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September 5, 2012

VIA EMAIL
Julius Genachowski
Chairman
Federal Communications Commission
445 12th Street SW
Washington, DC 20554

Re: Completion of ATIS NGIIIF Call Completion Handbook (ATIS-0300106)

Dear Chairman Genachowski:

On behalf of the Alliance for Telecommunications Industry Solutions (ATIS), I am pleased to attach a copy of the ATIS Intercarrier Call Completion/Call Termination Handbook (ATIS-0300106). This handbook has been developed by the ATIS Next Generation Interconnection Interoperability Forum (NGIIF) as part of its work program to address call completion issues being experienced by customers of rural telecommunications carriers.

While the handbook may not address every circumstance in which a call completion issue may arise, it does provide valuable information to the industry. In particular, the handbook describes new and existing industry standards and practices to assist in ensuring call completion, such as signaling, trouble handling, and routing, which includes examples of potential call failure points. The handbook also lists best practices that may be useful in addressing call completion problems, especially for management of intermediate or underlying carriers.

The effort was in large part developed by the network interconnection and interoperability experts that work on the NGIIF. These include representatives: Alcatel-Lucent; AT&T; Bell Canada; CenturyLink; Hypercube; INdigital Telecom; IP Fabrics, Inc.; MicroDATA; National Communication System (NCS); Sprint Nextel; T-Mobile USA, Inc.; Telephone and Data Systems, Inc.; Time Warner Cable; and Verizon. More information on the NGIIF is attached. Valuable input was also received from state public services commissions, rural carriers and rural telco associations. This information was collected during NGIIF's open industry workshops and through its detailed industry surveys.

The NGIIF will continue its work to address call termination issues and anticipates that this document will evolve as its work continues. The NGIIF is also working collaboratively with rural associations on possible future call completion testing, which may provide additional information that may be useful in future updates to the handbook.

ATIS is taking steps to ensure that this valuable information is available to all industry stakeholders. The handbook is being made available free of charge to the public via the ATIS Document Store at http://www.atis.org/docstore/product.aspx?id=26780. ATIS has also scheduled a free webinar on this issue – the webinar will take place on Friday, October 5. We would welcome Commission staff attendance at this webinar.

If you have any questions, or if we can provide more information, please feel free to contact the undersigned.

Sincerely,

Thomas E. Goode
General Counsel

Attachment

ABOUT ATIS AND NGIIF

The Alliance for Telecommunications Industry Solutions (ATIS)

As a leading technology and standards development organization with 30 years of proven experience, ATIS brings together top ICT companies to advance the industry's most pressing business priorities. ATIS produces technical and operational standards essential to the operation and interconnection of legacy and next-generation networks, as well as an integrative hub for the communications industry, designing and delivering simple, federated solutions that help members better serve their customers and speed new solutions to market. Regulatory agencies rely upon ATIS standards in lieu of the development of governmental standards in key areas, including numbering, billing and emergency alerts.

Nearly 600 industry subject matter experts work collaboratively in ATIS' open industry committees, which address issues such as cloud services, device solutions, M2M communications, cyber security, ehealth, network evolution, quality of service, billing support, operations, and more. These priorities follow a fast-track development lifecycle—from design and innovation through solutions that include standards, specifications, requirements, business use cases, software toolkits, and interoperability testing.

NGIIF

The ATIS Next Generation Interconnection Interoperability Forum (NGIIF) addresses issues related to the fast-paced growth in emerging new technologies and applications and their impact on the integrity of the telecommunications network. This committee develops operating procedures involving the network aspects of architecture, disaster preparedness, installation, maintenance, management, reliability, routing, security, and testing between network operators. The NGIIF's work addresses some of the leading issues in facilitating the smooth transition from the PSTN to IP-based networks. Most recently, it also has made contributions to address the network traffic implications of auto-dialers.

ATIS-0300106

ATIS Standard on -

INTERCARRIER CALL COMPLETION/CALL TERMINATION HANDBOOK

As a leading technology and solutions development organization, ATIS brings together the top global ICT companies to advance the industry's most-pressing business priorities. Through ATIS committees and forums, nearly 200 companies address cloud services, device solutions, M2M communications, cyber security, ehealth, network evolution, quality of service, billing support, operations, and more. These priorities follow a fast-track development lifecycle—from design and innovation through solutions that include standards, specifications, requirements, business use cases, software toolkits, and interoperability testing.

ATIS is accredited by the American National Standards Institute (ANSI). ATIS is the North American Organizational Partner for the 3rd Generation Partnership Project (3GPP), a founding Partner of oneM2M, a member and major U.S. contributor to the International Telecommunication Union (ITU) Radio and Telecommunications sectors, and a member of the Inter-American Telecommunication Commission (CITEL). For more information, visit < www.atis.org >.

ATIS-0300106, *Intercarrier Call Completion/Call Termination Handbook*

Is an ATIS Standard developed by the **ATIS Next Generation Interconnection Interoperability Forum (NGIIF) Committee**.

Published by

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

Printed in the United States of America.

ATIS-0300106

ATIS Standard on

Intercarrier Call Completion/Call Termination Handbook

Alliance for Telecommunications Industry Solutions

Approved August 2012

Abstract

This handbook describes some of the problems being encountered by rural telephone company 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 carriers 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) Committee addresses next generation network interconnection and interoperability issues associated with emerging technologies. Specifically, it develops operational procedures which 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 which 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* denote 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, 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, Next Generation Interconnection Interoperability Forum, which was responsible for its development, had the following companies on its roster:

Alcatel-Lucent
AT&T
Bell Canada
CenturyLink
Hypercube
INDigital Telecom
IP Fabrics, Inc.
National Communications System
Sprint Nextel Corporation
T-Mobile USA
Telephone and Data Systems, Inc.
Telcordia Technologies
TEOCO
Time Warner Cable
Verizon

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Intercarrier Call Completion/Call Termination Handbook

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 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-based technologies, the number and diversity of these entities has grown. In this context, problems with call completion may arise. The NGIIF has been made aware of serious problems encountered by some rural telephone company customers in receiving long distance calls in particular, through an industry survey by rural associations in March 2011 and the FCC's Rural Call Completion Task Force meeting held in October 2011.

This handbook:

- Is a living document, which will be updated as applicable;
- Describes some of the problems being encountered;
- Discusses some of the 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; and
- Offers some best practices for ensuring call completion, especially in the management of intermediate or underlying carriers.

1.2 Purpose

This handbook was initiated to provide a resource to carriers to address issues as they are encountered related to intercarrier (long distance) call completion/call termination.

1.3 Application

1.3.1 Problems Reported

ATIS understands that, according to an industry survey, customers of some telecommunications service providers (particularly those in rural areas) have experienced difficulties with the receipt of long distance calls via their phone service, including problems that generally fell 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 dead air when the call is answered.
  - 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).
- *Misidentification of calling party*.

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

In addition, carriers reporting these problems attributed them to the acts and omissions of intermediate or underlying service providers handling such calls.

2 References

The following documents/standards contain information which is referenced within this guideline. At the time of publication, the editions indicated were valid. All documents/standards are subject to revision, and the reader is encouraged to investigate the possibility of applying the most recent editions of the standards and/or documents indicated below.

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"*.[2]

ATIS-0300010, *NGIIF Reference Document Part II- Installation and Maintenance Responsibilities for Switched Access Services Feature Groups "B," "C," and "D"*.[2]

ATIS 0300011, *NGIIF Reference Document Part II- Installation and Maintenance Responsibilities for Switched Access Services Feature Groups "B," "C," and "D"*.[2]

ATIS-0300012, *NGIIF Reference Document Part III- Attachment A- MTP Compatibility Tests*.[2]

ATIS-0300013, *NGIIF Reference Document Part III- Attachment B- ISUP Compatibility Tests*.[2]

ATIS-0300014, *NGIIF Reference Document Part III- Attachment C- SCCP Protocol Class 0 Compatibility Tests*.[2]

ATIS-0300015, *NGIIF Reference Document Part III- Attachment D- Test Severity Analysis Criteria*.[2]

ATIS-0300016, *NGIIF Reference Document Part III- Attachment E- SS7 Network Gateway Screening*.[2]

ATIS-0300017, *NGIIF Reference Document Part III- Attachment F- SS7 ISUP Tests for ISDN Network Interconnection*.[2]

ATIS-0300018, *NGIIF Reference Document Part III- Attachment G- SS7 Link Diversity Validation Guidelines*.[2]

ATIS-0300019, *NGIIF Reference Document Part III-Attachment H SS7 Cause Codes and Tones and Announcements*.[2]

ATIS-0300020, *NGIIF Reference Document Part III- Attachment I- SS7 Security Base Guidelines*.[2]

ATIS-0300021, *NGIIF Reference Document Part III- Attachment J- SS7 Software Validation*.[2]

ATIS-0300022, *NGIIF Reference Document Part III- Attachment K- Dual STP Failure Prevention Procedures*.[2]

ATIS-0300023, *NGIIF Reference Document Part IV- Installation and Maintenance Responsibilities for X.75 Gateway Services*.[2]

ATIS-0300024, *NGIIF Reference Document Part V- Test Line Guidelines*.[2]

ATIS-0300030, *NGIIF Reference Document Part IX- Installation, Testing, and Maintenance Responsibilities for Facilities*.[2]

ATIS-0300032, *Next Generation Interconnection Interoperability (NGIIF) Reference Document: Part X, Interconnection Between LECS Operations Handbook – Local Interconnection Service Arrangement*.[2]

ATIS-0300035, *NGIIF Reference Document Part XII- Toll Free Industry Test Plan*.[2]

ATIS-0300051, *Central Office Code (NXX) Assignment Guidelines*.[2]

ATIS-0300082, *Guidelines for Reporting Local Number Portability Troubles in a Multiple Service Provider Environment*.[2]

ATIS-0300105, *NGIIF Auto Dialers Reference Document - Auto Dialers Basics*.[2]

[2] This document is available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G Street N.W., Suite 500, Washington, DC 20005. < https://www.atis.org/docstore/default.aspx >

ATIS-0300209.2003 (R2007), *Operations, Administration, Maintenance and Provisioning (OAM&P) – Network Tones and Announcements*.[2]

ATIS-1000002, *Number Portability Switching Systems*.[2]

ATIS-1000113.2005 (R2010), S*ignaling System No. 7 (SS7) – Integrated Services Digital Network (ISDN) User Part*.[2]

ATIS-1000679.2004 (R2010), *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control or ISDN User Part*.[2]

ATIS-1000607.2000 (R2009), *Integrated Services Digital Network (ISDN) – Layer 3 Signaling Specification for Circuit Switched Bearer Service for Digital Subscriber Signaling System Number 1 (DSS1)*.[2]

IETF RFC 3261, *SIP: Session Initiation Protocol*.[3]

IETF RFC 3325, *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks*.[3]

3GPP TS 24.229, *IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3*.[4]

3 Definitions, Acronyms, & Abbreviations

The *ATIS Telecom Glossary* is an American National Standard that is composed of terms and accompanying definitions that address the disciplines of:

- telephony;
- NS/EP (National Security/Emergency Preparedness);
- NII (National Information Infrastructure);
- spectrum sharing;
- radar;
- radio communications (including HF ALE radio);
- television (including UHF, VHF, cable TV, and HDTV);
- facsimile;
- networks (e.g., intelligent networks, open network architecture, ISDN, broadband ISDN, and network management);
- fiber optic communications;
- communications security;
- data processing;
- premises wiring;
- photonics; and
- telegraphy.

The intent is to provide a uniform, up-to-date set of definitions for the general terminology used in telecommunications.

Terms and definitions used in this document are aligned with the ATIS Telecom Glossary, which can be accessed online at < http://www.atis.org/glossary >.

3.1 Acronyms & Abbreviations

3GPP	3rd Generation Partnership Project
ABD	Average Business Day

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

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

ANI	Application Network Interface
ANSI	American National Standards Institute
AOCN	Administrative Operating Company Number
ATIS	Alliance for Telecommunications Industry Solutions
BICC	Bearer-Independent Call Control
BIRRDS	The Telcordia™ Business Integrated Routing and Rating Database System
BT	Busy Tone
CLEC	Competitive Local Exchange Carrier
CMRS	Commercial Mobile Radio Service
CN	Charge Number
COCAG	Central Office Code (NXX) Assignment Guidelines
CPC	Calling Party's Category
CPE	Customer Provided 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
EO	End Office
ETS	Emergency Telecommunications Service
FCC	Federal Communications Commission
IAM	Initial Address Message
IETF	Internet Engineering Task Force
INT	Intercept
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
ITU	International Telecommunication Union
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
LERG	The Telcordia™ Local Exchange Routing Guide
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
NECA	National Exchange Carriers Association
NGIIF	Next Generation Interconnection Interoperability Forum
NPA	Numbering Plan Areas
NPAC	Number Portability Administration Center
NXX	Central Office Code
OAM&P	Operations, Administration, Maintenance and Provisioning
OLI	Originating Line Information
PBX	Private Branch eXchange
POS	Packet Over SONET
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
RI	Redirection Information
RO	Reorder Tone
ROA	Reorder Announcement
SDP	Session Description Protocol
SIP	Session Initiated Protocol
SS7	Signaling System No. 7
TDD	Telecommunications Devices for the Deaf
TDM	Time Division Multiplexing
TRA	Telcordia Routing Administration
UDI	Unrestricted Digital Information with Tone and Announcement
VCA	Vacant Code Announcement
VoIP	Voice over Internet Protocol
WATS	Wide Area Telephone Service

4 Applicable Standards/Guidelines

This section identifies some of the existing applicable standards and/or guidelines, which may have relevance to issues related to long distance call completion/call termination.

4.1 Signaling

4.1.1 Identification of Calling Party

The origin of a call can be identified in signaling through several parameters. In SS7 ISUP, the Calling Party Number (CPN) parameter is carried in the Initial Address Message and contains the telephone number of the originating end user. [See ATIS-1000113.2.2005 (R2012), section 2.16, and ATIS-1000113.3.2005 (R2012), section 3.7]

CPN is the parameter that determines what the called user sees as caller ID, the telephone number 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 (MF) signaling in the call path, CPN will not be received at the terminating switch.

In SIP signaling, the phone number of the calling end user is populated in P-Asserted-Identity header [RFC 3325][5]. ATIS-1000679.2004 (R2010) defines interworking between SS7 ISUP and SIP, including a mapping between the SS7 CPN parameter and the SIP P-Asserted-Identity header[6].

The NGIIF recommends that the Calling Party Number field should be populated, by the originating network, with a valid 10-digit NANP subscriber line number or directory number. More information can be found in section 7, *Existing Regulatory Environment*.

4.1.1.1 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 some entity in the call path. Missing CPN may interfere with terminating the call – e.g., if anonymous call rejection is engaged. More information can be found in section 7, *Existing Regulatory Environment*.

4.1.1.2 Bad or Incorrect Caller ID

The issue of bad or incorrect Caller ID, as seen by the called party, may be the 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 Original Called Number parameter in the Initial Address Message.) Changes in the CPN delivered may also interfere with terminating the call. More information can be found in section 7, *Existing Regulatory Environment*.

4.1.1.3 Identification of the Chargeable Party

SS7 Initial Address Messages 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 Initial Address Message in the Charge Number (CN) parameter [See ATIS-1000113.2.2005 (R2012), section 2.25A, and ATIS-1000113.3.2005 (R2012), section 3.10]. Standards provide that where the Charge Number is the same as the CPN, the originating exchange can signal just the CPN and a valid OLI parameter [ATIS-1000113.2.2005 (R2012), section 2.1.9.3A]. 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-1000679.2004 (R2010). The IETF work developing 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 in 3GPP TS 24.229, Annex J, 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.

4.1.2 Cause Codes, Tones, & Announcements

Cause codes, tones, and announcements play an important role in call completion. On the one hand, they are key to the identification (and thus resolution) of network problems and, on the other, their misuse may exacerbate problems. Rural telephone companies 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

[5] 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.

[6] 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.

signals, when erroneously applied, not only mislead the caller but may mask call completion problems from detection by the caller's long distance provider. This section provides guidance for the proper use of cause codes, tones, and announcements.

4.1.2.1 NGIIF SS7 Cause Code and 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*.

4.1.2.1.1 General Information

ATIS-0300019, *NGIIF Reference Document Part III-Attachment H SS7 Cause Codes and Tones and Announcements*, describes the tones and announcements that are used to inform customers and 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 operators with additional information and to suggest action that should be taken.

4.1.2.1.2 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[7]
01 = OTHER INTERNATIONAL
10 = ANSI
11 = RESERVED

The Coding Standards are used in the subfields of the cause indicator parameter field:

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.

4.1.2.1.3 Mapping Matrix

[7] Currently known as the ITU-T.

The cause code mapping matrix located in ATIS-0300019, *NGIIF Reference Document Part III-Attachment H SS7 Cause Codes and Tones and Announcements*, identifies the action and direction of the Cause Codes as they traverse the network.

4.1.2.1.3 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.

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 Massage	80-94	95
110	Protocol Error	96-110	111
111	Interworking	112-126	127

4.1.2.1.5 Cause Code Treatments

This section provides the definition of the different announcements that can be applied to the appropriate Cause Codes.

ATIS-0300019, *NGIIF Reference Document Part III-Attachment H*, 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 rural telephone companies; 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 carrier maintaining the retail long distance relationship with caller from detecting call completion problems.

4.1.2.1.5.1 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".*

4.1.2.1.5.2 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".*

4.1.2.1.5.3 Intercept (INT)

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

"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".

4.1.2.1.5.4 Vacant Code Announcement (VCA)

This announcement is played when a customer mis-dialed or the NPA/NXX is not valid. An example of such a recording is as follows:

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

4.1.2.1.5.5 Ineffective Other

There should be announcements with the appropriate verbiage for the following situations, and will be determined by the service provider:

- 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.

4.1.2.1.5.6 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 (60 IPM).

4.1.2.1.5.6 Reorder Tone (RO)

Reorder tone 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 impulses per minute (120 IPM). This tone may be applied in lieu of an announcement.

4.1.2.1.6 Mapping Matrix

Indicated below are examples of Cause Codes received by each network or Customer Premises Equipment (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.

Table 1: Examples of Cause Codes Received and Sent by Each 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

✶ At the end office, treat the call per Cause Code XX for analog or Non-ISDN call.

4.1.2.2 Other Applicable Standards Addressing Tones & Announcements

4.1.2.2.1 ATIS-0300209.2003 (R2007), *Operations, Administration, Maintenance and Provisioning (OAM&P) – Network Tones and Announcements*

This standard addresses Tones and Announcements associated with ineffective call attempts. This standard addresses the following Tones:

- Busy Tone;
- Reorder Tone; and
- Special Information Tones.

Announcements addressed include:

- Reorder;
- No Circuit;
- Vacant Code;
- Intercept; and
- Ineffective Other.

This standard 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-1000113.2005 (R2010) and ATIS-1000607.2000 (R2009), respectively, and the tones and announcements identified in this standard. This mapping is for use in call processing involving two or more interconnecting networks (ICNs) when Signaling System Number 7 (SS7) is used for call control. This standard considers the provision of these tones and announcements by originating, intermediate, and terminating ICNs.

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

The Integrated Services Digital Network (ISDN) User Part defines the protocol which supports the signaling functions required to provide voice and non-voice services in an Integrated Services Digital Network. This standard is based on the Specification of Signaling System No. 7 for international use issued by ITU-T Study Group 11 in the year 2000 and subsequent amendments. This standard is based on and uses, where applicable, the same signaling procedures, parameters, and message types as the internationally specified ISDN User Part of the ITU-T Signaling System No. 7.

Chapter 4 describes the basic signaling procedures for the set-up and clear-down of national and international ISDN and non-ISDN connections. The messages and signals are defined in chapter ATIS-1000113.2 and their format and content are given in chapter ATIS-1000113.3. This chapter also includes generic procedures for supplementary services. Service-specific procedures for supplementary services are contained in separate American National Standards (see clause 2.2 of chapter ATIS-1000113.1).

4.1.2.2.2.1 Address signaling

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

4.1.2.2.2.2 Basic Procedures

The basic call control procedure is divided into three phases: call set-up, the data/conversation phase, 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 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.

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

More information can be found in Chapter 4, Clause 2, about basic call control and signaling procedures. Specifically, this section addresses successful call set-up, and unsuccessful call set-up. With regarding to unsuccessful call set-up, tones and announcements are addressed for speech, 3.1 kHz audio, and 64 kbit/s unrestricted digital information with tone and announcement (UDI-TA).

4.1.3 Interconnection Parameters & Looping

Companies who have completed SS7 interconnection have a need and responsibility to communicate to their interconnected network providers the passage of certain parameters, some of which are identified below.

Certain 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 Routing section). Loop detection and control is important in such circumstances because, in addition to causing call failure, loops consume network capacity and diminish the likelihood of completion of calls that do not involve looping as well.

4.1.3.1 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 incoming Initial Address Message (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 I-IWU acting as an independent exchange should perform the normal 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*, sections 6.1.3.8 and 7.1.4, describe the interworking of the SS7 hop counter and the analogous SIP max forwards parameter.

4.1.3.2 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 TDM or IP networks. The current state of the implementation in the industry will not support a full resolution of problems induced by customer call forwarding using these capabilities; therefore, the NGIIF offers the following recommendations in addition to the recommendations listed above and located in NGIIF documentation:

- Network providers should review the use of simultaneous call forwarding restrictions that limits the number of times calls can be forwarded through a single line.
- Service providers 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 a customer 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 rural call completion problems, it may be the result of either translations errors or the involvement of multiple call completion vendors who are unaware of which other vendors' networks the call may have already traversed.

4.1.4 SS7 Inter Network Trunk Signaling Testing

The 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 (2) interconnecting networks may have some additional tests they may wish to perform during interconnection. In this case, these tests should be part of the bilateral agreements developed for SS7 network interconnections.

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

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 defines 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 (ATIS-0300011 to ATIS-0300022, *NGIIF Reference Document Part III- Attachments A through K*) 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.

4.1.5 Call Set-up Delay

Call set up is part of the TDM SS7 and IP SIP protocols in the Public Switched Telephone Network, for the connection of the Calling Number to the Called Number across the Public Switched Telephone Network.

Callers expect that the call set up happens in a timely manner. While ITU Recommendation E.721 suggests a mean of 5.0 second a 95%ile of 8.0 for "national" (as opposed to local) calls, conventional wireline-to-wireline calls are often considerably faster. Thus, 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 hearing “dead air”.)

For SS7, the standards for the timing between SS7 messages are specified in ATIS – 1000113.4.2005 (R2010) in Table 3, page 4-123. This table is for Timers. In that table, T 11 shows the Address Complete Message timing as 15 - 20 seconds in response to the IAM. Along with the T 7 Initial Address Message timing to Address Completion Message of 20-30 seconds, thus 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 and generally scale from estimate of round trip transmission time which 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 7.7.1: SIP timers

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 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 Section 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 dialing 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 providers either fail to release in a timely fashion 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, carriers should expeditiously release calls that they cannot complete. Carriers should not queue calls for an extended period or cycle calls through further underlying carriers.

Even without failure to release in a timely fashion, complicated routing arrangements can result in undesirable post dial delays. Consider a group of underlying carriers each of which has some potential routes to a given termination. Those routes may include, however, making use of another carrier in the set whose routes may in turn include some of the same routes used by the initial underlying carrier as well as additional members of the set of underlying carriers. This situation is analogous to that in which a group of LAN bridges is interconnected and the "spanning tree"[8] algorithm must be employed to enable only certain paths so as to avoid looping. In the absence of information about other carriers' routes, looping may result as well as fruitless reattempts of routes already tried by other carriers. The potentially large number of routing attempts by itself may introduce unacceptable post dial delay.

4.1.5.1 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 Address Complete Message.

Delayed ringing may be due to call set up delay; see 4.1.5 above.

In SIP signaling, ring back should depend on receipt of a 180 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. See section 4.1.2.1, *NGIIF SS7 Cause Code and Tones & Announcements*, and IETF RFC 3261 for more information.

4.1.5.2 Called Party Hears "Dead Air"

There have been reports that a called party may answer a call and hear only 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.

4.2 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, etc.

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.

Important parameters for transmission quality are included in the best practices for managing underlying carriers.

4.2.1 Fax

Time division multiplexing circuits using the G.711 codec support facsimile as well as voice transmission. VoIP can support fax as well 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.

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

G.726 is a waveform codec that operates at lower bitrates than G.711. Some VoIP providers 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 at fax or modem transmission speeds slower than normal.

4.2.2 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.

4.3 Routing

Multiple entities will be involved in handling all calls except those that originate and terminate within the network of a single provider. At a minimum, there will be an originating and a terminating provider and, generally, there will be an interexchange carrier or carriers involved on long distance calls. Coordination in routing is thus a prerequisite to successful call completion.

Each entity has particular responsibilities. The terminating provider, LEC, CLEC, or CMRS, that serves the called number must populate the necessary information into industry databases (LERG and NPAC) so that originating and intermediate entities can determine the serving switch and its homing arrangements.

The entity providing Long Distance (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 carrier makes use of subcontracting entities, it needs to agree with those entities on the reachability to be provided.

This section discusses industry data sources and procedures for their population and use, as well as other issues related to NPA/NXX routing.

4.3.1 NPA/NXX Routing

NPA/NXX routing is the analytics of digits dialed, jurisdictional decisions, and subsequent route selection used 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 interconnection, routing, and billing components that work concurrently together in call completion.

- Facility provisioning and interconnection is assumed to be in place prior to establishing routing for those facilities.
- Routing table translations.
- Traffic routing.

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., should exist:

- In-depth understanding of numbering resources which includes ordering, billing, and notification processes (Telcordia's BIRRDS products – e.g., LERG, TPM):
    - o The LERG is referenced in various ATIS guidelines – e.g., COCAG, TPBAG – and is considered an integral part of the routing data exchange among service providers.
    - o The LERG 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 Timeliness of receiving BIRRDS products notifications (daily/monthly/quarterly) is important to the maintenance and integrity of routing tables.

  - NPA/NXX numbering resources including thousands-block data entry into BIRRDS products is important to ensure call completion and assist in trouble shooting.
- Knowledge and understanding of the operation of the Public Switched Telephone Network (PSTN):
  - Switches.
  - Interconnection.
- Knowledge and understanding of VoIP and SIP interaction with TDM and SS7 interaction in the PSTN.
- Knowledge and understanding of EAS or franchise and LATA/MTA boundaries, Exchanges/Rate Centers (handling of interLATA EAS – LNP queries when interLATA LRN is returned).
- Knowledge and understanding of the various tandem functions within a given network architecture (access, local, intraLATA, interLATA, intermediate, operator services, 911).
- Knowledge and understanding of network interconnection homing hierarchies.

4.3.2 Interconnection Agreements

For calls to originate and terminate within the Public Switched Telephone Network (PSTN), numerous companies must interface physically, thus “interconnecting” with each other. Interconnection is *not* automatic. Contractual agreements must be established between *all* physically interconnecting companies.

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

NOTE: An IXC provider follows the 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 service providers in the call path to complete a local or toll call.

4.3.3 Homing Arrangements

Homing arrangement(s) is the last choice trunk group(s) between switching system(s) in a specific routing ladder. An SP’s subtending switch serving a portion of an incumbent LEC’s franchise territory should home on the appropriate tandem as designated by the incumbent LEC.[9] A competitive tandem in the same territory may have a different serving arrangement from the incumbent LEC. An SP needs to negotiate serving areas with the competitive tandem company to determine the homing arrangements for the SP’s subtending switch.

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

Once a valid effective date is determined, the NPA/NXX, valid switch, and supporting homing arrangement information must be entered in a timely manner into the Telcordia™ Business Integrated Routing and Rating Database System (referred to as BIRRDS), which is operated and maintained by Telcordia TRA, for notification to other carriers via the Telcordia™ LERG Routing Guide and related output from this database. Delays in entering this data will increase the probability of calls being blocked on the effective date.

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.

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

Incorrect homing arrangements in BIRRDS may result in blocked calls destined for a switching entity. For example, if the BIRRDS data entries for a switching entity indicate that the switch homes on a particular local tandem when in fact, it does not, the local tandem company will know, in all probability, how to route calls correctly, which originate from its own subscribers. Other companies, however, will route the calls to the local tandem in accordance with LERG entries. The local tandem may block the calls, if there is no connectivity between the local tandem and the terminating switching entity. Likewise, there may not be interconnection between the local tandem and a toll tandem owned by the same company. Once the calls reach the local tandem there is nowhere for the local tandem to terminate the traffic, and it will be blocked.

4.3.4 Routing Implementation

- *General*:
  - There are industry recommended minimum BIRRDS data entry time intervals for network activity that should be followed to minimize problems associated with call termination/call completion. 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 a code):
    - 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.
    - If the time intervals are not followed it may result in call failure, network blockage, credit cards may not be able to go through processing, etc.
    - Expedites for establishing a new NPA/NXX or Modification/Disconnect of a NPA/NXX may run the risk of inadequate time for other service providers impacted by the code activity to implement the code establishment or modification in their networks.
- *External* (Industry Notification):
  - Switching information (End Office, Tandem homing arrangements) published in BIRRDS/LERG:
    - The BIRRDS/LERG information for routing traffic to a rural company is the appropriate source for parties terminating calls to rural companies to use to set up their routing. Providers intending to route non-local traffic to a rural carrier, which does not indicate a toll tandem, should direct their traffic as shown in the LERG. Parties are encouraged to keep their routing information updated in the LERG.
  - NPA/NXX and NPA/NXX-X assignments published in BIRRDS/LERG. Parties are encouraged to ensure that their block information is updated in the LERG.
  - Some companies may not subscribe to the BIRRDS products, and may use alternate sources for obtaining data; however, they need to make sure they have current data populated in their switches.
  - SPs may use third party vendors for inputting the data into BIRRDS. 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 and NPA/NXX-X and switch data into BIRRDS within seven calendar days of assignment by the NANPA or pooling administrator per the INC Central Office Code (NXX) Assignment Guidelines (COCAG) and Thousands-Block Pooling Assignment Guidelines (TBPAG).
  - To ensure that traffic is properly routed to rural carriers, parties should negotiate the appropriate agreements for traffic termination through the appropriate tandem.
  - Currently, there has been no industry consensus resulting in a requirement that the NPA/NXX or NPA-NXX-X serving switch record be populated to reflect the local exchange tandem homing arrangement(s) in BIRRDS/LERG. However, parties are encouraged to ensure that their information is correctly and completely populated in the LERG.

- *Internal* (Building Routing Instructions):
    - Obtain interconnection layout for each office for which you are doing the routing.
    - Apply applicable knowledge as outlined above, relative to what you need for each office in which you are doing the routing. This includes end offices, wireless offices, POIs, remotes, and tandems.
    - Pass routing information to routing translations department for implementation in the appropriate switching infrastructure.
    - Some originating companies may elect to implement alternate routes other than what is published in the LERG.

4.3.4.1 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., LERG).
- 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 a service provider should take into account when selecting a primary route:
    - Routing must be based on dialed digits.
    - Routing arrangements between originating office(s) and terminating office(s).
    - IntraLATA versus interLATA, if applicable.
    - Local calls should be routed as local calls.
    - Toll calls should be routed as toll calls.
    - Optional calling plans.
    - Signaling required on the terminating end (e.g., 7 digits versus 10 digits).
    - Determine dialing patterns on number of digits that can be dialed on a given call.
    - Determine when dialed digits require digits to be deleted and/or prefixed.
- Additional Routing Considerations:
    - It is important to understand the LERG 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. Thus, the LERG does not provide end-to-end routing information. Interexchange carriers must on their own establish routes between originating offices and terminating local network as defined in the LERG. Providers generally create an automated tool for this function. There is no industry database of record for this information, which is considered proprietary.
    - Not all originating service providers may 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 on in the LERG. Additionally, originating service providers may opt to route through other providers due to various network conditions (for example, network congestion) to reach the terminating end office.
    - Originating service providers, 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 section 5, *Management of Underlying Carriers*, of this Handbook for additional information.

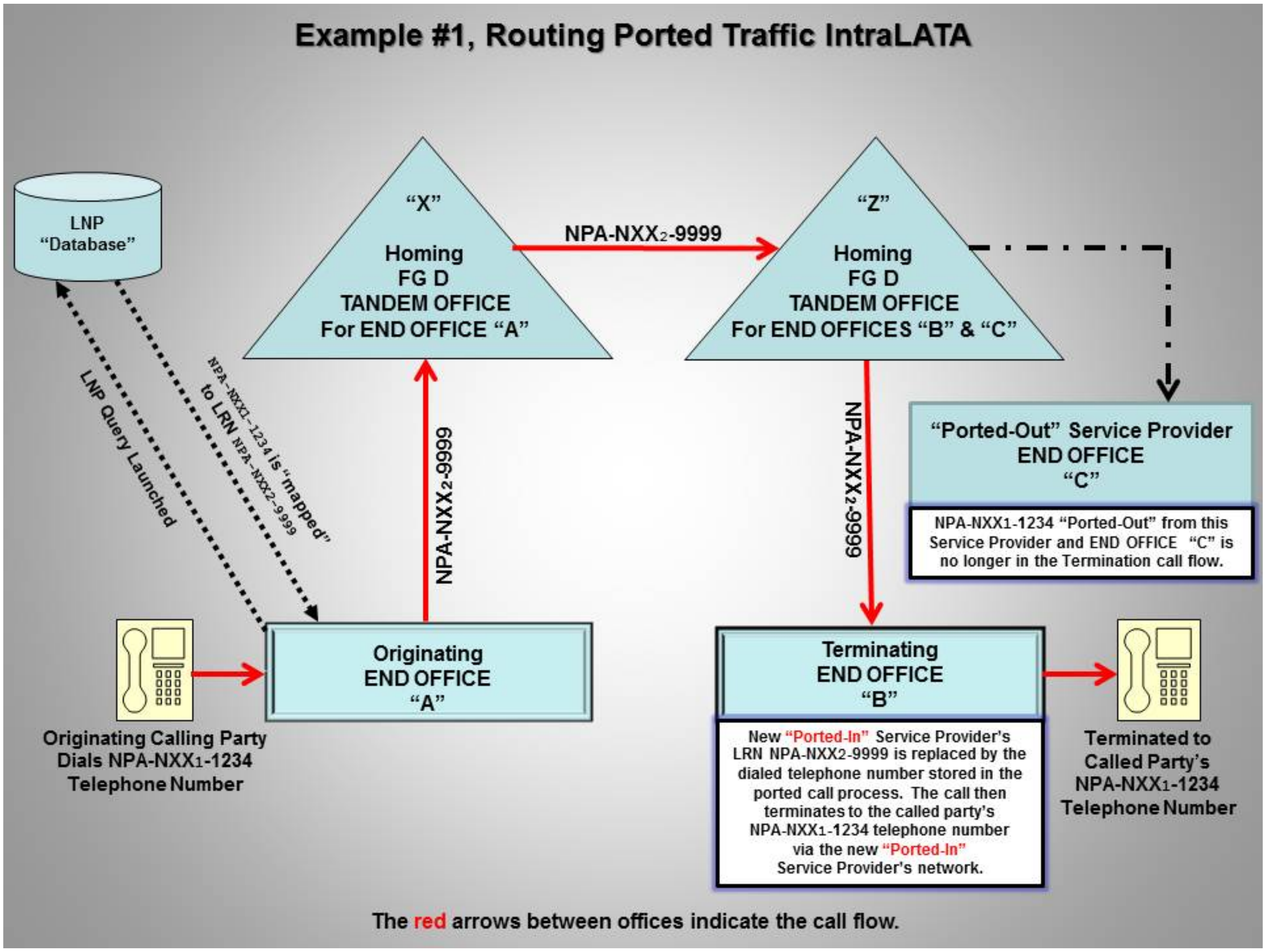
4.3.4.2 Example #1 - Routing Ported Traffic – Intra LATA

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, service providers, network components, technologies, etc. There can be variations as to which switch launches the LNP query dependent upon where a call originates, point of termination, switch capabilities, etc. (e.g., interLATA versus intraLATA, interLATA EAS, end office versus tandem, etc.).

A Code Holder is the SP assigned an NPA/NXX code (NXX-A) record or a pooled NPA/NXX thousands block as shown in the Telcordia 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 codes where the SP is the Code Holder. More information on Code Holder responsibilities can be found in the INC COCAG and TBPAG (see *References* section).

An LRN is a unique 10-digit number assigned by a Code Holder. The Code Holder selects the 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-NXX1-1234) is determined to be "ported" via a database dip that occurs in the call setup. The dialed number, NPA-NXX1-1234 is "mapped" in the LNP database to the new SP's Location Routing Number (LRN) of NPA-NXX2-9999. The NPA-NXX2-9999 LRN is processed through the call setup as if it were the called number and routed accordingly to/toward the new SP. The actual dialed number (NPA-NXX1-1234) 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-NXX1-1234) in turn replaces the LRN and the call completes to the dialed number via the new SP's network.



Example #1: Routing Ported Traffic IntraLATA

4.3.5 Potential Call Failure Points

It is understood that "call failures" may occur at any point in the path a call takes, from the point of origination to the point of termination. In today's environment, a call may traverse multiple service providers, multiple switching entities, multiple LATAs, multiple technologies, etc. It has become commonplace for a call that originates from a PSTN network, for instance, to terminate ultimately in an IP network or cellular network and vice versa. Further, there are service providers who enter into contractual arrangements with other providers, sometimes referred to as intermediaries, to "carry" their traffic. The intermediary service providers, may in turn, have contractual arrangements with other intermediary providers, and so on. Multiple technologies, service providers, etc., create a cascade effect in the network and generate more points of vulnerability.

Service providers who are not constrained by LATA boundaries may not require the services of an Interexchange Carrier in the call flow process. Wireless service providers, 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.

A "call failure" may occur anywhere in the call path, beginning with the originating calling party dialing or keying a telephone number all the way to the call completion point.

Following is a partial list of situations that may lead to network "call failures":

- Originating switching entity does not have the NPA, NXX, or NPA-NXX opened in translations.
- LNP Database has data that has not been updated for LNP query.
- LNP Database returns incorrect data.

- Call misrouted in any switching entity in the call flow.
- Signaling from one switching entity to another is incorrect.
- Originating or terminating wireless device may be in an area that has limited or no cell coverage.

Examples #2-#5

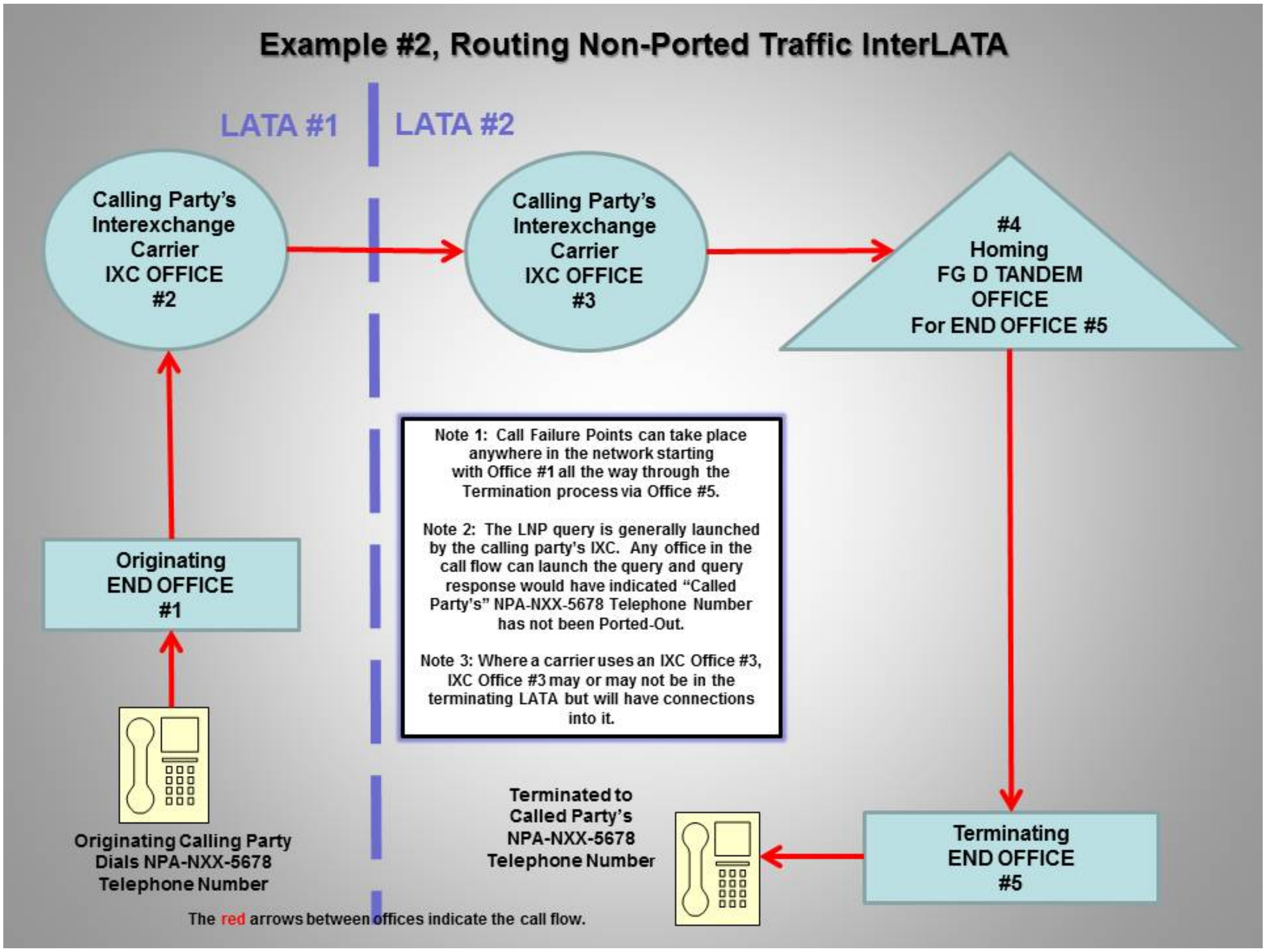
Although Examples #2-#5 depict non-ported scenarios it should be noted that an LNP "Database" query is still performed, for ported and pooled areas, to determine whether or not a telephone number has ported or is affected by pooling.

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

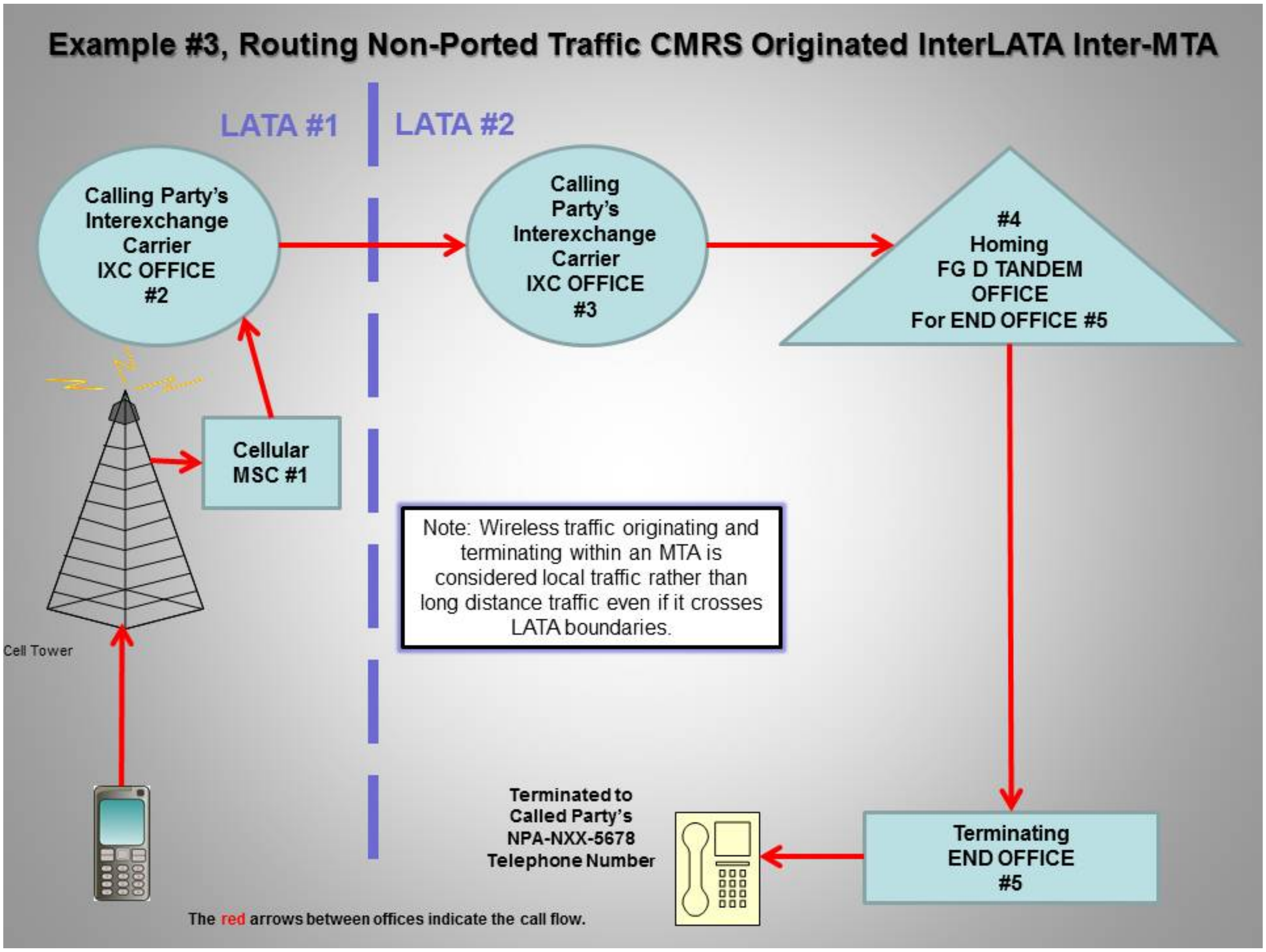
Example #4 shows a more complicated routing scenario in which the IXC does not perform an LNP query on a call to a ported number.

Example #5 shows a case where the calling party's IXC makes use of multiple intermediate carriers.

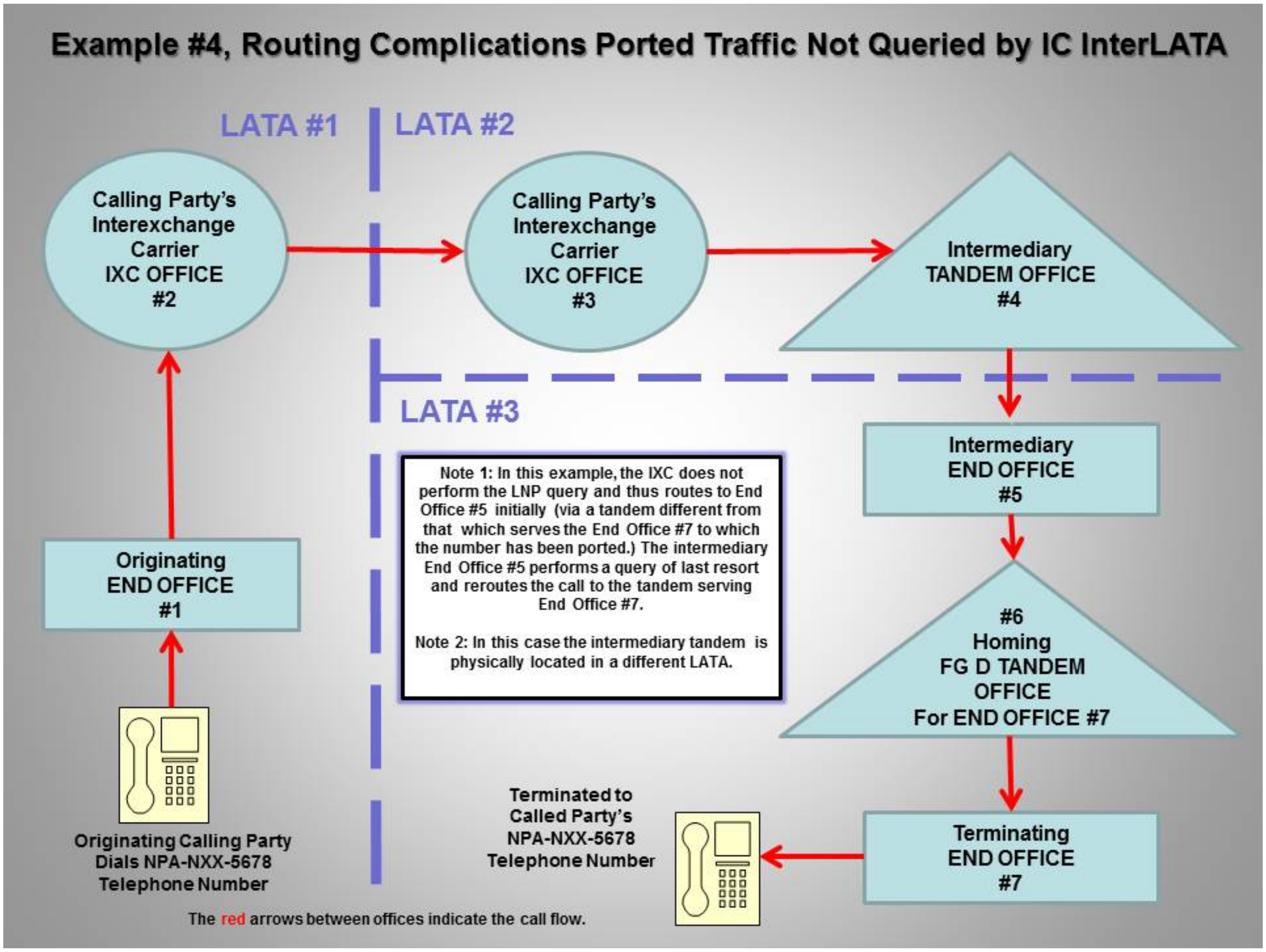
These examples are for illustrative purposes only and are not intended to represent all possible call paths, service providers, network components, technologies, etc. Each of these examples depict a "possible" call path.



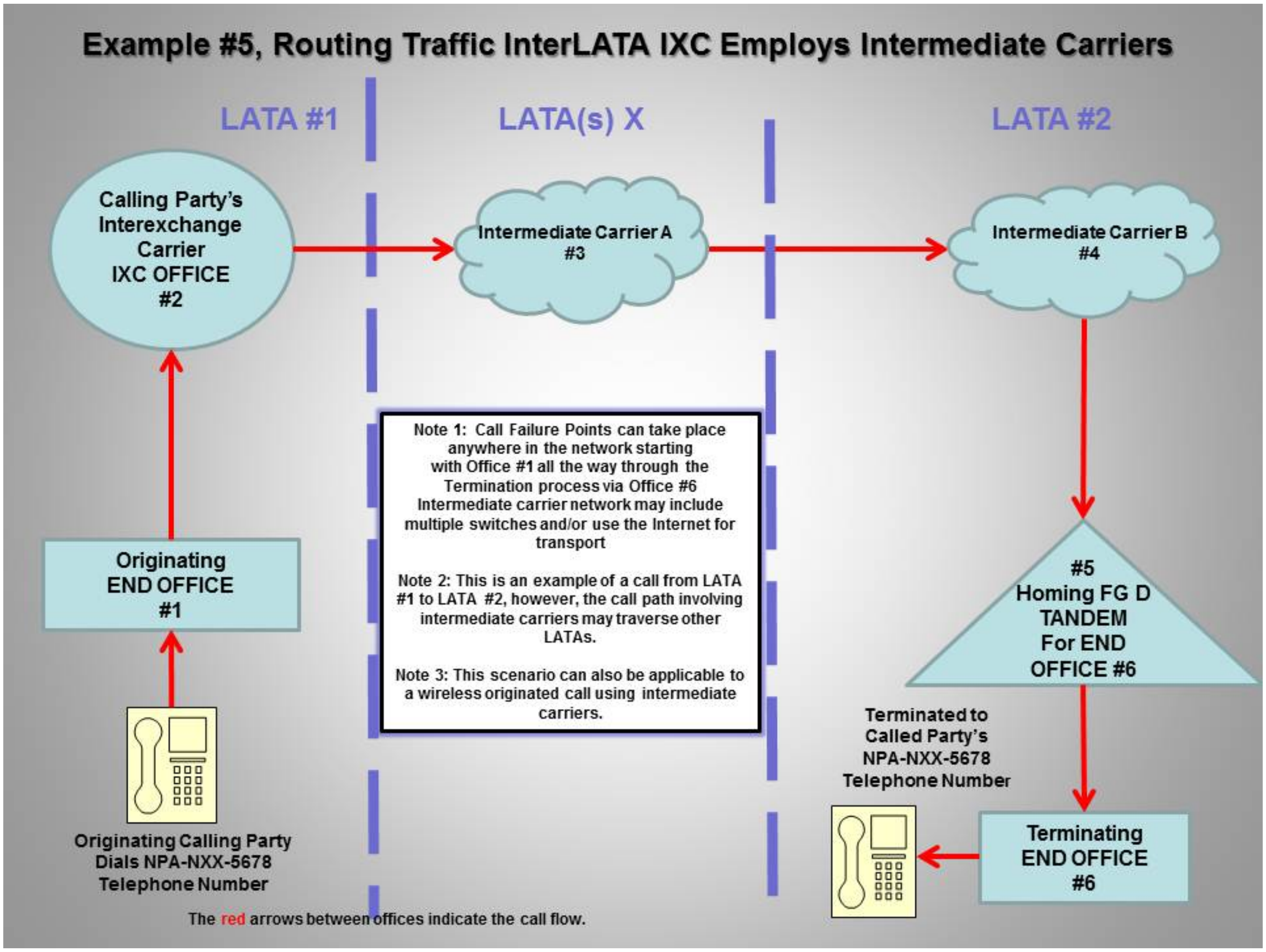
Example #2: Routing Non-Ported InterLATA Traffic



Example #3: Routing Non-Ported Traffic CMRS Originated InterLATA Inter-MTA



Example #4: Routing Complications Example, Ported Traffic Not Queried by IXC



Example #5: Routing Traffic InterLATA, IXC Employs Intermediate Carriers

4.3.5.1 Translations

Software translations configure various components of the network (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 into the network incorrectly resulting in calls being misrouted, call failures, etc.

4.3.5.2 Toll Free

Completion of toll free calls may be impacted by the call completion issues discussed in this handbook when the toll free number translates to a POTS number.

In addition, toll free numbers involve the SMS800 database and associated SCPs. Toll free number customers, through the Resp Orgs updating the SMS800 database and in management of SCPs, can control the routing of toll free calls 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 established in the 800SMS database and SCPs, purposely or otherwise.

4.3.6 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 carrier's network.

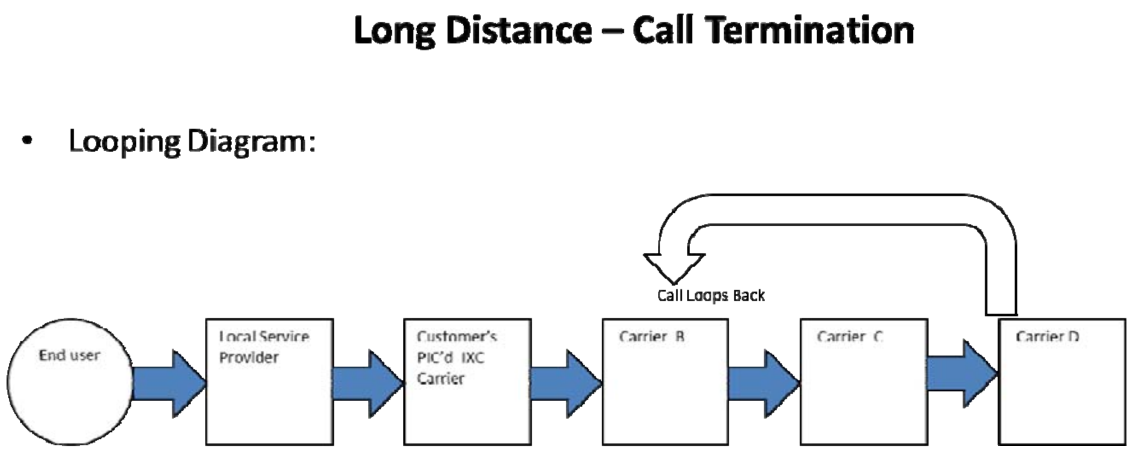


Figure 1: Basic Example of Long Distance Call Looping

4.3.7 LNP Implications

Call Termination issues may occur in some ported TN scenarios. In particular, when a native TN ports away from the native SP, and, then, at some point, the TN is ported back to the native SP, 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, in lieu of an LRN. It has been reported that, where a native TN was ported back to the native switch via an LRN, instead of the port to original option it is possible for the N-1 carrier to not recognize the LRN as valid and create a "looping" or call failure scenario, thus preventing the call from properly terminating at the native SP's switch. Although failure to properly complete a call ported back to the code holder via an LRN represents a violation of LNP processing standards [ATIS-1000002, October 2004], carriers may are encouraged to use the port to original option. More information can be found in section 7.6, *LNP Impacts*.

4.4 Network Congestion

This section will discuss or identify some situations where customers may experience call completion or call termination cases of trouble. Typically network congestion results when traffic (customer calling) exceeds the capacity of the network, unless the network has already been degraded by some other physical situation. When very high calling volume occurs, the direct engineered route may overflow to the designed alternate routing path(s). Up to a point, this is normal and has been anticipated since the additional traffic that overflows the direct route takes advantage of the spare capacity in the network. 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. The result of this condition often results in lost calls.

A common cause of network congestion may be attributed to the mass calling generated by such calling situations as telemarketing, political campaigns, or Emergency Notifications 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 Network (NGN) systems. This situation may be due to one or more of the following overload factors detailed below.

4.4.1 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 also limit the call handling capacity of the Network and contribute to Network Congestion.

- Switching systems.
- Facility systems.
- Signaling systems.
- Wireless systems.
- Next Generation Networks (example VoIP).
- Routing assignments.

4.4.2 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 making the trouble report may not be aware of congestion events occurring in other areas of the country or networks between point A and point B.

Networks are normally designed to accommodate Average Business Day (ABD) 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. Some auto dialer call types may concern/confuse customers who hear “dead air” or experience “abandoned call” situations as discussed below.

4.4.2.1 Auto Dialers

4.4.2.1.1 “Dead Air” or “Abandoned Call” Situations

Consumers hearing “dead air” or experiencing an “abandoned call” may not be aware that they are 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 (2) seconds of the person's greeting, FCC regulations consider the call "abandoned" and require the dialer to play a recorded message. The FCC requires that predictive dialers 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 dialer, and the called party receives a silence “dead air” or a tone from the telephone company 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 await answer before hanging up.

There may be FTC or FCC regulations and/or requirements however; there is no way to ascertain if these requirements are followed by generators of auto dialed type calls, or if consumers are aware that they are receiving an auto dialer type of call.

Additional information on auto dialers is available in ATIS-0300105, *NGIIF Auto Dialer Reference Document Auto Dialer Basics*.

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

A consumer may generate a customer complaint or trouble report when they are aware of an automated announcement being 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 many factors which go into the determination of ENS call completion. Some of those factors include, but are not limited to, time of day, holidays, trunk capacity, host or remote switching configurations, traffic patterns, length of announcement, or if the ENS call was originated locally or from another area traversing between multiple networks. Other factors may include completeness or accuracy of the calling system telephone number database and reaction of calling systems to answering machines, voice mail, no answer, or busy line conditions. Consumers may utilize call blocking or call selection type features which may prevent receiving calls from unknown calling systems.

4.4.3 Fraud

Call completion may also be impacted by fraudulent activity in the network. Individuals or entities may purchase wireless service and use the associated SIM cards together in devices through which they offer to terminate long distance traffic by re-originating it as wireless calls. The concentration of wireless calls originating within a cell site area may congest the wireless network, resulting in poor call completion for the traffic offered to the vendors using the SIM boxes. As this usage violates wireless terms of service, carriers will shut the associated service down, but not necessarily before congestion-induced blocking occurs. Even after shutdown, calls may continue to route to the entity that had set up the SIM box and engender call completion problems.

Wireline-based fraud schemes also exist and likewise impact call completion when their turndown takes out routes that carriers may have unwittingly counted on.

4.4.4 Force Majeure & Disasters

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

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

4.4.5 Human-related Issues

Human-related congestion issues, particularly accidents, is another area that has the opportunity to not only cause physical damage to a Network, but compound the situation by also generating excessive customer calling attempts. This includes:

- Accidents.
- Human error.
- Planning or forecasting miscalculations.

4.4.6 Traffic Pumping & Access Stimulation

Access stimulation occurs when a LEC with high switched access rates enters into an arrangement with a provider of high call volume operations such as chat lines, adult entertainment calls, and “free” conference calls. 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 “free” service provider, or offers some other benefit to the “free” service provider (¶ 656 of the USF/ICC Reform Order).

One condition of access stimulation is when a LEC has had a greater than 100 percent increase in interstate originating and/or terminating switched access Minutes of Use (MOU) in a month compared to the same month in the preceding year. (¶ 667 of the USF/ICC Reform Order) Such a sudden increase in traffic could cause network congestion if trunk groups are not properly sized for this volume of traffic. At least initially, until such time trunks can be augmented and/or some of the traffic rerouted, this network congestion may result in call completion/call termination issues.

5 Management of Underlying Carriers

Some carriers have suggested that reported call completion/call termination problems involve the use of underlying carriers, so-called “least cost routers” by the carriers that actually have the retail long distance relationship with the calling party. Least cost routing has been used in the industry for many years, including carriers in all geographies. As a result of this experience, carriers have developed ways to effectively manage underlying carriers. This section identifies best practices for management of underlying carrier networks.

The responsibilities of underlying carriers pertinent to call completion/call termination should be defined in an agreement between the service provider and the underlying carrier with which it contracts.

This contractual arrangement between the service provider and the intermediary will determine and specify procedures for testing, updating, implementing, and maintaining the geographic area(s) and/or NPA-NXXs for which a given underlying carrier has the responsibility for completing calls to the appropriate end destination.

5.1 Manage the Number of Intermediate Providers

As the number of providers handling a call increases, there is the potential for lengthier call setup delay and other impairments. Troubleshooting may also prove more difficult. Some carriers have found it useful to limit underlying carriers to including no more than one additional provider (not including the terminating carrier) in the call.

5.2 Management of Direct & Indirect Looping

There may be cases where an underlying carrier may purchase wholesale service from the IXC that is initially handing off the call to the underlying carrier. This behavior may result in looping as well as adding delay and other impairments in the call setup.

5.3 Crank-back on Failure to Find a Route

If an underlying carrier cannot find a route to the termination, it should release the call back to the original IXC in such a manner as to allow the IXC to attempt to complete the call over its own facilities rather than dropping the call as some underlying carriers do.

5.4 Maintain Sufficient Direct Termination Capacity

In conjunction with crank-back, it is important for the original IXC to maintain sufficient termination facilities that it can complete on its own traffic that a call completion vendor cannot. Following are two reasons why this is important:

- First, given the incentives to maintain a lean network that underlying carriers face and their aggregation of loads from multiple IXCs, there is a greater chance that, on a moment-to-moment basis, they will not have capacity to complete a call.
- Second, maintaining its own termination capacity gives an IXC flexibility to quickly stop using an underlying carrier should performance problems develop.

5.5 Do Not Terminate & Re-originate Calls

Underlying carriers 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 caller may receive a ringing indication well before the called party is alerted, which is one of the problems reported by the rural LECs.

5.6 Direct Measures of Quality

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

Table 2: 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., TDD and 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	% 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

5.7 Do Not Manipulate Signaling

Underlying carriers should not manipulate signaling information, especially the Calling Party Number (CPN), so as to obscure proper jurisdiction for settlements, such as through inappropriate out of country routing.

5.8 Inheritance of Restrictions

Where an underlying carrier makes use of an additional underlying carrier to reach the terminating carrier, the first underlying carrier contracting with the IXC should in turn manage their underlying carrier to the same standards required by the original IXC.

5.9 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.

5.10 Require Acceptance Testing

Before offering live traffic to an underlying carrier, an IXC should conduct acceptance testing with the underlying carrier to ensure compliance with call processing requirements and DMoQs.

6 Trouble Reporting & Contact Directories

This section provides information on trouble reporting and the Service Provider Contact Directory, which includes IXC carrier to carrier contacts. Among other things, this section lays out trouble reporting responsibilities of carriers that provide originating or terminating access for long distance calls and their local service customers, the interexchange carriers.

Rural LECs have pointed out the difficulties involved in resolving troubles that largely present themselves in the form of calls that do not reach their customers or even their networks. Effective trouble isolation depends on identifying the carrier(s) that actually handle a given call. Originating callers long distance traffic may not be handled by or through their local service provider or its affiliate, and where callers are PIC'd to an independent entity, the originating local service provider may be restricted from providing that information to a terminating LEC reporting a completion issue. As unwieldy as it may be, the terminating LEC needs to get the wireline originating caller to confirm their IXC carrier 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 carrier 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 a call failure occurs. Delays in reporting the case of trouble degrades the ability of the carrier to identify, isolate, and investigate the cause of the trouble reported. Untimely trouble reporting can result in network circumstances or routing table changes which may have occurred; retention periods for items like trouble logs may have passed; the ability to duplicate the specific trouble situation may no longer be possible. These are just a few examples of why it is important to place a trouble report after the problem occurs to enable the carrier to successfully investigate the case of trouble.

The NGIIF has multiple documents addressing trouble detection, reporting, management, etc., for different aspects of telecommunications. Some of those documents are identified below.

- 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 section 6.1.

6.1 Trouble Reporting

The NGIIF recommends immediate reporting of non circuit-specific troubles in order to facilitate the rapid restoral of service.

Each company will provide contact information for customer trouble reporting.

Existing trouble handling procedures for interexchange calls focus on the case where a trouble is reported by the calling customer to their carrier and where there is a direct connection between the IXC and terminating local service provider. 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[10], normal procedures can be used to resolve the issue. Where this is not possible, the called party's local service provider will seek to contact the carrier they believe to be the calling party serving IXC[11]. Except by report of the calling party directly or as reported by the calling party to the called party, the terminating service provider will not be able to identify the responsible IXC directly. Instead, they may determine the caller's local service provider and contact them. Where the local service provider is also the calling party's long distance provider, the trouble can be addressed by the long distance entity on behalf of the calling party. If the terminating carrier tries to report the

[10] To confirm the identity of their IXC, the calling party should dial 700-555-4141 for wireline originations.

[11] Carriers should contact the appropriate IXC via the NGIIF's Service Provider Contact Directory.

issue over the normal long distance repair line of the presumed IXC, the trouble may not be accepted, because industry guidelines expect each carrier to only work troubles reported by their customer.

If the report is made by a separate carrier-to-carrier channel, which some IXCs have set up in response to the rural call completion situation, and the caller's local service provider also happens to be the caller's IXC as well, they may be able to address the problem, although CPNI restrictions may prevent them from working the trouble without first contacting their customer and obtaining permission. CPNI restrictions may also prevent sharing full details of resolution with the called party's carrier. Where the caller's local service provider is not the caller's IXC, CPNI restrictions will prevent the caller's local service provider 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 carriers related to trouble resolution.

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

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

- Trouble report number or equivalent.
- Contact telephone number.
- Contact ID (i.e., name or initials).
- Time and date report was received from the responsible carrier.
- Responsible carrier testing testing information (If requested by any receiving carrier(s).
- Circuit ID (41-Character 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.

6.1.1 Trouble Reporting Detection Responsibilities & Processes

This section provides operation's personnel of interconnecting carriers with guidelines for trouble reporting.

This section 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.

6.1.2 Responsibilities

Carriers receiving a trouble report have the following baseline responsibilities when investigating their trouble report with other carriers.

6.1.2.1 Carriers Generating Trouble Reports

- Provide trained personnel.
- Advise the relevant carriers 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 their end users.
- Accept trouble reports from other carriers. Ensure the test equipment used is compatible with the other relevant carrier's test equipment.
- Assume control functions for maintenance of its trunk(s).
- Consult with other relevant carriers before requesting any changes, except under emergency conditions.
- Sectionalize and clear the trouble in its own network.
- Test cooperatively with other relevant carriers to identify and clear a trouble, when the trouble has been sectionalized to a network.
- Keep their 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 carriers using the trouble reporting procedures.
- Dispatch its own maintenance forces.
- Perform verification tests to ensure that trouble has been cleared.
- Participate cooperatively with other carriers to further isolate and clear the trouble when trouble exists and cannot be sectionalized to a particular carrier portion.
- Where it is technically feasible, signaling for all internetwork calls to a 10 digit telephone number should always be sent or received using 10 digits for the called party number, independent of how the call is dialed.

6.1.2.2 Carriers Receiving Trouble Reports

- Provide trained personnel.
- Advise the relevant carriers 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 carriers before requesting any changes, except under emergency conditions.
- Provide access to test lines where appropriate.
- Notify the receiving carrier of any changes affecting the service requested, including the service due date.
- Accept trouble reports from carriers generating trouble reports.
- Sectionalize and clear the trouble in its own network.
- Test cooperatively with other relevant carriers to identify and clear a trouble when the trouble has been sectionalized to the other carrier's network.
- Perform cooperative analysis to determine if a trouble pattern exists.
- Refer troubles to other relevant carriers using the trouble reporting procedures.
- Dispatch its own maintenance forces.
- Clear troubles in its own network.
- Perform verification tests to ensure that trouble has been cleared.
- Provide status reports to the carrier who generated the trouble report.

6.1.2.3 CPNI

Customer Proprietary Network Information (CPNI) must be protected as carriers work cooperatively to resolve trouble reports related to call completion/call termination. Refer to Section 7.5 below for further explanation of CPNI.

6.1.2.4 Sectionalization

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

6.1.2.5 End-To-End & Intercarrier Testing

Circuit networks comprising carrier services may experience trouble conditions that cannot be isolated by each carrier testing and maintaining its own services. Although the portions provided by each carrier apparently perform properly, a trouble may be identifiable on an end-to-end service. In such cases, the carrier generating the trouble report may require coordinated testing. See Section 6.2, *Use of Test Lines for Call Completion Trouble Resolution*.

As a best practice, following the NGIIF Guidelines noted below, service providers should publish test numbers associated with specific NPA/NXX in the LERG (in accordance with the guidelines below), so that originating carriers can make test calls to test call quality proactively and to test when any customer or carrier refers a call quality issue to the originating provider. Without such capability, the originating provider 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*: 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 company opening a new 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.

6.1.2.6 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.
- Other.

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

6.1.2.7 Trouble Report Clearing Information

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

6.2 Use of Test Lines for Call Completion Trouble Resolution

One way in which terminating service providers may be able to expedite trouble resolution, in cases where the trouble has been reported by the called rather than the calling party, is to provide a test line number for the destination end office in their trouble report. As discussed in Section 6.1, CPNI issues can complicate working troubles in the called-party-complains scenario. An IXC 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.

6.3 Call Setup Time Trouble Reporting & Sectionalization

ATIS-0300010 defines installation and maintenance responsibilities of LECs providing switched access service (Access Service Providers or ASPs) and IXCs obtaining switched access service from them (Access Service Customers or ASCs). The terms used in this section are the same as within ATIS-0300010 (ASP and ASC). The document thus focuses on situations in which IXCs and LECS are directly connected. In many of the scenarios of concern in this document, it is the underlying carriers employed, sometimes indirectly, by the IXC with the retail relationship with the calling party, that are the customers of the ASP. 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.

- *Access Service Customer (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.
- *The Access Service Provider (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 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 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 2- 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 Customer Premises Equipment (CPE, etc.).
- Type of access (e.g., 101XXXX, DDD).
- Directionality of the call(s) on which trouble was reported.
- Calling and called telephone number.
- Time of day the reported problem is experienced.
- End user's estimation of call setup time.
- Any other pertinent information which can be supplied by the end user.

Contributing factors to call setup time troubles could include:

- Manual/auto dialing.
- Customer call forwarding options.
- 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 700-958-1102 or 700-959-1020 as appropriate (see Figure 3 or 4).

Access performance limits have been established based on the information contained in the Local Switching System Generic Requirements (LSSGR) 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 be limited to, the following information:

- ASC determined ASP call setup time;
- Call direction;
- FG B-D;
- Direct versus tandem routing;
- End Office CLLI; and
- Test line telephone number used.

Upon receipt of the trouble report from the ASC, the ASP will initiate their 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 3 and 4. A description of those tests follows:
    - o 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 700-958-1102 or 700-959-1020 as appropriate to access the ASC 102 test line.

      NOTE: See Call Setup Time: Figure 3 - 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 4 - Terminating Test Procedures.

If the ASP determines there is a problem in their network, they 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.

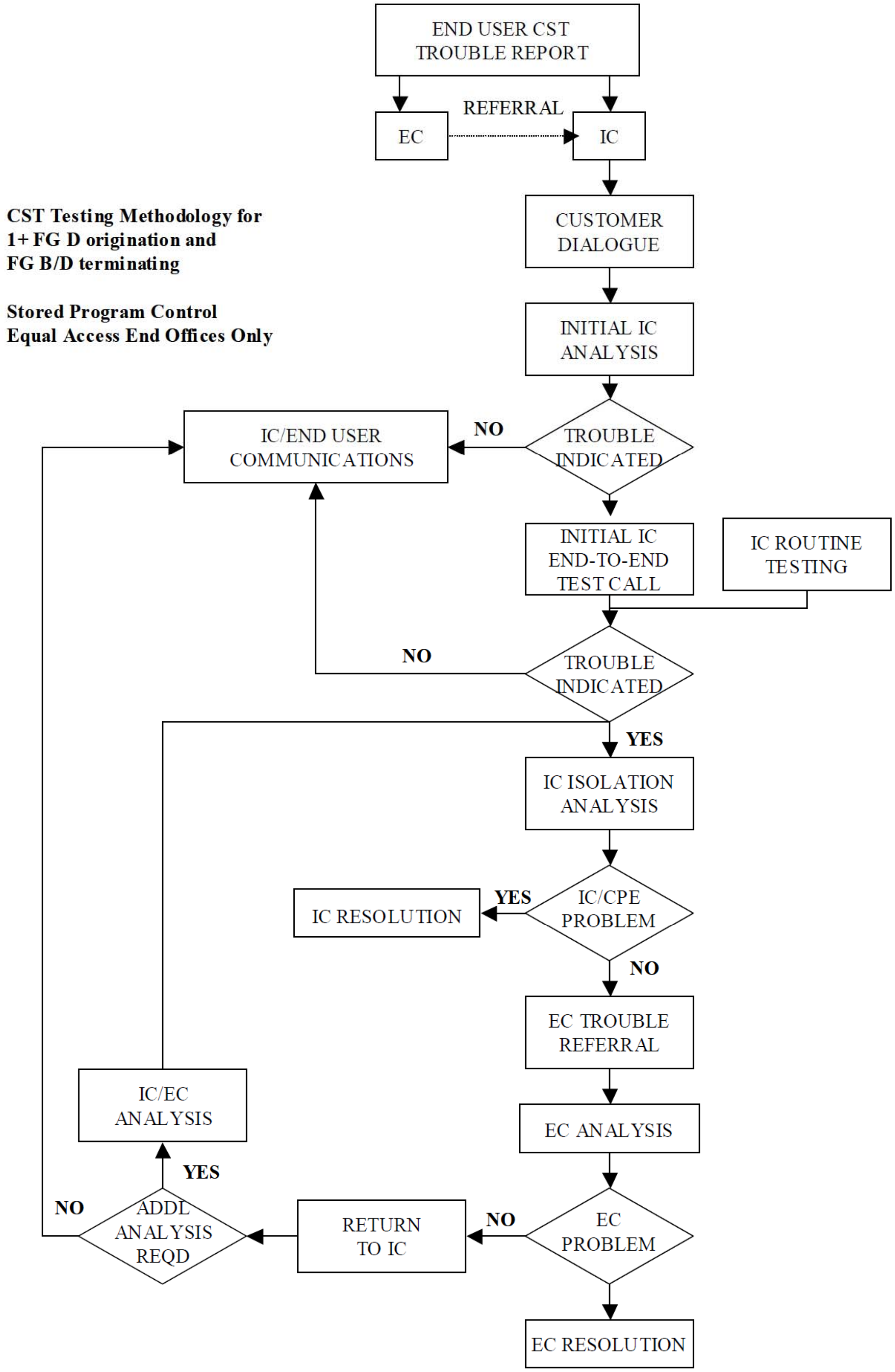


Figure 2: CST Testing Methodology

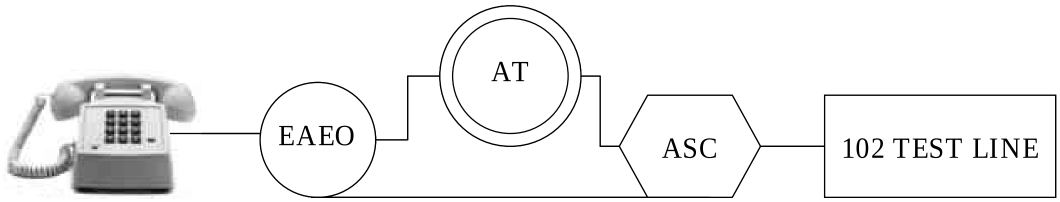


Figure 3: Call Setup Time

6.3.1 Originating Test Procedure

1. Place call from line side of originating EO to 102 test line in ASC switch. It is recommended that the ASC 102 test line be accessed by dialing 700 958-1102 or 700 959-1020, as appropriate.
2. Time stamp:
   - Start at end of last digit dialed.
   - Stop at Network response.

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

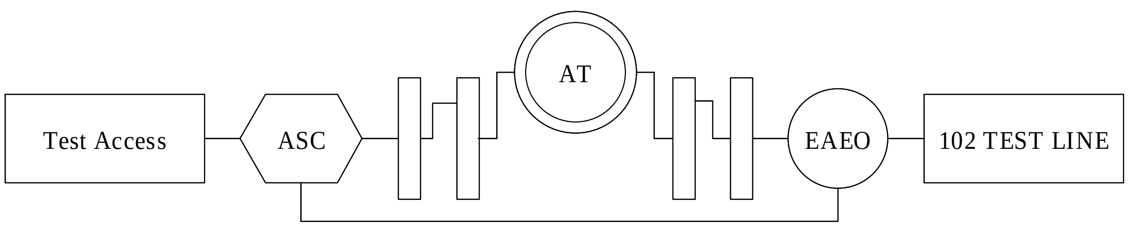


Figure 4: Call Setup Time

6.3.2 Terminating Test Procedures

1. Place call from Test Access in the ASC switch to 102 test line in EO:
2. Time stamp:
   - Start at ASP trunk seizure.
   - Stop at network response.

   May be accomplished with personal computer, stopwatch, or other test equipment, as available.
3. ASC switch time not included.

6.4 ATIS Contact Directories

The NGIIF maintains contact directories on its website, which are available at no charge to the telecommunications industry. The two directories updated and published by the NGIIF are the Service Provider and National LNP Contact Directories. Additional details regarding access and other information related to the contact directories is located below.

6.4.1 Service Provider & National LNP Contact Directories

6.4.1.1 Service Provider Contact Directory

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

The SPCD identifies intercompany contact points for providing information in a multi-platform technology arena. Any information that may be of concern to the interconnecting company's network – i.e., modifications, outages, network survivability, network congestion, testing and/or maintenance, IXC carrier-to-carrier information – should be passed on.

It is a recommendation by the NGIIF that all service providers list and update their contacts on a regular basis. The NGIIF sends annual reminders requesting updates.

6.4.1.2 National LNP Contact Directory

The purpose of the LNP Contact Directory is to provide contact numbers to the telecommunication industry for requesting interconnecting company assistance on service-related situations relating to Local Number Portability. 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 service providers list and update their contacts on a regular basis. The NGIIF sends annual reminders requesting updates.

6.4.2 How to Gain Access

The Service Provider Contact Directory and the National LNP Contact Directory can be accessed via the NGIIF webpage at < www.atis.org/ngiif >. From the NGIIF home page, select "NGIIF Documents" from the left hand toolbar, and then "Complimentary Documents". Both the Service Provider Contact Directory and the National LNP Contact Directory are listed in this section of the NGIIF website.

6.4.3 How to Add or Update Contact Information

To include and/or update your company's information in the Service Provider Contact Directory, complete a blank Service Provider Contact Directory Form. To include and/or update your company's information in the National LNP Contact Directory, complete a blank National LNP Contact Directory Update Form. Electronic copies of these forms can be accessed via the NGIIF webpage at < www.atis.org/ngiif >. From the NGIIF home page, select "NIIF Documents" from the left hand toolbar. Please email the completed form(s) to the NGIIF Manager of Standards Development, Veronica Lancaster, at < vlancaster@atis.org >. Instructions for the forms are located at: < http://www.atis.org/ngiif/contactdir.asp >.

7 Existing Regulatory Environment

This section describes various existing rules and regulations which may assist in investigating and/or resolving some of the long distance call completion/call termination issues. Although certain FCC rules and regulations are referenced in this handbook, this section is not to provide legal guidance, but instead serves as a reference.

7.1 USF/ICC Reform Order

The FCC's Report and Order and Further Notice of Proposed Rulemaking 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 Universal Service Fund and Intercarrier Compensation. Thus, carriers should be mindful of affected rules relating to long distance call completion/call termination.

7.1.1 Phantom Traffic

In paragraph 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 paragraph (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 carriers and providers 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 provider in the path from the originating provider to the terminating provider. This provision applies regardless of the voice call signaling and transmission technology used by the carrier or VoIP provider. 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 providers, and are required to transmit the calling party's charge number (CN) in the SS7 ISUP CN field to interconnecting providers 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 providers within an interstate or intrastate call path that originates and/or terminates on the PSTN must pass unaltered to subsequent providers 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 providers 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 carrier or VoIP provider.

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 paragraph 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, paragraph 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.

7.1.2 Caller ID

In addition to the rules set forth in the USF/ICC Reform Order, in the 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. Report and Order In the Matter of Rules and Regulations Implementing the Truth in Caller ID Act of 2009, WC Docket No. 11-39, adopted June 20, 2011 and released June 22, 2011 (FCC 11-100) ("Truth in Caller ID Order"). Additionally, the FCC stated at paragraph 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 carrier, spoofing provider or other service provider, 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.

7.1.3 Calling Party Number (CPN)

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

- Service providers 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 telephone number associated with the calling party to the next provider in the call path.
- Intermediate providers must pass calling party number or charge number signaling information they receive from other providers unaltered, to subsequent providers in the call path.

7.2 FCC Decisions on Call Delivery Areas

The FCC has established rules related to call delivery, and are highlighted in the USF/ICC Reform Order in the following sections.

- Footnote 1234: "Carriers are generally prohibited from blocking calls. *See Establishing Just and Reasonable Rates for Local Exchange Carriers; Call Blocking by Carriers,* WC Docket No. 07-135, 22 FCC Rcd 11629 (2007) (*Call Blocking Declaratory Ruling*). …"
- Paragraph 734 and Footnote 1279: "In the *2007 Call Blocking Order*, the Wireline Competition Bureau emphasized that…'Commission precedent provides that no carriers, including interexchange carriers, may block, choke, reduce or restrict traffic in any way.'"; *Call Blocking Declaratory Ruling* 22 FCC Rcd 11631, para. 6.
- Paragraph 973: "*No Blocking.* In addition to the protections discussed above to prevent unilateral actions disruptive to the transitional VoIP-PSTN Intercarrier compensation regime, we also find that carriers' blocking of VoIP calls is a violation of the Communications Act and, therefore, is prohibited just as with the blocking of other traffic."

Further, the FCC's Wireless Competition Bureau issued a Declaratory Ruling to clarify the scope of the Commission's prohibition on blocking, choking, reducing or restricting telephone traffic. 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, adopted and released February 6, 2012 (DA 12-154) ("Declaratory Ruling"). In the Declaratory Ruling, the FCC reminds carriers of the Commission's longstanding prohibition on carriers blocking, choking, reducing, or otherwise restricting traffic. The FCC also clarifies that this prohibition extends to routing practices that have the effect of blocking, choking, reducing, or otherwise restricting traffic.

7.3 Practices to Support Proper Jurisdictionalization of Traffic

The called party's service provider'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 service provider'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 Local Exchange Carriers and Commercial Mobile Radio Service Providers, CC Docket No. 95-185, adopted August 1, 1996 and released August 8, 1996 (FCC 96-325) ("Local Competition First Report and Order").

7.4 CPNI

In the regulatory environment, one key area to be mindful of in investigating information regarding long distance calls between service providers is the regulatory construct related to Customer Proprietary Network information (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.

7.5 LNP

Where the called party's TN has been ported, an LNP dip is required to be done. Notably, the "N-1" carrier, specifically the IXC on a toll call, is responsible for performing or arranging for any needed LNP query. Also, code holders should query and route calls that are default routed to them without a query having been performed. Regulations related to LNP can be found in 47 C.F.R. §52.26.

8 Summary

This handbook is intended for all carriers involved in long distance call completion. It has attempted to provide the existing standards and guidelines that may be relevant to the 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 carriers. 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 100% call completion at all times and that variations in completion rates will occur subject to variations in offered load on a diurnal and seasonal basis and 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 providers 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 these factors from the persistent difficulties that rural LECs have reported and not treat all instances of call failure as indicative of discrimination.

It is hoped, however, that this handbook will help mitigate the more serious issues that led to its development. This handbook is intended to be a living document, which will be updated to reflect further learnings.