it provides.

Where the ASC is unable to perform cooperative testing at its POT, the ASP will provide test results from the nearest ASP test access point, toward the ASC's POT. An Access Service Provider Coordinator (ASPC) will perform the control function for the installation of Feature Group (FG) B, C, and D SAS provided to the ASCs.

End user reported troubles of excessive call setup time, for interLATA FG-D originating and/or interLATA FG B/D terminating will be analyzed by the ASC. If the ASP receives a call setup time trouble from an end user for an interLATA call(s), the end user will be referred to the ASC.

NOTE: See flowchart – Figure 7.1- Call Setup Time (CST) Testing Methodology.

Upon receiving a call setup time trouble report, the ASC will obtain specific information from the end user to aid in the trouble analysis process. The dialogue should include, but is not limited to, the following questions:

- Type of CPE, etc.

- Type of access (e.g., 101XXXX, DDD)

- Directionality of the call(s) on which trouble was reported

- Calling and called TN

- Time of day the reported problem is experienced

- End user's estimation of call setup time

- Any other pertinent information that can be supplied by the end user

Contributing factors to call setup time troubles could include:

- Manual/auto dialing

- Customer call forwarding options

- Private Branch eXchange (PBX) equipment

- Dial repeating tie lines

The ASC is responsible for sectionalizing the call setup time trouble to the:

- Terminating CPE

- Terminating ASP

- ASC network

- Originating ASP

- Originating CPE

Should the sectionalization/analysis require that a test call(s) be made, it is recommended that the test call be made to the 102 type test line. Testing to a 105 type test line may distort the intended call setup time results.

Originating ASP and/or ASC test calls to the ASC's first point of switching should be placed to 1-700-958-1102 or 1-700-959-1020 as appropriate (see Figures 7.2 and 7.3).

Access performance limits have been established based on the information contained in the Local Switching System Generic Requirements (LSSGR) [see GR-317-CORE, *LSSGR: Switching System Generic Requirements For Call Control Using The Integrated Services Digital Network User Part (isdnup)* and GR-394-CORE, *LSSGR: Switching System Generic Requirements For Interexchange Carrier Interconnection (ICI) Using The Integrated Services Digital Network User Part (ISDNUP)*] and other performance criteria, to aid in the isolation of any suspected trouble associated with call setup time. The ASC should specifically identify any parameters that have been exceeded when referring the trouble.

The ASP will accept a trouble report from the ASC when sectionalized to the ASP's network. The trouble report should include, but is not to be limited to, the following information:

- ASC determined ASP call setup time
- Call direction
- FG B-D
- Direct versus tandem routing
- End Office CLLI Code
- Test line TN used

Upon receipt of the trouble report from the ASC, the ASP will initiate its own analysis and treat the report as an impaired trunk report. This analysis will include the following components as necessary:

- *Pattern analysis* – the process of analyzing known information to determine particular scenarios where certain events are repeated.
- *Translations verification* – particularly trunk group routing, timing, and overlap outpulsing operation.
- *Placing of test calls* – including those identified in Figures 7.2 and 7.3. A description of those tests follows:
  - The ASP places a call from the line side of the originating end office to a 102 test line in the ASC Switch (first point of switching in the ASC Network). It is recommended that dialing 1-700-958-1102 or 1-700-959-1020 as appropriate to access the ASC 102 test line.
  NOTE: See Call Setup Time: 7.2 – Originating Test Procedures.
  - The ASC places a call from a test access point in the last point of switching in the ASC network to the 102 test line in the terminating ASP end office. Terminating ASC-ASP test calls should be placed to 7-digit directory number of the end office 102 test line.
  NOTE: See Call Setup Time: Figure 7.3 – Terminating Test Procedures.

If the ASP determines there is a problem in its network, it will exercise diligence in repairing the out-of-limits parameters. If the trouble cannot be found in the ASP's network, this information will be communicated to the ASC. If the ASC and ASP agree there appears to be no call setup time problem, the ASC will discuss this with the end user. If the end user is still encountering a call setup time trouble, further analysis/joint testing may be conducted between the ASC and ASP.

END USER CST
TROUBLE REPORT

REFERRAL

EC ....→ IC

**CST Testing Methodology for
1+ FG D origination and
FG B/D terminating**

**Stored Program Control
Equal Access End Offices Only**

CUSTOMER
DIALOGUE

INITIAL IC
ANALYSIS

IC/END USER
COMMUNICATIONS ←**NO** TROUBLE
INDICATED

INITIAL IC
END-TO-END
TEST CALL

IC ROUTINE
TESTING

NO TROUBLE
INDICATED

YES

IC ISOLATION
ANALYSIS

IC RESOLUTION ←**YES** IC/CPE
PROBLEM

NO

EC TROUBLE
REFERRAL

IC/EC
ANALYSIS

EC ANALYSIS

YES

NO ADDL
ANALYSIS
REQD ← RETURN
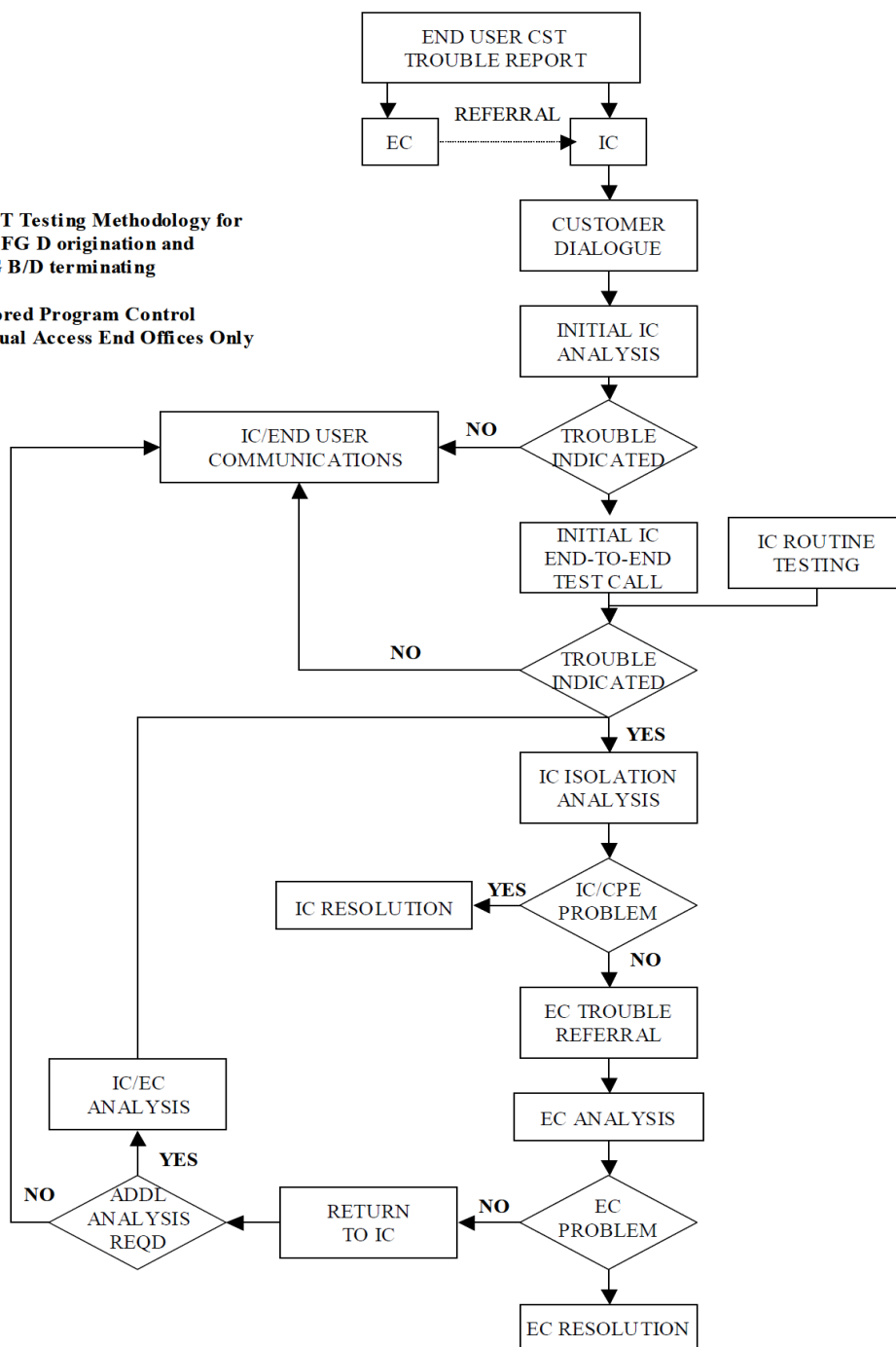TO IC ←**NO** EC
PROBLEM

EC RESOLUTION

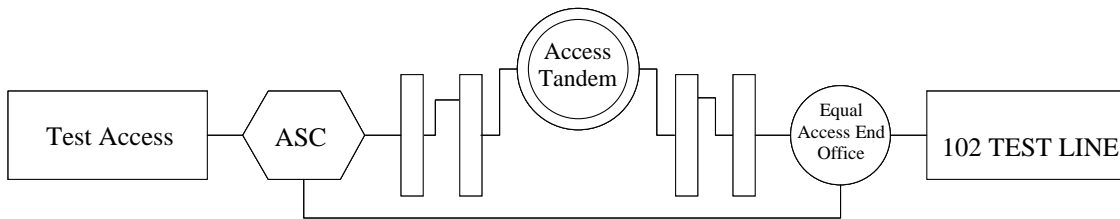**Figure 7.1 - Call Setup Time (CST) Testing Methodology**

Figure 7.2 - Call Setup Time (CST)

7.3.1 Originating Test Procedure

1. Place a call from line side of the originating end office to 102 test line in ASC switch. It is recommended that the ASC 102 test line be accessed by dialing 1-700-958-1102 or 1-700-959-1020, as appropriate.

2. Time stamp:

   • Start at end of last digit dialed.

   • Stop at network response.

   This may be accomplished with personal computer, stopwatch, or other test equipment, as available.
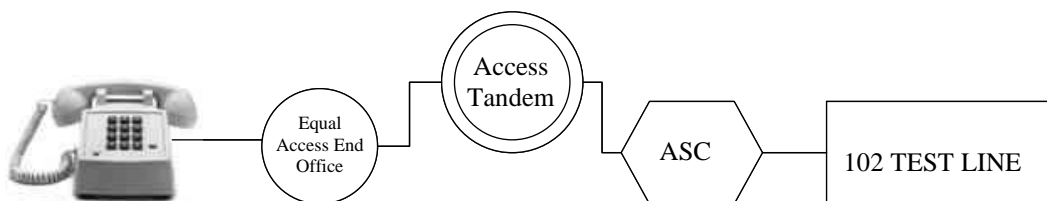

Figure 7.3 - Call Setup Time

7.3.2 Terminating Test Procedures

1. Place a call from the Test Access in the ASC switch to 102 test line in End Office (EO):

2. Time stamp:

   • Start at ASP trunk seizure

   • Stop at network response

   This may be accomplished with personal computer, stopwatch, or other test equipment, as available.

3. ASC switch time not included

7.4 Contact Directories

Industry accessible Contact Directories are available to SPs for trouble reporting.

7.4.1 ATIS NGIIF SPCD & National LNP Contact Directory

The ATIS NGIIF developed and maintains two contact directories, the Service Provider Contact Directory (SPCD) and National LNP Contact Directory. The Directories are available at no charge to the telecommunications industry.

ATIS sends an annual invitation requesting new and/or updated contact information; however, submissions are accepted at any time during the year.

7.4.1.1 Service Provider Contact Directory (SPCD)

The purpose of the SPCD document is to provide contact numbers to the telecommunication industry for requesting interconnecting SP assistance on service-related situations, applying to emerging technology, consolidated centers, multiple platforms (TDM, wireless, IP), or SP specific departments.

The SPCD identifies intercompany contact points for providing information in a multi-platform technology arena. Contact information for those who address trouble resolution activities (e.g. modifications, outages, network survivability, network congestion, testing and/or maintenance, IXC carrier-to-carrier information, analytics engine/provider identification) should be provided for inclusion in the SPCD. Of particular relevance to resolving call completion issues is inclusion of IXC carrier-to-carrier information, subpoena, traceback, and analytic engine provider information, such that SPs have a readily available source for appropriate contacts at other SPs or analytics engine providers. It is recommended that all SPs list and update their contacts on a regular basis.

7.4.1.2 National LNP Contact Directory

The purpose of the LNP Contact Directory is to provide contact numbers to the telecommunication industry for requesting interconnecting SP assistance on service-related situations relating to LNP. Any associated LNP contact information related to TDM, wireless, or IP should be included in the LNP Contact Directory. It is a recommendation by the NGIIF that all SPs list and update their contacts on a regular basis.

7.4.1.3 How to Gain Access

The SPCD and the National LNP Contact Directories can be accessed via the NGIIF webpage at https://www.atis.org/01_committ_forums/ngiif/contact-directories/.

7.4.1.4 How to Add or Update Contact Information

To include and/or update your SP's information in the SPCD and/or National LNP Contact Directory, SPs must provide their information in the SPCD and/or LNP form found in the *NGIIF CD Entry Instructions* document request form, located online via the NGIIF webpage at https://www.atis.org/01_committ_forums/ngiif/contact-directories/. SPs then need to submit their completed form (s) to the NGIIF Administrator via email at ngiif-admin@atis.org.

7.4.2 Updating Contact Information and Test Numbers in the LERG Routing Guide

The LERG Routing Guide provides a useful vehicle for SPs to provide up-to-date technical contact information and test equipment numbers for their switches. Specifically, Table 6 contains a data field for switch test numbers and Table 1 contains data fields for contact information for SP personnel. SPs should maintain up-to-date information in the LERG Routing Guide identifying technical points of contact that LD SPs can work with on call completion issues. SPs should also publish their test line TNs in the LERG Routing Guide [i.e., 102 Milliwatt, 105 remote office test lines (ROTL), etc.] in order to facilitate LD testing to their switches. Being able to quickly test connectivity and establish contact with SP technical personnel often facilitates troubleshooting and remediation of RCC issues.

8 Regulatory Environment

This section describes various FCC rules and regulations, as of the date of publication of this document, with the intent to assist in investigating and/or resolving some of the LD call completion/call termination issues addressed herein. Noted references are not all inclusive nor are they intended to provide legal guidance and, based on date of this document, some may have been subsequently revised. State commissions may also have issued rules and regulations on the subject addressed by this document.

8.1 Rural Call Completion (RCC) Orders

8.1.1 The FCC's Report and Order (R&O) and Further Notice of Proposed Rulemaking (FNPRM), FCC 13-135, WC Docket No.13-39

Adopted October 28, 2013, and released November 8, 2013, this Order created reporting and retention requirements for certain SPs related to call completion. The *RCC Order* was published in the Federal Register on March 4, 2015, with an effective date of April 1, 2015, to implement the reporting and retention requirements.

The *RCC Order* amended Part 64, *Miscellaneous Rules Relating to Common Carriers*, of the FCC's rules (as set forth in Appendix A to the RCC Order), including:

- Definitions of terms used throughout the *RCC Order*, such as Call Attempt, Covered SP, Intermediate SP, Rural OCN

- Retention and Reporting Requirements ("Appendix C" or "Form 480")[27]

- Safe Harbor

- Disclosure of Data

- Ringing Indication Requirements

8.1.2 The FCC's Report and Order and Further Notice of Proposed Rulemaking, FCC 17-151, CG Docket No. 17-59

Adopted November 16, 2017, and effective February 11, 2018, this order allows SPs to block calls from invalid, unallocated and/or unassigned telephone numbers.

8.1.3 The Improving Rural Call Quality and Reliability Act

This Act requires "intermediate providers" to register with the FCC unless they meet the safe harbor exclusion. One key item to underscore is that while the title of the bill references rural areas, the statute and the FCC implementing rules are not limited to rural areas.

8.1.4 The FCC's Second R&O and Third Further Notice of Proposed Rulemaking (FNPRM) FCC 18.45, WC Docket No. 13-39

This rule was adopted April 17, 2018, to make available a point of contact to address rural call completion issues effective October 24, 2018.

8.1.5 The FCC's Third R&O, WC Docket No. 13–39

Adopted August 13, 2018, this rule:

- Governs the registration of intermediate SPs
- Requires covered SPs (i.e. originating SPs) to only use registered intermediates SPs anywhere in the call flow
- Requires covered SPs to be capable of disclosing to the FCC the identity of all intermediate SPs in call path

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[27] This requirement has since been removed.

8.2 USF/ICC Reform Order

The FCC's *R&O* and *FNPRM* in CC Docket Nos. 96-45 and 01-92; GN Docket No. 09-51; WC Docket Nos. 03-109, 05-337, 07-135 and 10-90, and WT Docket No. 10-208, adopted October 27, 2011 and released November 18, 2011 (FCC 11-161), and as amended by the FCC on December 23, 2011 (FCC 11-189) (the "*USF/ICC Reform Order*") modified FCC rules related to the USF and ICC system. Thus, SPs should be mindful of affected rules relating to LD call completion/call termination.

8.2.1 Phantom Traffic

In ¶703 of the *USF/ICC Reform Order*, the FCC states that "phantom traffic" refers to traffic that terminating networks receive that lacks certain identifying information. Amended FCC rules relating to phantom traffic are found in 47 CFR § 64.1600 and 47 CFR § 64.1601 (a). Specifically, new ¶(f) in §64.1600 adds the term "Intermediate Provider." The term Intermediate Provider means any entity that carries or processes traffic that traverses or will traverse the PSTN at any point insofar as that entity neither originates nor terminates that traffic. The FCC revised §64.1601 (a) to read as follows:

> *§ 64.1601 Delivery requirements and privacy restrictions.*
>
> (a) Delivery*.* Except as provided in paragraphs (d) and (e) of this section:
>
> (1) Telecommunications SPs and SP of interconnected Voice over Internet Protocol (VoIP) services, in originating interstate or intrastate traffic on the public switched telephone network (PSTN) or originating interstate or intrastate traffic that is destined for the PSTN (collectively "PSTN Traffic"), are required to transmit for all PSTN Traffic the telephone number received from or assigned to or otherwise associated with the calling party to the next SP in the path from the originating SP to the terminating SP. This provision applies regardless of the voice call signaling and transmission technology used by the SP or VoIP SP. Entities subject to this provision that use Signaling System 7 (SS7) are required to transmit the calling party number (CPN) associated with all PSTN Traffic in the SS7 ISUP (ISDN User Part) CPN field to interconnecting SP, and are required to transmit the calling party's charge number (CN) in the SS7 ISUP CN field to interconnecting SP for any PSTN Traffic where CN differs from CPN. Entities subject to this provision who use multi-frequency (MF) signaling are required to transmit CPN, or CN if it differs from CPN, associated with all PSTN Traffic in the MF signaling automatic numbering information (ANI) field.
>
> (2) Intermediate SP within an interstate or intrastate call path that originates and/or terminates on the PSTN must pass unaltered to subsequent SP in the call path signaling information identifying the telephone number, or billing number, if different, of the calling party that is received with a call. This requirement applies to SS7 information including but not limited to CPN and CN, and also applies to MF signaling information or other signaling information intermediate SP receive with a call. This requirement also applies to VoIP signaling messages, such as calling party and charge information identifiers contained in Session Initiation Protocol (SIP) header fields, and to equivalent identifying information as used in other VoIP signaling technologies, regardless of the voice call signaling and transmission technology used by the SP or VoIP SP.

Of particular importance, footnote 1196 says: "…Although 47 C.F.R. §64.1601 requires that the CPN be transmitted where technically feasible, the technical content and format of SS7 signaling is governed by industry standards rather than by Commission rules."

IP signaling is addressed in ¶717: "the rules we adopt today also apply to interconnected VoIP traffic." Note that the signaling rules do not yet apply to one-way VoIP. Finally, ¶723 and footnote 1249 advise: "Parties seeking limited exceptions or relief in connection with the call signaling rules we adopt can avail themselves of established waiver procedures at the Commission. To that end, we delegate authority to the Wireline Competition Bureau to act upon requests for a waiver of the rules adopted herein in accordance with existing Commission rules."; 47 C.F.R. § 1.3.

8.2.2 Caller ID

In addition to the rules set forth in the *USF/ICC Reform Order*, in the *Report and Order*, In the Matter of Rules and Regulations Implementing the Truth in Caller ID Act of 2009, WC Docket No. 11-39, FCC 11-100, adopted June 20, 2011 and released June 22, 2011 ("Truth in Caller ID *Order*"), the FCC revised CPN rules to be modeled on the

Communications Act of 1934, as amended ("the Act") prohibition against knowingly engaging in caller ID spoofing with fraudulent or harmful intent.. Additionally, the FCC stated at ¶20 of the Truth in Caller ID *Order* that the person or entity that knowingly causes caller ID services to transmit or display misleading or inaccurate information may, in some cases, be a SP, spoofing SP or other SP, and the FCC does not exempt such conduct from the purview of the FCC rules. New §64.1604 was added as a result of the Truth in Caller ID *Order*. Specifically, §64.1604 (a) reads as follows.

> *§ 64.1604 Prohibition on transmission of inaccurate or misleading caller identification information.*

> (a) No person or entity in the United States shall, with the intent to defraud, cause harm, or wrongfully obtain anything of value, knowingly cause, directly or indirectly, any caller identification service to transmit or display misleading or inaccurate caller identification information.

8.2.3 Calling Party Number (CPN)

The *USF/ICC Reform Order*, at ¶ 704, modifies call signaling rules as follows.

- SPs that originate interstate or intrastate traffic on the PSTN, or that originate inter- or intrastate-interconnected VoIP traffic destined for the PSTN, will now be required to transmit the TN associated with the calling party to the next SP in the call path.

- Intermediate SP must pass calling party number or charge number signaling information they receive from other SP unaltered, to subsequent SP in the call path.

8.3 Practices to Support Proper Jurisdictionalization of Traffic

The Called Party's SP's expected termination path, for the routing designation, is based on the regulatory requirements in 47 C.F.R. §51.701(b), as well as the SP's filed tariff(s), if any.

Additional information regarding jurisdiction of traffic is spelled out in the Local Competition *First Report and Order, In the Matter of Implementation of the Local Competition Provisions in the Telecommunications Act of 1996* (CC Docket 96-98) *and Interconnection between LECs and Commercial Mobile Radio Service (CMRS) Providers* (CC Docket No. 95-185), adopted August 1, 1996 and released August 8, 1996 (FCC 96-325) ("Local Competition First R&O").

8.4 CPNI

In the regulatory environment, one key area to be mindful of in investigating information regarding LD calls between SPs is the regulatory construct related to CPNI. The CPNI rules are called out below.

In the Act, "customer proprietary network information" consists of information relating to the "quantity, technical configuration, type, destination, location, and amount of use of a telecommunications service subscribed to by any customer of a telecommunications carrier." 47 U. S. C. § 222(h)(1). This statutory definition of "customer information" encompasses customers' particular calling plans and special features, the pricing and terms of their contracts for those services, and details about who they call and when.

8.5 LNP

Where the Called Party's TN has been ported, an LNP dip is required to be done. See LNPA best practices.[28] Regulations related to LNP can be found in 47 C.F.R. §52.26.

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[28] https://numberportability.com/industry-info/lnpa-working-group/lnp-best-practices/

9 Summary

This handbook is intended for all SPs involved in LD call completion. It attempts to identify the existing standards and guidelines that may be relevant to call completion problems that have been reported by rural telephone companies and to delineate the responsibilities of different industry segments in using these standards and guidelines to avoid call completion failures. The handbook also outlines trouble handling procedures and discusses how the new call scenarios in today's more diverse, converged, and complex networks may complicate trouble resolution. It offers best practices for management of underlying or intermediate SPs. Finally, it summarizes some of the applicable current regulatory environment and identifies obligations.

It is important to understand that the PSTN is not engineered for on hundred percent call completion at all times and that variations in completion rates are subject to variations in offered load on a diurnal and seasonal basis as well as due to extraordinary circumstances such as disasters and media-stimulated calling. Despite the redundancy engineered into many components of the network, there will be occasional failures resulting in outages and, despite the care that most SP take to prevent them, there will be human errors that result in calls failing. It is important to distinguish transient variations in call completion rates due to the factors noted above, and not to treat all instances of call failure as discriminatory behavior.

It is intended that this handbook will help mitigate the more serious issues that led to its development. This handbook is viewed as a living document that will be updated over time to reflect further learnings and any changes to pertinent standards and regulations that may arise.